Aastra Models 9000i and 6700i Series IP SIP Phones

Administrator Guide

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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 9000i (9143i, 9480i, 9480i CT)) and 6700i Series SIP IP Phones (6730i, 6731i, 6753i, 6755i, 6757i, and 6757i 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.

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.
- ***Models 9000i and 6700i Series 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
An overview of the IP Phones and the IP Phone firmware installation information	Chapter 1
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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

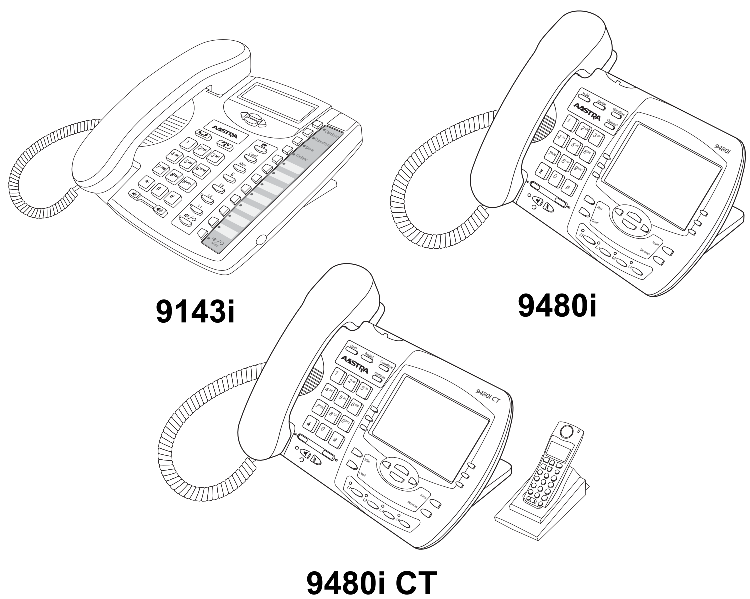
All Aastra SIP IP Phone Models communicate over an IP network allowing you to receive and place calls in the same manner as a regular business telephone.

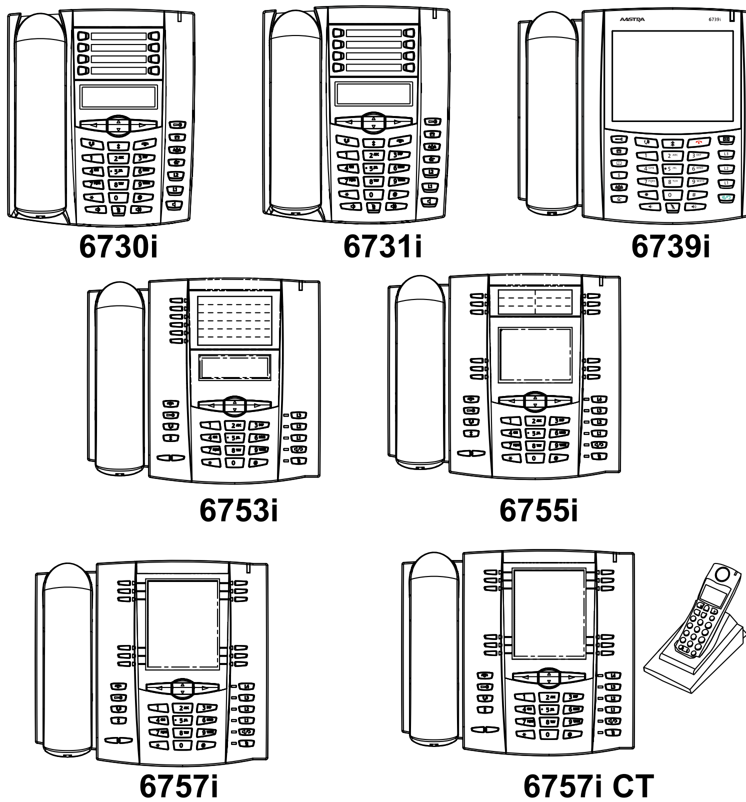
All phone models support the Session Initiation Protocol (SIP). The 9480i CT and 6757i CT offers the base phone along with a cordless extension.

References

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

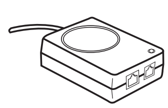
The following illustrations show the types of IP Phone Models.



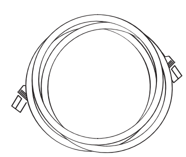


Optional Accessories (for all 9000i and 6700i IP Phones)

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.

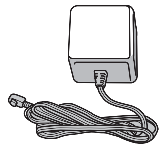
Warning: Do not use this inline PoE power injector to power other devices. See your phone-specific Installation Guide for more information.

Reference

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

Optional Accessories (for 6731i and 6739i IP Phones)

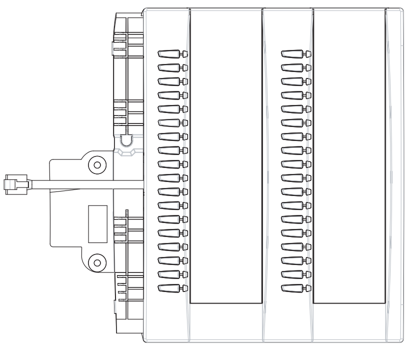
The following are optional accessories for the 6731i and 6739i IP Phones.



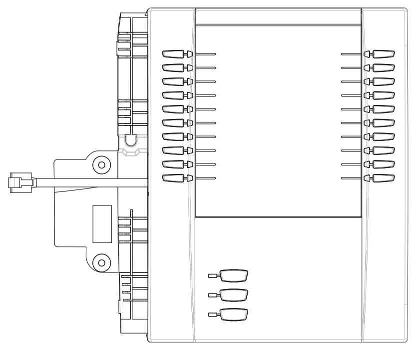
Power Adapter

Optional Accessories
(for 6739i, 6753i, 6755i, 6757i, and 6757i CT IP Phones)

The following are optional accessories for the 6753i, 6755i, 6757i, and 6757i CT IP Phones.



M670i Expansion Module for 6739i, 6753i, 6755i, 6757i, and 6757i CT



M675i Expansion Module for 6739i, 6755i, 6757i, and 6757i CT

The M670i module adds 36 additional softkeys to the IP phone models 6739i, 6753i, 6755i, 6757i, and 6757i CT. The M670i provides paper labels for each softkey. Up to 3 modules can be piggy-backed to provide up to 108 additional softkeys for the phone.

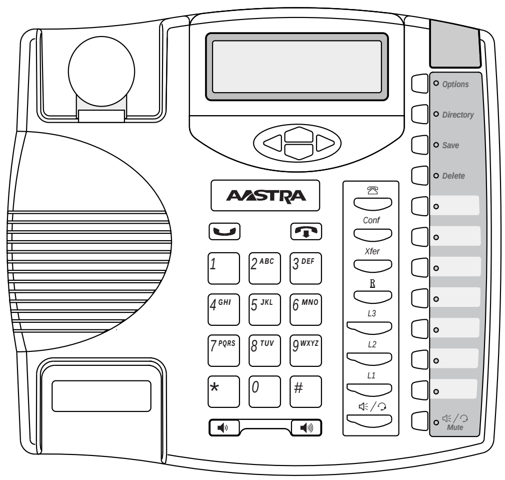
The M675i module adds 60 additional softkeys to the IP phone models 6739i, 6755i, 6757i, and 6757i CT (using the 3 function keys on the bottom right of the unit). The M675i module provides an LCD display for display softkey labels. Up to 3 modules can be piggy-backed to provide up to 180 additional softkeys 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 9143i IP Phone

This section provides brief information about the Model 9143i IP Phone. It includes a list of features and describes the hard keys on the 9143i.

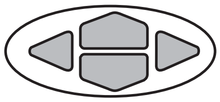
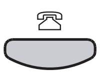
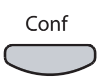
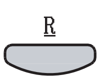
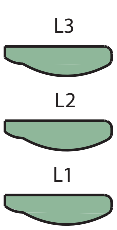


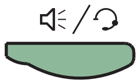
9143i Phone Features

- 3-line LCD screen
- 3 call appearance lines with LEDs
- 7 programmable keys
- Press-and-Hold speed dial key configuration feature
- Full-duplex speakerphone for handsfree calls
- Supports up to 9 call lines
- Headset support (modular connector)
- Built-in two-port, 10/100/1000 Ethernet ports - 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.

9143i Symbol Key Descriptions*

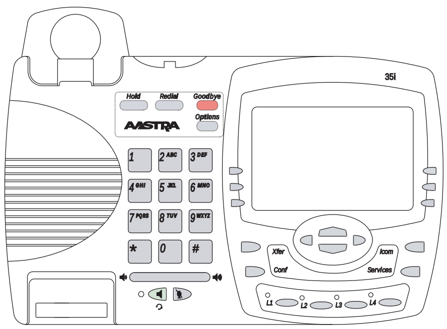
Keys	Key Description
	**Navigation keys** - Pressing the UP and DOWN arrow keys lets you view different status and text messages on the LCD display (if there is more than 1 line of status/text messages). These buttons also let you scroll through menu selections, such as the Options List. Pressing the 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.
	**Goodbye key** - Ends an active call. The Goodbye key also exits an open list, such as the Options List, without saving changes.
	**Hold key** - Places an active call on hold. To retrieve a held call, press the call appearance button beside the light that is flashing.
	**Volume control key** - Adjusts the volume for the handset, headset, ringer, and handsfree speaker.
	**Callers List key** - Allows you to access the Callers List that stores up to 200 of the last calls received.
	**Conference key** - Allows you to begin a conference call with up to 3 active call parties.
	**Transfer key** - Allows you to perform blind or consultative transfer of an active call to another number. In blind transfer, you transfer the call without waiting for the far end to answer. In consultative transfer, you wait for the far end to answer before transferring the call.
	**Redial key** - Redials a previously dialed number. The Redial key stores up to 100 previously dialed numbers you can select from. Pressing the Redial key twice simultaneously redials the last dialed number.
	**Line/Call Appearance key** - Connects you to a line or call. The Aastra 9143i IP Phone supports up to 3 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.
	**Options key** - Accesses options to customize your phone. Specific options are available to your System Administrator only and are password protected. Contact your System Administrator for more information.
	**Directory key** - Accesses the Directory on the phone. The Directory List is your personal phone book, conveniently stored within your phone. You can enter up to 200 entries into the 9143i Directory by adding them manually, or by saving the number and name from other lists stored on your phone. Each entry can contain a maximum of 16 letters and numbers.
	**Save key** - Allows you to save entries when storing numbers and names in Directory. Also allows you to save Option settings when using the programmable keys.
	**Delete key** - Allows you to remove entries from the Redial, Directory, or Callers Lists.
	**Programmable keys** - Allows you to use the feature configured for that key. You can program up to 7 keys with a specific function. By default, there are no functions configured on the programmable keys (keys are configured as “None”). **Note**: For more information about configuring the programmable keys 1 through 7 to perform specific functions, see Chapter 5, “Configuring Operational Features” the section, “Softkeys/Programmable Keys/Feature Keys/Expansion Module Keys” on page 5-130.
	**Mute key** - Mutes the microphone so that your caller cannot hear you (the light indicator flashes when the microphone is on mute).

*See the Aastra 9143i User Guide for more information about each of these keys.

Model 9480i and 9480i CT IP Phones

This section provides brief information about the Models 9480i and 9480i CT IP Phones. It includes a list of features and describes the hard keys on these models.



9480i

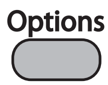
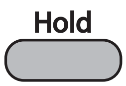
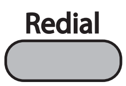
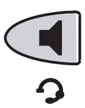
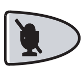
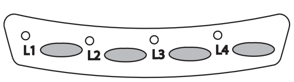
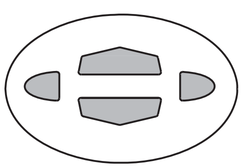
9480i CT

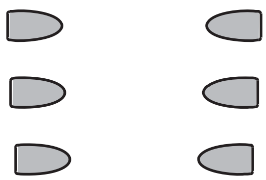
9480i and 9480i CT Phone Features

- 5 line graphical LCD screen with large backlit display
- 6 multi-functional, state-based softkeys
- Press-and-Hold speed dial key configuration feature
- 4 call appearance lines with LEDs
- Supports up to 9 call lines
- Speakerphone for handsfree calls
- Headset support (modular connector)
- Built-in-two-port, 10/100/1000 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.

9480i and 9480i CT Key Descriptions*

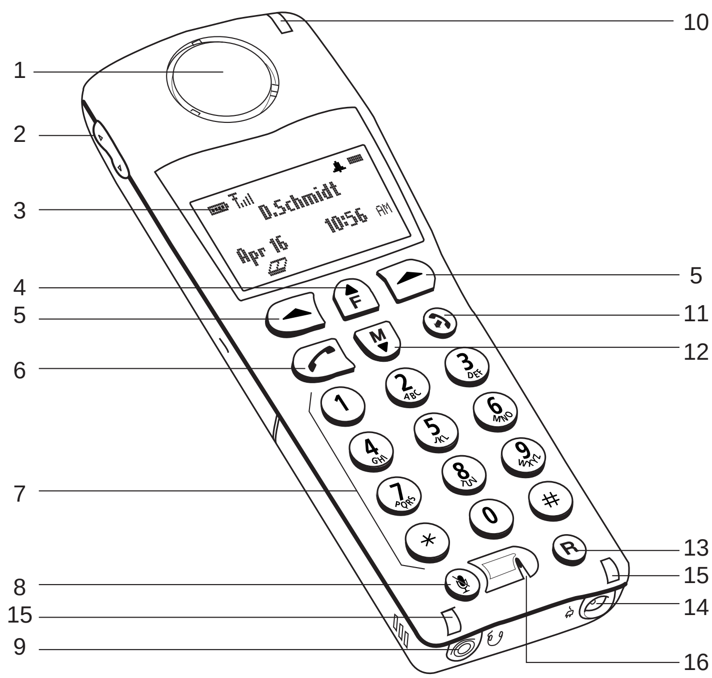
Keys	Key Description
	**Goodbye key** - Ends an active call. The Goodbye key also exits an open list, such as the Options List, without saving changes.
	**Options key** - Accesses options to customize your phone. Your System Administrator may have already customized some of your settings. Check with 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.
	**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).
	**Line/Call Appearance key** - Connects you to a line or call. The Aastra 9480i and 9480i CT IP phones support up to 4 line keys.
	**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.
	**Transfer key** - Transfers an active call to another number.
	**Conference key** - Begins a conference call with the active call.

Keys	Key Description
	**Icom key** - Begins an intercom call to a remote extension and answers incoming intercom calls. The 9480i and 9480i CT IP Phones also have default softkey 4 configured as **Icom**.
	**Services key** - Displays a list of Services available to your phone, if specific services have been configured. The available Aastra services include Directory & Callers Log. The 9480i and 9480i CT IP Phones also have default softkey 1 configured as **Services**. **Note:** Availability of the services feature is dependant on your phone system and/or service provider.
	**Softkeys** - 6 state-based softkeys on the 9480i and 9480i CT IP Phones. When you pick up the handset, the following displays on key 1: **Dial** - After entering a phone number from the keypad, you can press the Dial softkey to immediately dial the 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 5-130.

*See the Aastra 9480i and 9480i CT User Guides for more information about each of these keys.

9480i 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



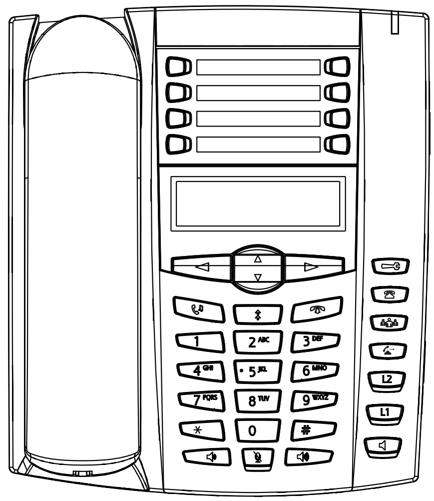
9480i CT Cordless Handset Key Descriptions*

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
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

*See the Aastra 9480i CT User Guide for more information about each of these keys.

Model 6730i IP Phone

This section provides brief information about the Model 6730i IP Phone. It includes a list of features and describes the hard keys on the 6730i. The 6730i is available with a symbol keypad or a text keypad.



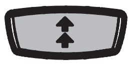
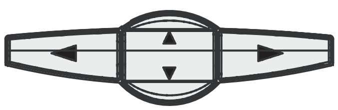
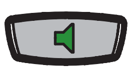
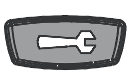
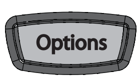
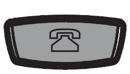
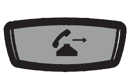
6730i with Symbol or Text Keys

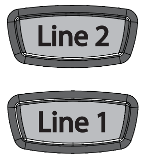
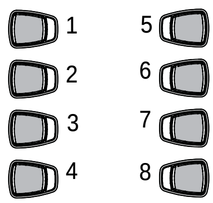
6730i Phone Features (Symbol and Text)

- 3-line LCD screen
- 8 programmable top keys
- Press-and-hold speed dial key configuration feature
- Supports up to 6 call lines with LEDs
- Full-duplex speakerphone for handsfree calls
- Headset mode support (via handset jack)
- AC power adapter (included)
- Enhanced busy lamp fields*
- Set paging*

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

6730i Symbol and Text Key Descriptions*.

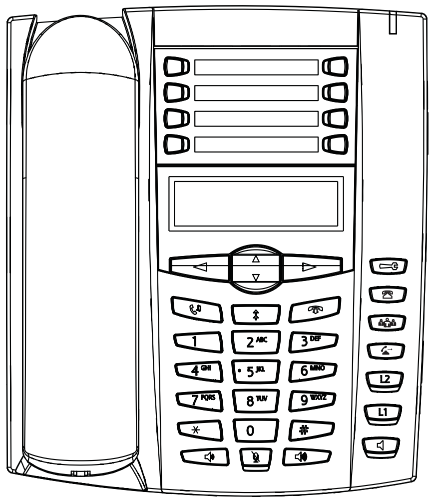
Symbol Keys	Text Keys	Key Description
		**Hold key** - Places an active call on hold. To retrieve a held call, press the call appearance button beside the light that is flashing.
		**Redial key** - Redials up to 100 previously dialed numbers. Pressing the Redial key twice redials the last dialed number.
		**Goodbye key** - Ends an active call. The Goodbye key also exits an open list, such as the Options List, without saving changes.
		**Volume control key** - Adjusts the volume for the handset, ringer, and handsfree speaker.
		**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.
		**Speaker key** - Transfers the active call to the speaker, allowing handsfree use of the phone. Switched between headset and speaker depending on audio mode setting.
		**Options key** - Accesses services and options to customize your phone. Your System Administrator may have already customized some of your settings.
		**Callers List key** - Accesses the last 200 calls received.
		**Conference key** - Begins a conference call with the active call.
		**Transfer key** - Transfers the active call to another number.

Symbol Keys	Text Keys	Key Description
		**Line/Call Appearance keys** - Connect you to a line or call. The Aastra 6730i IP phone supports 2 line keys, each with LED indicator lights. Additional lines (up to 6 in total) can be added to the programmable keys.
		**Programmable keys** - 8 Top Keys - all 8 keys are programmable. Keys 5 and 6 are designated as the SAVE and DELETE keys, respectively. These keys must be made configurable by the System Administrator before they can be changed. The following are the default functions for the programmable keys on the 6730i IP phone: **1** - **None** **2** - **None** **3** - **None** **4** - **None** **5** - **SAVE** - Allows you to save numbers (preconfigured)and/or names to the Directory. Using this key, you enter the number, name, and line (or speed dial key) to record in the Directory List. **6** - **DELETE** - Allows you to delete entries (preconfigured)from the Directory List and Callers List. (Must enter the Directory or Callers list and select an entry, then press twice to delete entry). **7** - **DIRECTORY** - Displays up to 200 names (preconfigured)and phone numbers (stored in alphabetical order) **8** - **SERVICES** - Accesses enhanced features (preconfigured)and services through the Services menu. **Note:** For more information about configuring the programmable keys 1 through 8 to perform specific functions, see Chapter 5, "Configuring Operational Features" the section, "Softkeys/Programmable Keys/ Feature Keys/Expansion Module Keys" on page 5-130.

*See the Aastra 6730i User Guide for more information about each of these keys.

Model 6731i IP Phone

This section provides brief information about the Model 6731i IP Phone. It includes a list of features and describes the hard keys on the 6731i. The 6731i is available with a symbol keypad or a text keypad.



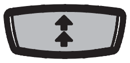
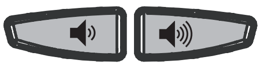
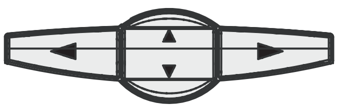
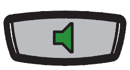
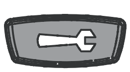
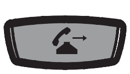
6731i with Symbol or Text Keys

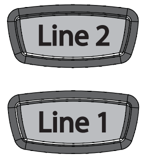
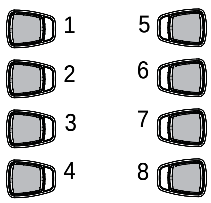
6731i Phone Features (Symbol and Text)

- 3-line LCD screen
- 8 programmable top keys
- Press-and-hold speed dial key configuration feature
- Supports up to 6 call lines with LEDs
- Full-duplex speakerphone for handsfree calls
- Headset mode support (via handset jack)
- Built-in two-port, 10/100/1000 Ethernet ports - lets you share a connection with your computer
- Inline power support (based on 802.3af standard) which eliminates power adapters
- AC power adapter (optional equipment - not included)
- Enhanced busy lamp fields*
- Set paging*

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

6731i Symbol and Text Key Descriptions*

Symbol Keys	Text Keys	Key Description
	Hold	**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	**Redial key** - Redials up to 100 previously dialed numbers. Pressing the Redial key twice redials the last dialed number.
	Goodbye	**Goodbye key** - Ends an active call. The Goodbye key also exits an open list, such as the Options List, without saving changes.
		**Volume control key** - Adjusts the volume for the handset, ringer, and handsfree speaker.
		**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.
	Speaker	**Speaker key** - Transfers the active call to the speaker, allowing handsfree use of the phone. Switched between headset and speaker depending on audio mode setting.
	Options	**Options key** - Accesses services and options to customize your phone. Your System Administrator may have already customized some of your settings.
	Callers	**Callers List key** - Accesses the last 200 calls received.
	Conf	**Conference key** - Begins a conference call with the active call.
	Xfer	**Transfer key** - Transfers the active call to another number.

Symbol Keys	Text Keys	Key Description
		**Line/Call Appearance keys** - Connect you to a line or call. The Aastra 6730i IP phone supports 2 line keys, each with LED indicator lights. Additional lines (up to 6 in total) can be added to the programmable keys.
		**Programmable keys** - 8 Top Keys - all 8 keys are programmable. Keys 5 and 6 are designated as the SAVE and DELETE keys, respectively. These keys must be made configurable by the System Administrator before they can be changed. The following are the default functions for the programmable keys on the 6730i IP phone: **1** - **None** **2** - **None** **3** - **None** **4** - **None** **5** - **SAVE** - Allows you to save numbers (preconfigured)and/or names to the Directory. Using this key, you enter the number, name, and line (or speed dial key) to record in the Directory List. **6** - **DELETE** - Allows you to delete entries (preconfigured)from the Directory List and Callers List. (Must enter the Directory or Callers list and select an entry, then press twice to delete entry). **7** - **DIRECTORY** - Displays up to 200 names (preconfigured)and phone numbers (stored in alphabetical order) **8** - **SERVICES** - Accesses enhanced features (preconfigured)and services through the Services menu. **Note:** For more information about configuring the programmable keys 1 through 8 to perform specific functions, see Chapter 5, "Configuring Operational Features" the section, "Softkeys/Programmable Keys/Feature Keys/Expansion Module Keys" on page 5-130.

*See the Aastra 6731i User Guide for more information about each of these keys.

Model 6739i IP Phone

This section provides brief information about the Model 6391i IP Phone. It includes a list of features and describes the hard keys on the 6739i. The 6739i is available with a symbol keypad or a text keypad.

6739i with Symbol or Text Keys

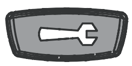
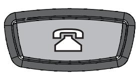
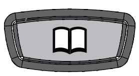
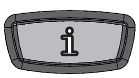
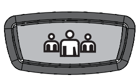
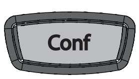
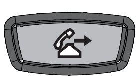
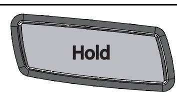
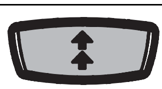
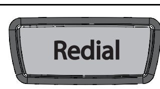
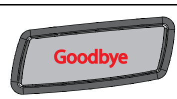
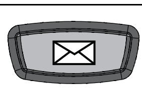
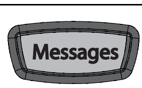
6739i Phone Features (Symbol and Text)

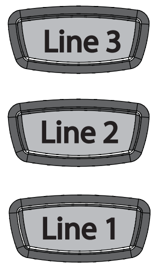
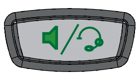
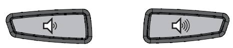
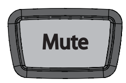
- Large 5.7” full VGA (640x480) color touch screen display and backlight
- Advanced and expandable Executive Level SIP Phone
- Intuitive graphical user interface and navigation menus
- Two built-in 10/100/1000 Gigabit Ethernet switch ports - lets you share a connection with your computer.
- Inline power support (based on 802.3af Power-over-Ethernet (PoE) standard) which eliminates power adapters
- Built-in Bluetooth technology for headset support
- Existing 675xi Expansion Module support
- USB port support (for future use)
- Aastra Hi-Q Audio™ Technology
- Full-duplex speakerphone for handsfree calls
- Additional headset connection options: modular RJ jack, built-in EHS/DHSG port (refer to the *IP Phone 6739i Installation Guide* for information on installing a DHSG headset on your phone.)
- Up to 9 lines with 3 call appearance lines with multi-proxy support
- Up to 55 programmable softkeys

- Picture ID feature (during calls and in the Directory, Callers List, and Redial List)
- XML support for productivity-enhancing applications
- AC power adapter (sold separately)

6739i Symbol and Text Key Descriptions*

The following table identifies the keys on the key panel of your 6739i IP phone that you can use for handling calls. Your phone may contain symbol keys or text keys, depending on which Model 6739i phone you purchased.

Symbol Keys	Text Keys	Key Description
		**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.
		**Callers List Key** - Accesses a list of calls received by the phone.
		**Directory Key** - Accesses a directory of names and phone numbers (stored in alphabetical order).
		**Services Key** - Accesses enhanced features and services set up by your System Administrator.
		**Conference Key** - Begins a conference call with the active call.
		**Transfer Key** - Transfers the active call to another number.
		**Hold Key** - Places an active call on hold. To retrieve a held call, press the call appearance button beside the light that is flashing or press the Hold key again.
		**Redial Key** - Redials previously dialed numbers. Also accesses a Redial List of up to 100 stored numbers that called your phone.
		**Goodbye Key** - Ends an active call. The Goodbye key also exits an open list, such as the Options List.
		**Messages Key** - Accesses your phone's voice mailbox to retrieve and listen to stored messages.

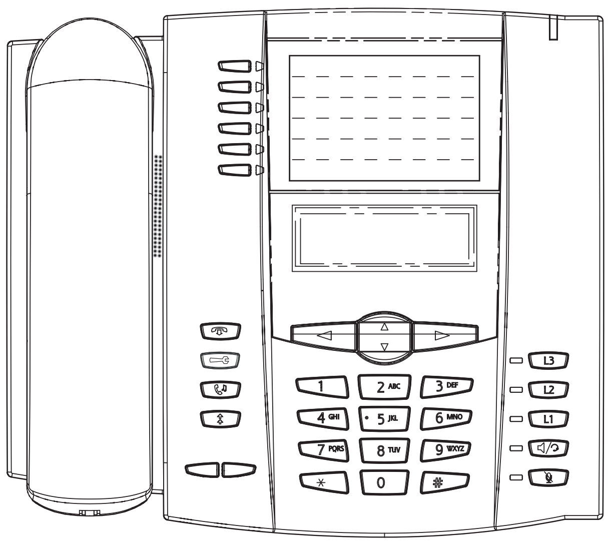
Symbol Keys	Text Keys	Key Description
		**Line/Call Appearance Keys** - Connect you to a line or call. The Aastra 6730i IP phone supports 3 line call appearance keys.
		**Speakerphone/Headset Key** - Activates handsfree for making and receiving calls without lifting the handset. When the audio mode option is set, this key is used to switch between a headset and the handsfree speakerphone.
		**Volume Control Key** - Adjusts the volume for the handset, headset, ringer, and handsfree speaker.
		**Mute key** - Mutes the microphone so that your caller cannot hear you (the light indicator flashes when the microphone is on mute).

*See the Aastra 6739i User Guide for more information about each of these keys.

Model 6753i IP Phone

This section provides brief information about the Model 6753i IP Phone. It includes a list of features, and describes the hard keys and default programmable keys on the 6753i. The 6753i is available with a symbol keypad or a text keypad.

6753i Phone Features (Symbol and Text)

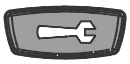
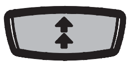
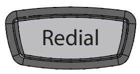
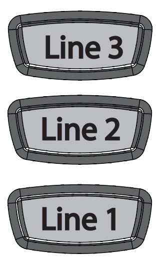
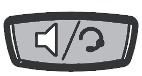
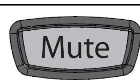
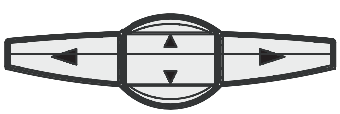


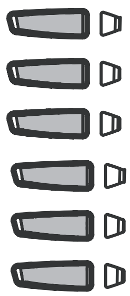
6753i with Symbol or Text Keys

- 3-line LCD screen
- 6 programmable top keys
- 3 call appearance lines with LEDs
- Press-and-Hold speed dial key configuration feature
- Supports up to 9 call lines
- Full-duplex speakerphone for handsfree calls
- Headset support (modular connector)
- Built-in two-port, 10/100/1000 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.

6753i Symbol and Text Key Descriptions*

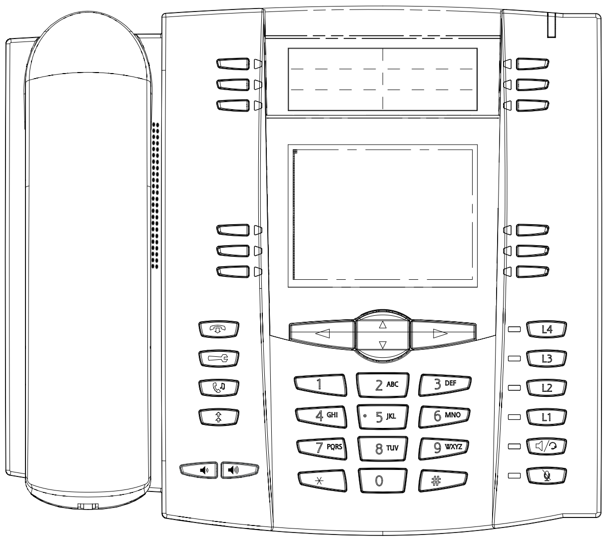
Symbol Keys	Text Keys	Key Description
		**Goodbye key** - Ends an active call. The Goodbye key also exits an open list, such as the Options List, without saving changes.
		**Options key** - Accesses options to customize your phone. Your System Administrator may have already customized some of your settings. 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 6753i 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).
		**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.

Symbol Keys	Text Keys	Key Description
		**Programmable keys** - 6 Top Keys - all 6 keys are programmable. Keys 1 and 2 are designated as the SAVE and DELETE keys, respectively. These keys are locked but can be unlocked and made configurable if required by the System Administrator. The following are the default functions for the programmable keys on the 6753i IP phone: **1** - **SAVE** Allows you to save numbers and/or names to theDirectory. Using this key, you enter the number, name, and line (or speed dial key) to record in the Directory List. **2** - **DELETE** Allows you to delete a single entry or all entries fromthe 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. **Notes:** **1.** For more information about programming the SAVE and DELETE keys, see "Locking/Unlocking the SAVE and DELETE keys (6753i)" on page 5-56. **2.** For more information about configuring keys 1 thru 6 to perform specific functions, see Chapter 5, "Configuring Operational Features" the section, "Softkeys/Programmable Keys/Feature Keys/Expansion Module Keys" on page 5-130.

*See the Aastra 6753i User Guide for more information about each of these keys.

Model 6755i IP Phone

This section provides brief information about the Model 6755i IP Phone. It includes a list of features, and describes the hard keys, default programmable keys, and default softkeys on the 6755i. The 6755i is available with a symbol keypad or a text keypad.



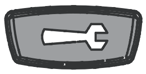
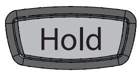
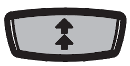
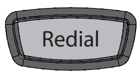
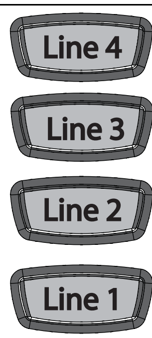
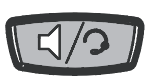
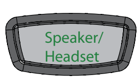
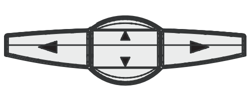
6755i with Symbol or Text Keys

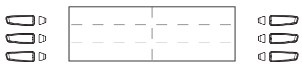
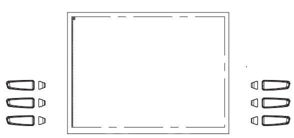
6755i Phone Features (Symbol and Text)

- 8 line graphical LCD screen (144 x 75 pixels) with white backlight
- 12 programmable keys
  - — 6 programmable hard keys on the top
  - — 6 programmable state-based softkeys on the bottom
- 4 call appearance lines with LEDs
- Press-and-Hold speed dial key configuration feature
- Supports up to 9 call lines
- Full-duplex speakerphone for handsfree calls
- Headset support (modular connector)
- Built-in-two-port, 10/100/1000 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.

6755i Key Descriptions*

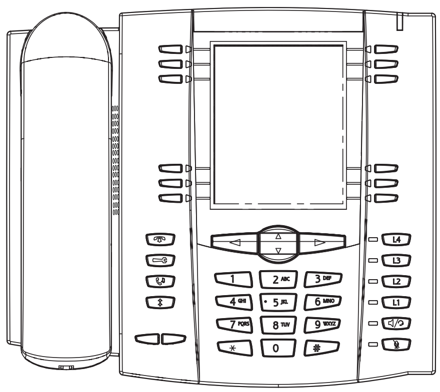
Symbol Keys	Text Keys	Key Description
		**Goodbye key** - Ends an active call. The Goodbye key also exits an open list, such as the Options List, without saving changes.
		**Options key** - Accesses options to customize your phone. Your System Administrator may have already customized some of your settings. 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 6755i IP phone supports up to 4 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).
		**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.

Symbol Keys	Text Keys	Key Description
		**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 6755i 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 5-130.
		**Softkeys** - 6 Bottom keys: programmable state-based softkeys (up to 20 programmable functions). By default, the bottom softkeys 2 through 6 have no assigned functions. You can configure all 6 bottom softkeys to perform specific functions on the 6755i 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 5-130.

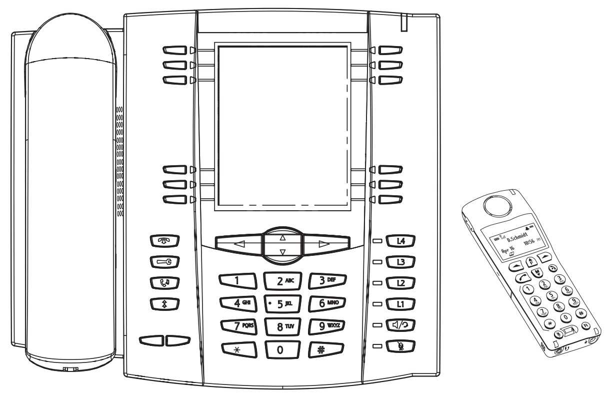
*See the Aastra 6755i User Gu 90 ide for more information about each of these keys.

Model 6757i and 6757i CT IP Phones

This section provides brief information about the 6757i IP Phone. It includes a list of features, and describes the hard keys and default softkeys on the 6757i.The 6757i is available with a symbol keypad or a text keypad.



6757i with Symbol or Text Keys



6757i CTwith Symbol or Text Keys

6757i and 6757i CT Phone Features (Symbol and Text)

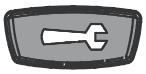
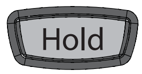
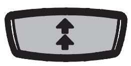
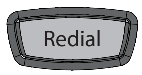
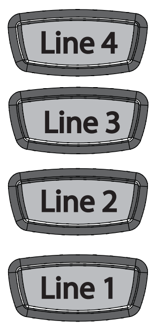
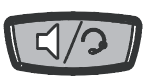
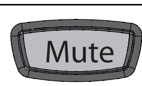
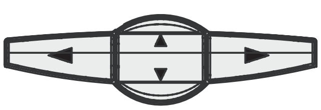
- 11 line graphical LCD screen (144 x 128 pixels) with white backlight
- 12 multi-functional softkeys

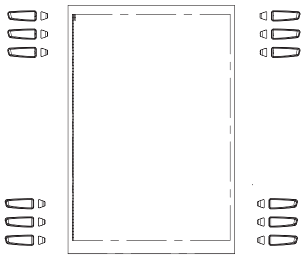
  - 6 Top Keys: programmable static softkeys

  - 6 Bottom Keys: programmable state-based softkeys
- 4 call appearance lines with LEDs
- Press-and-Hold speed dial key configuration feature
- Supports up to 9 call lines
- Full-duplex speakerphone for handsfree calls
- Headset support (modular connector)
- Built-in-two-port, 10/100/1000 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.

6757i and 6757i CT Key Descriptions*

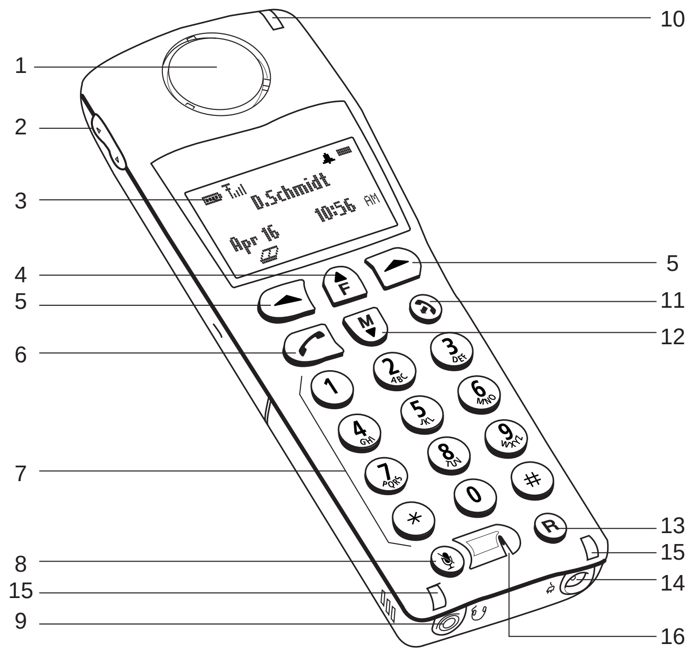
Symbol Keys	Text Keys	Key Description
		**Goodbye key** - Ends an active call. The Goodbye key also exits an open list, such as the Options List, without saving changes.
		**Options key** - Accesses options to customize your phone. Your System Administrator may have already customized some of your settings. 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 6757i IP phone supports up to 4 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).
		**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.

Symbol Keys	Text Keys	Key Description
		**Softkeys** - 12 softkeys on the 6757i 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 6757i 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. **7 - NONE** No assigned function. By default, the bottom softkeys 8 through 12 have no assigned functions. You can configure all 6 bottom softkeys to perform specific functions on the 6757i 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 5-130.

*See the Aastra 6757i or 6757i CT User Guide for more information about each of these keys.

6757i 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



6757i 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
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 User Interface via the keypad (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 4-23 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 IP phone.
- 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 power adapter required.)
  - For Ethernet networks that DO NOT supply power to the phone:
    - For power, use the Power Adapter 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. A PoE power injector is available as an optional accessory from Aastra Telecom. Contact your Reseller for more information.

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

Reference

To set the protocol for your configuration server, see Chapter 4, “Configuring Network and Session Initiation Protocol (SIP) Features”, the section, “Configuring the Configuration Server Protocol” on page 4-104.

To update the firmware on your phone, see Chapter 8, “Upgrading the Firmware”.

Firmware and Configuration Files

Description

By default on startup, the phone downloada its firmware and configuration files from the configuration server you have set; or you can manually download the firmware from the configuration server. The phone supports TFTP, FTP, HTTP and HTTPS configuration servers.

> **Note:** Automatic download is dependant on your configuration server setup. For more information about manual and automatic download of firmware, see Chapter 8, "Upgrading the Firmware." For more information on changing the download protocol on your phone, see Chapter 4, the section, "Configuring the Configuration Server Protocol" on page 4-104.

The IP Phone firmware file (*.st*) include all the necessary files you need for your phone.

The firmware consists of the following file:

- ***<phone model>.st*** - This file contains information about the specific IP Phone model and contains the language packs to load to the phone.

The configuration files consist of two files called:

- ***aastra.cf*g** - This file contains configuration information about the IP Phone.
- ***<MAC>.cfg*** *(*for example, 00085D1610DE.cfg*)* **-** This file contains configuration information about the IP Phone.

The following table provides the files that the phone requests from the configuration server during bootup of the phone:

IP Phone Model	Associated Firmware	Configuration Files	Language Files
9143i	9143i.st	aastra.cfg <MAC>.cfg (for example, 00085D1610DE.cfg)	lang_de.txt (German) lang_dk.txt (Danish) lang_es.txt (Spanish) lang_es_mx.txt (Mexican Spanish) lang_fi.txt (Finnish) lang_fr.txt (French) lang_fr_ca.txt (Canadian French) lang_it.txt (Italian) lang_no.txt (Norwegian) lang_pt.txt (Portuguese) lang_pt_br.txt (Brazillian Portuguese) lang_ru.txt (Russian) lang_sv.txt (Swedish)
9480i	9480i.st		
9480i CT	9480iCT.st		
6730i	6730i.st		
6731i	6731i.st		
6739i	6739i.st		
6753i	53i.st		
6755i	55i.st		
6757i	57i.st		
6757i CT	57iCT.st		

Reference

For more information about loading language files and using the various languages on the IP phone, see Chapter 5, the section, "Language" on page 5-40.

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 4-3.
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., 9143i.st, 9480i.st, 9480iCT.st, 6730i.st,6731i.st, 6739i.st, 51i.st, 53i.st, 55i.st, 57i.st, 57iCT.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 3-14.

Multiple Configuration Server Support

An Administrator has the option of specifying whether the phones get their firmware file, directory files, language packs, TLS certificate files, 802.1x certificate files, and HTTPS files from the original configuration server or from another server in the network. This feature allows you to specify the URL of other servers from which the phone can get this information.

Firmware Files and Multiple Configuration Servers

The firmware file for the phones can be downloaded from the original configuration server or from another server specified by a URL. You can specify a valid URL (server IP address) from which the phones get the firmware using a new parameter called, "**firmware server**" in the configuration files. If a URL is specified for this parameter, the phones in the network get the *security.tuz*, *<mac>.cfg*, and *aastra.cfg* files from the original configuration server, and the firmware files from the server specified in the URL.

➡ **Note:** The default method for the download of all files and firmware to the phones is from the original configuration server. The Administrator must **specify a correct server URL** for the phones to get their firmware information from that server. If the URL is incorrect, no firmware download occurs to the phones from the specified server.

Examples

To download all configuration and firmware files from the original configuration server:

```
firmware server:
```

Leaving this parameter blank downloads all configuration and firmware files from the original configuration server.

To download all firmware files from another specified server:

```
firmware server: tftp://10.30.102.158/test1
```

The above example uses TFTP to download all firmware files that exist in the "test1" directory on the specified server, to the phones.

➡ **Note:** Specifying the download of a "**.st**" file is not supported. For example, the following filename should NEVER be entered as a value string for the "**firmware server**" parameter:

firmware server: tftp://10.30.102.158/test1/**57i.st**

Specifying a Server to Download Firmware Files

You can use the following parameter to specify a server other than the original configuration server from which the phones get their firmware:

- **firmware server.**

Configuration Files

For the specific parameter you can set in the configuration files, see Appendix A, the section, "Multiple Configuration Server Settings" on page A-29.

Directory Files, Language Packs, TLS Certificates, 802.1x Certificates, HTTPS Files and Multiple Configuration Servers

The directory files, language packs, TLS certificate files, 802.1x certificate files, and HTTPS files can also be downloaded to the phone from a server other than the configuration server. For each of these types of files, you can specify a URL (server IP address) from which the phone gets these files. You can use existing parameters on the phone to specify the URL. For applicable parameters, see "Specifying a Server Using Existing Parameters on the IP Phones" on page 1-40.

The following table specifies the files that the original configuration server downloads, and the files that another server can download to the phone.

Files always downloaded from original configuration server are, by order:	**All files that can be downloaded from original configuration server OR another specified server are, by order:**
security.tuz *<mac>.cfg/<mac>.tuz* *aastra.cfg/aastra.tuz*	Directory Files • directory 1 • directory 2 Language Pack Files • language 1 • language 2 • language 3 • language 4 Transport Layer Security (TLS) Certificate Files • sips root and intermediate certificates • sips local certificate • sips private key • sips trusted certificates 802.1x Security Authentication Certificate Files • 802.1x root and intermediate certificates: • 802.1x local certificate: • 802.1x trusted certificates HTTPS Files • https user certificates

Specifying a Server Using Existing Parameters on the IP Phones

The following table provides the parameters on the phone that you can use to download directory files, language packs, TLS certificates, 802.1x certificates, and HTTPS files from the original configuration server OR from another server in the network.

Type of File	Parameters that support the Multiple Configuration Server feature are:
Directory Files	directory 1: directory 2:
Language Pack Files	language 1: language 2: language 3: language 4: Valid files names you can specify for languages are: lang_de.txt (German) lang_dk.txt (Danish) lang_es.txt (Spanish) lang_es_mx.txt (Mexican Spanish) lang_fi.txt (Finnish) lang_fr.txt (French) lang_fr_ca.txt (Canadian French) lang_it.txt (Italian) lang_no.txt (Norwegian) lang_pt.txt (Portuguese) lang_pt_br.txt (Brazillian Portuguese) lang_ru.txt (Russian) lang_sv.txt (Swedish)
Transport Layer Security (TLS) Certificate Files	sips root and intermediate certificates: sips local certificate: sips private key: sips trusted certificates:
802.1x Security Authentication Certificate Files	802.1x root and intermediate certificates: 802.1x local certificate: 802.1x trusted certificates:
HTTPS Files	https user certificates

Reference

For more information on each of these parameters, refer to Appendix A, "Configuration Parameters."

Examples

- **Phone Directory Files**

The following example downloads no directory:

```
directory 1:
```

The following example downloads a company directory from the original configuration server:

```
directory 1:companylist.csv
```

The following example downloads a company directory file from the specified server in the "path" directory:

```
directory 1: tftp://10.30.102.158/path/companylist.csv
```

Note: To download a specific file, the string value MUST HAVE A FILENAME at the end of the string. For example:

directory 1: tftp://10.30.102.158/path/**companylist.csv**

where "**path**" is the directory and "**companylist.csv**" is the filename. If you do not specify a filename, the download fails.

- **Language Pack Files**

The following example downloads no language pack file:

```
language 1:
```

The following example downloads the German language pack to the phones from the original configuration server:

```
language 1: lang_de.txt
```

The following example uses FTP to download the firmware file "lang_de.txt" (German language pack) from the "path" directory on server 1.2.3.4 using port 50:

```
language 1:ftp://admin:admin!@1.2.3.4:50/path/lang_de.txt
```

- **Transport Layer Security (TLS) Certificate Files**

The following example downloads no local certificate file:

```
sips local certificate:
```

The following example downloads the local certificate file from the original configuration server.

```
sips local certificate: phonesLocalCert.pem
```

The following example uses FTP to download the firmware file "phonesLocalCert.pem" (local certificate file) from the "path" directory on server 1.2.3.4 using port 50.

```
sips local certificate:ftp://admin:admin!@1.2.3.4:50/path/phonesLocalCert.pem
```

- **802.1x Security Authentication Certificate Files**

The following example downloads no 802.1x local certificate file:

```
802.1x local certificate:
```

The following example downloads the 802.1x local certificate for the phone from the original configuration server.

```
802.1x local certificate: 8021xlocalCert.pem
```

The following example uses FTP to download the firmware file “8021xlocalCert.pem” (802.1x local certificate file) from the “path” directory on server 1.2.3.4 using port 50.

```
802.1x local certificate:ftp://admin:admin!@1.2.3.4:50/path/8021xlocalCert.pem
```

- **HTTPS User Certificate Files**

The following example downloads no HTTPS user certificate files:

```
https user certificates:
```

The following example downloads the HTTPS user certificates for the phone from the original configuration server.

```
https user certificates: trustedCerts.pem
```

The following example uses FTP to download the firmware file “user.crt.pem” (https user certificate file) from the “test1” directory on server 12.43.33.234 using port 50.

```
https user certificates: ftp://test:password@12.43.33.234:50/test1/
user.crt.pem
```

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 phone model 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: Not all parameters are available from all three 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. Access to specific features and functions are restricted to the Administrator. 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.

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 5-55.

For more information about the softkeys/programmable keys, see Chapter 5, the section, "Softkeys/Programmable Keys/Feature Keys/Expansion Module Keys" on page 5-130.

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. The Administrator must enter a password for administrator options.

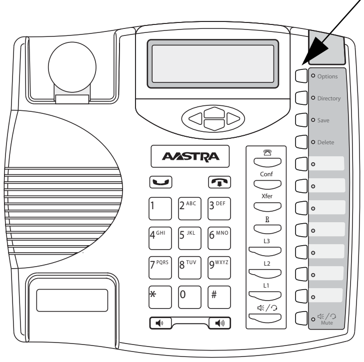
Note: An Administrator can apply a simplified options menu to the IP phones. An Administrator can also enable and disable the use of an Administrator password protection in the IP phone UI. These features are configurable using the configuration files only.

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 5-7.

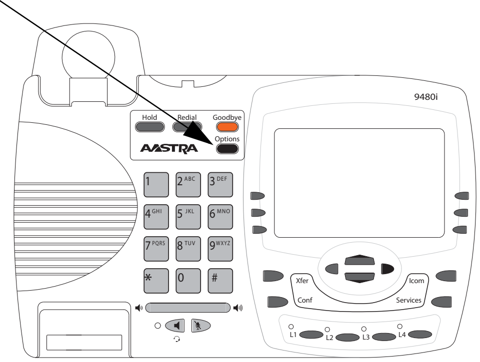
This document describes the administrator options only. For a description of the user options in the "Options List", see your model-specific *SIP IP Phone User Guide*.

The following illustrations indicate the location of the Options Key on each phone model.

Options Key

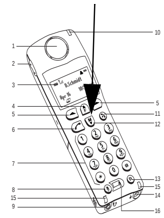


9143i

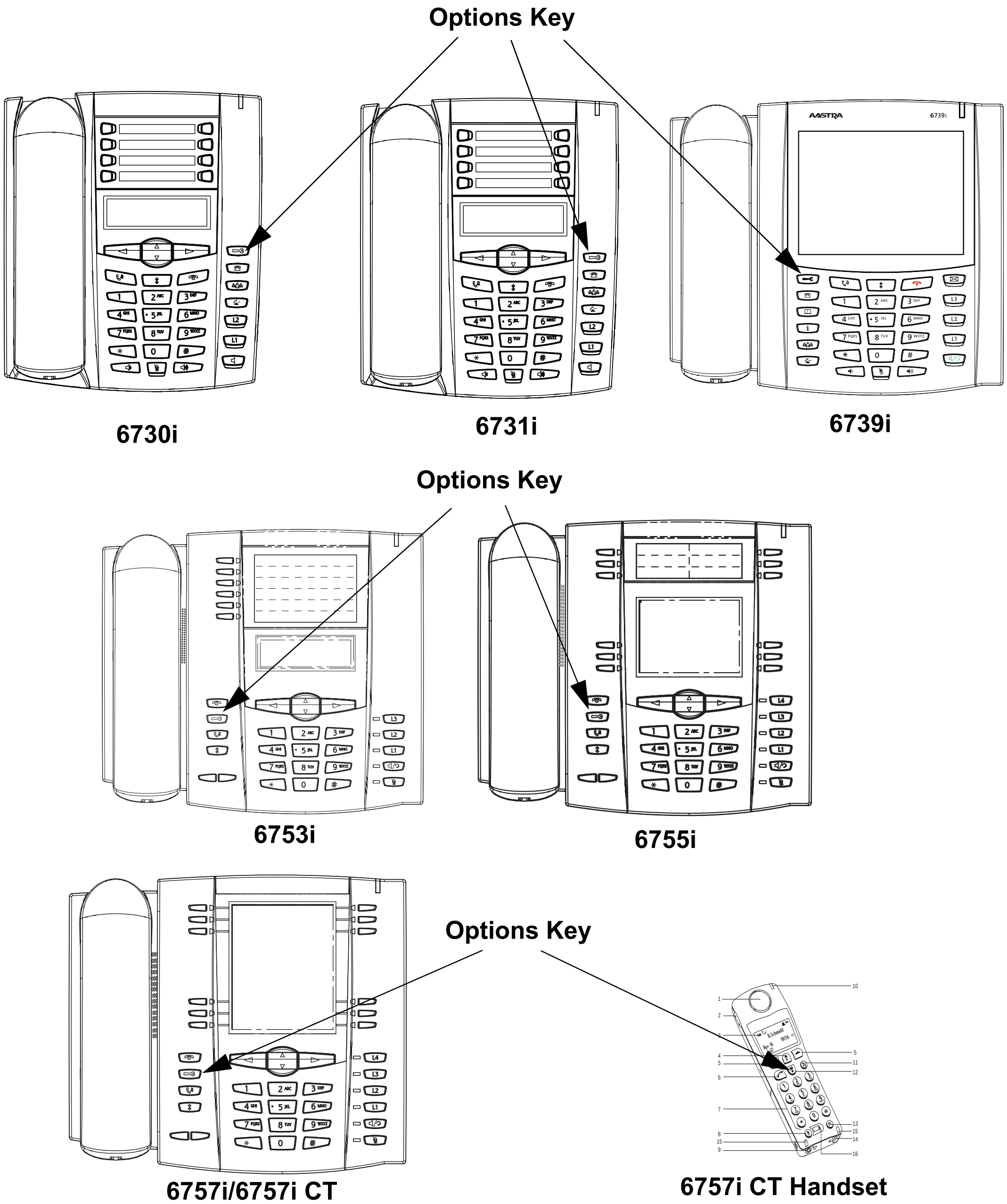


9480i/9480i CT

Options Key



9480i CT Handset


Options Key
6730i
6731i
6739i
Options Key
6753i
6755i
Options Key
6757i/6757i CT
6757i CT Handset

Using the Options Key

Step	Action
1	Press the **Options** key on the phone to enter the Options List.
2	Use the r and s to scroll through the list of options.
3	***On 3-line LCD phones:*** 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 8 and 11-line LCD phones:*** To select an option, press the **Select** softkey, press 4 , or select the number on the keypad that corresponds to the option in the Option List.
4	***On 3-line LCD phones:*** Use the **Set** softkey after making a change to an option, to save the change. ***On 8 and 11-line LCD phones:*** 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 3 , press , or press Goodbye any time to exit without saving changes.

From the CT handsets:

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

Using the Options Key on the 6739i

Step	Action
1	Press the key on the phone to enter the Options List. A list of buttons display.
2	Press an option button to display a list of additional options.
3	Press a button to display the values for a selection or to display additional options.
4	Press a value to set the option on your phone.
5	Press the to return to the previous screen.
6	Press the button or the button at any time to return to the idle screen.

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.

Note: An Administrator can enable or disable the Aastra Web UI for a single phone or all phones in a network. For more information about enabling/disabling the Aastra Web UI, see "Enabling/Disabling the Aastra Web UI" on page 2-15.

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. Both the HTTP and HTTPS port numbers are configurable using the configuration files, the IP Phone UI, the Aastra Web UI and DHCP Option 66. For more information about configuring these ports, see Chapter 4, the section, "Configuring the Configuration Server Protocol" on page 4-104.

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

Non-Blocking HTTP Connections

The IP Phones support a non-blocking HTTP connection feature. This feature allows the user to continue using the phone when there is a delay during an HTTP connection while the phone is waiting for the HTTP server to respond. This feature also allows a user to abort the connection and perform other operations on the phone (which will abort the HTTP connection automatically). A user can also abort the HTTP loading by pressing the GOODBYE key while the phone is displaying "Loading Page.......".

Note: This feature impacts only the HTTP calls triggered by a phone key (softkey or programmable key); the HTTP calls performed by action URIs are still blocking.

Authentication Support for HTTP/HTTPS Download Methods for Broadsoft Client Management System (CMS)

The IP Phones have authentication support per RFC 2617 when using HTTP or HTTPS as download protocols. If a 5i Series phone is challenged by an HTTP or HTTPS server when the server attempts to download the *aastra.cfg* file, the phone automatically sends "**aastra**" as the default Username and Password back to the server. For more information about this feature, see Chapter 5, the section, "Authentication Support for HTTP/HTTPS Download Methods, used with Broadsoft Client Management System (CMS)" on page 5-333.

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

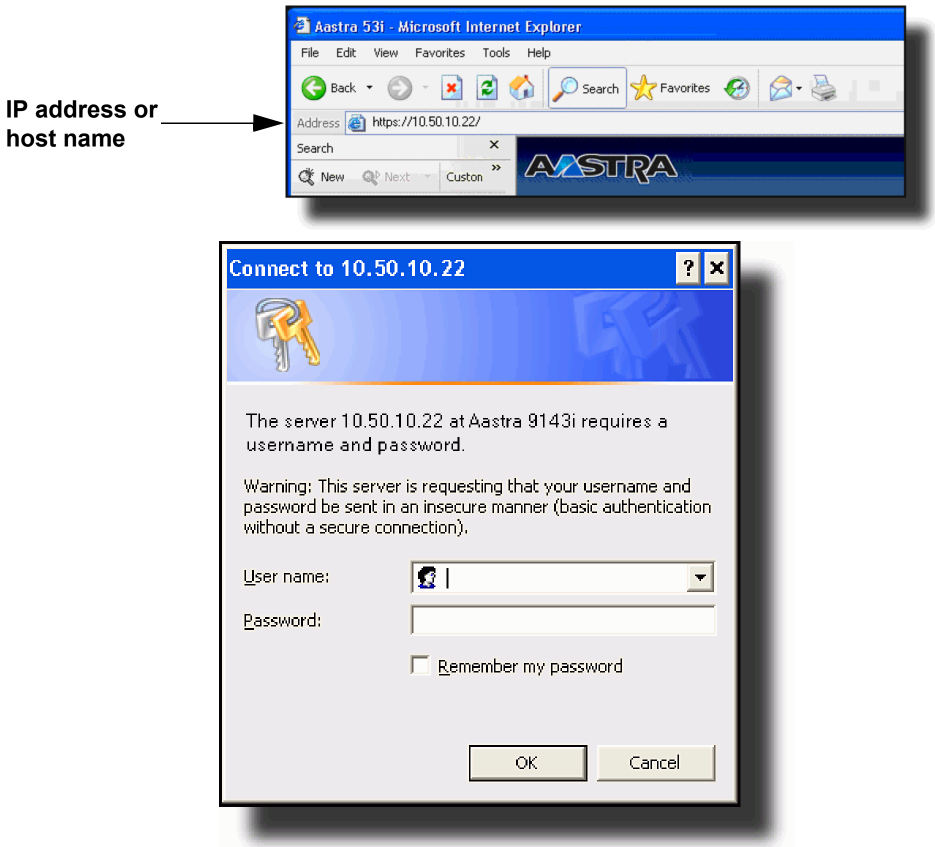
For more information on configuring the HTTPS protocol, see Chapter 4, the sections:

- "Configuring the Configuration Server Protocol" on page 4-104
- "HTTPS Client/Server Configuration" on page 4-41

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 is an example of the Login screen that displays.

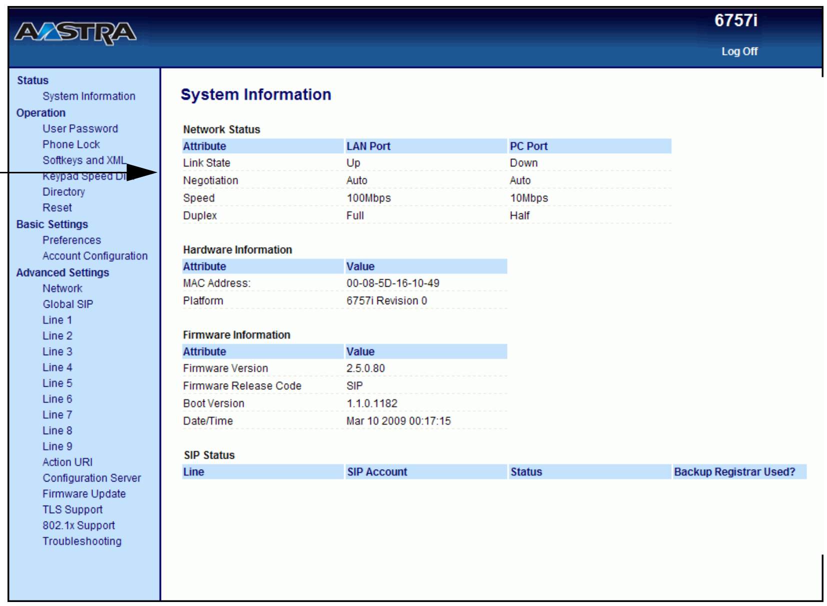


</td></tr>
</table>

<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 6757i IP phone.

6757i Network Status Window


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

Depending on the model phone you are accessing, the following categories display in the side menu of the Aastra Web UI: **Status, Operation, Basic Settings, Advanced Settings**.

> **Note:** Programmable Keys apply to the 9143i, 6730i, 6731i, 6753i, and 6755i. Softkeys apply to the 9480i, 9480i CT, 6739i, 6755i, 6757i, and 6757i CT. Expansion Modules apply to the 6739i, 6753i, 6755i, 6757i, and 6757i CT only.

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 status window also displays the SIP Account information for each account on the 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	9143i - 7 multi-functional programmable keys 6730i - 8 Top, multi-functional programmable keys 6731i - 8 Top, multi-functional programmable keys 6753i - 6 Top, multi-functional, programmable keys 6755i - 6 Top multi-functional, programmable keys (Applicable to User and Administrator)
Softkeys and XML	9480i/9480i CT 6 state-based, multi-functional softkeys 6755i 6 Bottom, state-based, multi-functional softkeys 6739i 55 state-based, multi-functional softkeys 6757i/6757i CT 6 Top, multi-functional, static softkeys 6 Bottom, state-based, multi-functional softkeys (Applicable to User and Administrator)

Heading	Description
Expansion Module <N>	The M670i has up to 36 configurable keys. The M675i 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. **Note:** Expansion Modules apply to the 6739i, 6753i, 6755i, 6757i, and 6757i CT only. (Applicable to User and Administrator)
Handset Keys (9480i CT and 6757i CT only)	Allows you to configure up to 15 softkeys on the handset. (Applicable to User and Administrator)
Keypad Speed Dial	Allows you to configure up to 9 speed dial keys. These fields map to the keypad digits 1 through 9 on the phone. You can also configure additional speed dials on the programmable keys, softkeys and expansion modules. See your model-specific User Guide for more information about this feature. (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) • Suppress DTMF Playback • Display DTMF Digits • Play Call Waiting Tone • Stuttered Dial Tone • XML Beep Support • Status Scroll Delay (seconds) • Incoming Call Interrupts Dialing • Switch Focus to Ringing Line • Call Hold Reminder During Active Calls • Call Hold Reminder • Call waiting tone period • Preferred Line • Preferred Line Timeout (seconds) • Goodbye Key Cancels Incoming Call • Message Waiting Indicator Line • DND Key Mode • Call Forward Key Mode • Use LLDP ELIN This section also allows you to set: • Outgoing Intercom Settings (Admin Only; Administrator can enable these for a User if required) • Incoming Intercom Settings • Group Paging RTP Settings • 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 (Only the Admin can specify the language pack names to load to the phone). Both the Admin and User can select the language type to display for the Web UI.
Account Configuration	Allows you to configure DND (Do Not Disturb) and/or Call Forwarding by specific account or by all accounts. Also allows you to enable/disable specific states for each account, specify different phone numbers for call forwarding, and specify number of rings for a “No Answer” state.

Advanced Settings (Applicable to Administrator Only)

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

Heading	Description
Network	Allows you to set Basic Network Settings, Advanced Network Settings, HTTPS Settings, Type of Service DSCP, and VLAN settings.
Global SIP	Allows you to set global Basic SIP Authentication Settings, Basic SIP Network Settings, Advanced SIP Settings, Real-time Transport Protocol (RTP) settings, Codec Preference List Settings, and Autodial Settings that apply to all lines on the IP phone.
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.
Action URI	Allows an administrator to specify a uniform resource identifier (URI) that triggers a GET when certain events occur. An Administrator can also specify a URI to be called, enable polling for the URI, and specify the interval between polls. (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 using any of the IP Phones supported protocols. (Applicable to Administrator Only)
TLS Support	Allows you to specify SIP Root and Intermediate Certificate files, local certificate files, private key filename, and/or trusted certificate filename to use when the phone uses the TLS transport protocol to setup a call. (Applicable to Administrator Only)

Heading	Description
802.1x Support	Allows you to enable/disable the 802.1x Protocol (Extensible Authentication Protocol (EAP)) to use on the IP phones for authentication purposes. Applicable choices are EAP-MD5 or EAP-TLS. (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. Also displays error messages if applicable. **Note:** You can also specify whether a user can upload system information automatically or manually by configuring a parameter in the configuration files. For more information on this feature, see **Chapter 9**, the section"Configuration and Crash File Retreival" on page 9-15. (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.

System Administrators can also disable Users ability to login to the Aastra Web UI. With the Aastra Web UI disabled, users will still be able to lock/unlock the phone with a PIN from the IP Phone. Administrators can disable the user Web UI using the configuration file. System Administrators have the option to either disable the Web UI for both the Administrator and User, enable for both the Administrator and User, or enable the Web UI only for the Administrator.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 for Administrators and Users. A value of **(1)** enables the Web UI for Administrators and Users. A value of (**2**) enables the Web UI for administrators only.
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 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.

Locking Parameters in the Configuration File

The IP Phones allow you to lock individual configuration parameters to prevent an end user from changing the configuration on the phone. This feature allows service providers to prevent the end-user from changing the values of specific parameters that would affect the service they provide.

An Administrator can lock parameters on the phone by placing an **exclamation mark** (**!**) before the parameter in the configuration file. For example,

```
!admin password: 22222
!emergency dial plan: 911|999
```

You can lock parameters on the phone using the configuration files only. Once the parameters are locked, they cannot be changed at all during the phones run-time. The parameters appear as read-only when accessing the Aastra Web UI and the IP Phone UI. In the Aastra Web UI, they appear grayed out. In the IP Phone UI the ability to change the parameters is removed. In addition, when parameters are locked, they cannot be changed via XML.

Notes:

1. The "parameter locking" feature applies to Release 2.4 and up. Any phones that have a previous release loaded on the phone will not be able to use the locking functionality in the configuration file.

2. Any parameter duplicated in the ***mac.cfg*** from the ***aastra.cfg*** is overwritten by the locking status and the value of the parameter found in the ***mac.cfg*** file.

Limitations

- A User possessing the Administrator password can bypass the locking of configuration server details by defaulting the phone.
- Parameters cannot be locked using XML.
- Configuration files that include locked parameters are not backwards compatible

Overwriting Parameters with Defaults in the Configuration Files

An Administrator can specify a “ **^** “ (caret character) before a configuration parameter in the *aastra.cfg* and *<mac>.cfg* configuration files which allows the parameter to be overwritten and reset back to a specified value. This can be convenient when changes are made by a user to specific parameters on the phone locally (via Aastra Web UI or IP Phone UI), and the Administrator wants to set the parameters back to the default values using the configuration files.

As an example, the following table describes how the parameter “**sip proxy ip**” is handled by the phone during phone bootup when either the “ ^ “ (default parameter) is used or the “ ! “ (locked parameter) is used.

IF	THEN
new *<mac>.cfg* file is loaded to the phone with “**^sip proxy ip**” and any other parameter(s) from the file specifying a “ **^** “	the “**^sip proxy ip**” and any other “ **^** “ parameters are overwritten if previously changed by the user.
new *aastra.cfg* file is loaded to the phone with “**^sip proxy ip**” and any other parameter(s) from the file specifying a “ **^** “	the “**^sip proxy ip**” and any other “ **^** “ parameters are overwritten if previously changed by the user.
the first instance is “**^sip proxy ip**” and second instance is “**!sip proxy ip**”in the *aastra.cfg* and/or *<mac>.cfg* file,	the value for the second instance of the parameter (“**!sip proxy ip**”) overwrites to the *aastra.cfg* and/or *<mac>.cfg files* previously on the phone.

Notes:

1. XML reboots take precedence over *server.cfg* values. Therefore, “ ^ “ parameters are ignored in the *aastra.cfg* file during XML reboots.
2. If a parameter has both a “ ^ “ and a “ ! “ preceding the same parameter (i.e., ^!sip proxy ip: pbx.company.com), then the parameter is ignored and NOT overwritten.

Example

The following example illustrates the use of the “ ^ “ in the configuration files.

aastra.cfg

```
^sip proxy ip: pbx.company.com
^sip proxy port: 5060
^sip registrar ip: pbx.company.com
^sip registrar port: 5060
```

In the above example, if an Administrator indicates the “ ^ “ before the parameters in the *aastra.cfg* file, and then loads the *aastra.cfg* file to the phone, these four parameters are reset to their default values, even if the parameters were previously changed on the phone.

Configuration Server Redundancy via DNS A Records

The phone sends a DNS query and in the DNS response, it accepts the first server IP address and contacts that server, ignoring any additional IP addresses in the response. This allows service providers to manage load balancing (via the DNS server putting different records first on each request), but does not provide redundancy.

The phones also provide support of multiple IP addresses being returned for the DNS lookup for server redundancy via multiple DNS A record entries. The phone tries to contact the first server address it receives, but if this fails, it now tries to contact the second server address, etc.

This feature supports all the download protocols (TFTP, FTP, HTTP, and HTTPS).

Notes:

1. Once the phone has failed over to a redundant server, it continues to use that server for all other server-related processes on the phone (i.e., firmware upgrades from the Web UI, boot-up process, etc.).

2. If a server fails while downloading a file(s) to the phone, the phone performs the discovery process of finding a redundant server that is available. When the boot is complete on the redundant server, the phone tries to download the file(s) again from the previous server. The check-sync process also performs the same way when a server fails.

3. The “**Skip**” softkey displays in the event of a network outage, the user can skip the configuration download and continue the boot.

4. All server failovers and failed server IP addresses are logged in the “Error Messages” page on the IP Phone UI at ***Options->Phone Status->Error Messages***.

Limitation

- In certain cases, the TFTP Protocol cannot distinguish between “server down” and “no file on server” error messages; therefore, the failover in these instances may fail.

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.

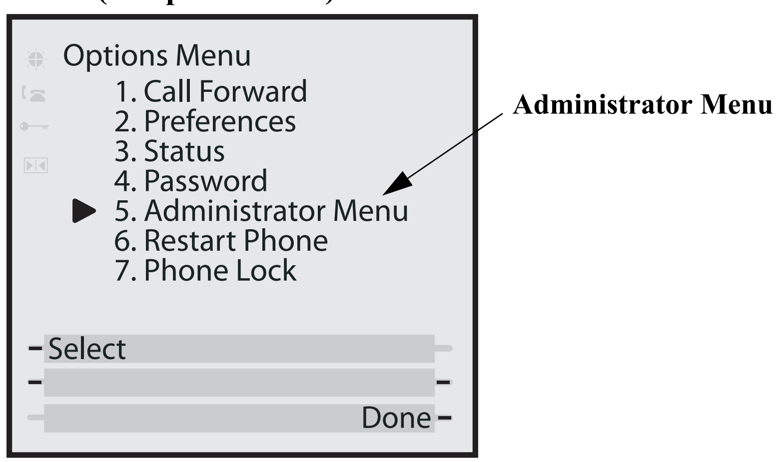
An Administrator can access and manage these options using the IP Phone UI, the Aastra Web UI, or the configuration files.

IP Phone UI Options

For all IP Phones Except the 6739i

Using the IP Phone UI, you can access the Administrator options at **Options->Administrator Menu** using the default password of "**22222**"

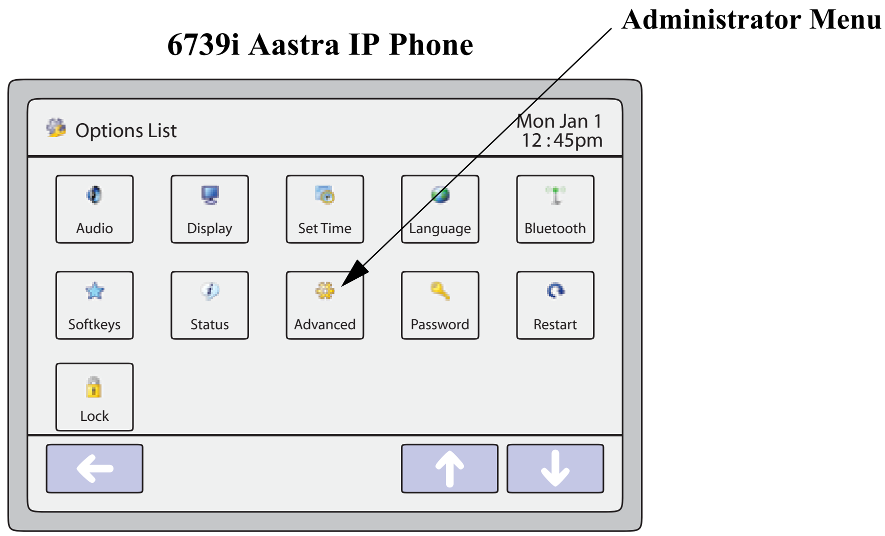
All Aastra IP Phones
(except the 6739i)



The following are administrator options in the "**Options List**" on all IP phones EXCEPT the 6739i:

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

The 6739i uses graphical icons to display the Administrator options on the 6739i phone, as shown in the following illustration.



The following are administrator options in the "**Options List**" on the 6739i:

- **Advanced Menu**
  - Configuration Server
  - SIP Settings
  - Network Settings
  - Reset (includes options for “Erase Local Config” and “Factory Default”

Note: An administrator has the option of enabling and disabling the use of password protection on the IP phone UI for all model phones. This is configurable using the configuration files only. For more information about this feature, see Appendix A, the section “Password Settings” on page A-14

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.

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

For all model phones EXCEPT the 6739i:

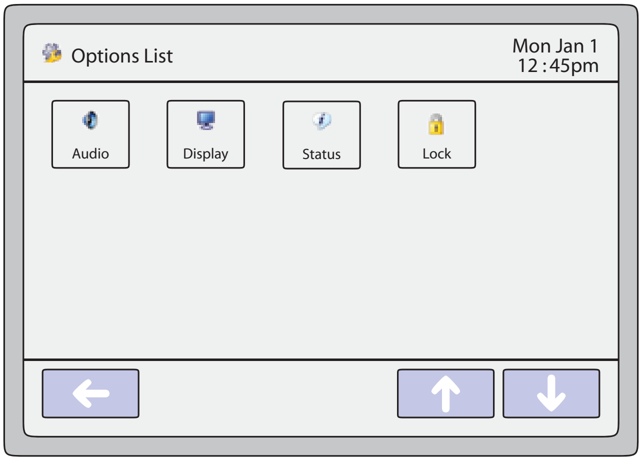
Full Options Menu	Simplified Options Menu
Call Forward	**Call Forward**
Preferences	**Preferences**
Status	**Status**
Password	Removed
Administrator Menu	Removed
Restart Phone	Removed
Phone Lock	**Phone Lock**

For the 6739i:

Full Options Menu	Simplified Options Menu
Audio	**Audio**
Display	**Display**
Set Time	Removed
Language	Removed
Bluetooth	Removed
Softkeys	Removed
Status	**Status**
Advanced	Removed
Password	Removed
Restart	Removed
Lock	**Lock**

An example of the simplified options menu on the 6739i is as follows:

6739i Simplified Options Menu



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.

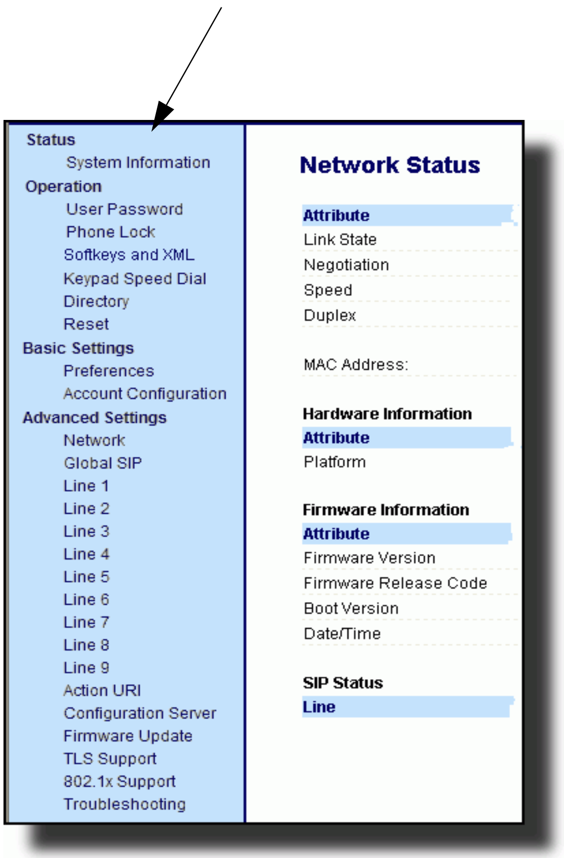
Configuration Files

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-6.

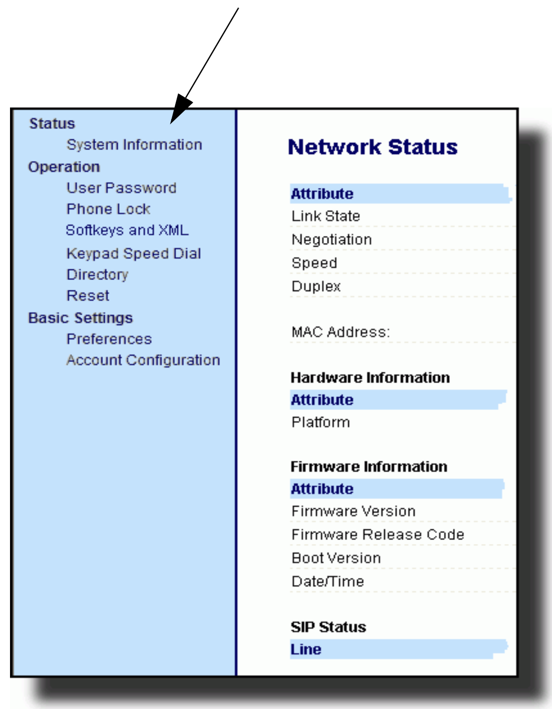
Aastra Web UI Options

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



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
  - 802.1x 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 3-14, and "Set Phone to Factory Defaults/Erase Local Configuration" on page 3-16.

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.

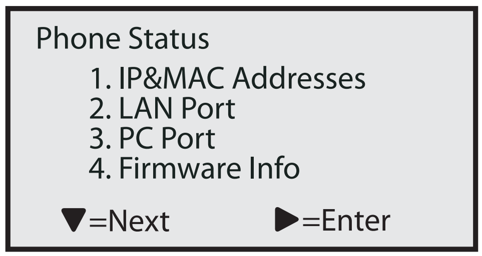
You can display phone status using the IP phone UI or the Aastra Web UI.

Phone Status via IP Phone UI

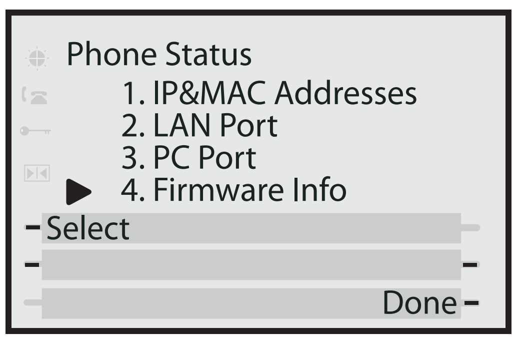
In the IP phone UI, the Phone Status options are available to the user and the administrator and do not require a password entry.

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

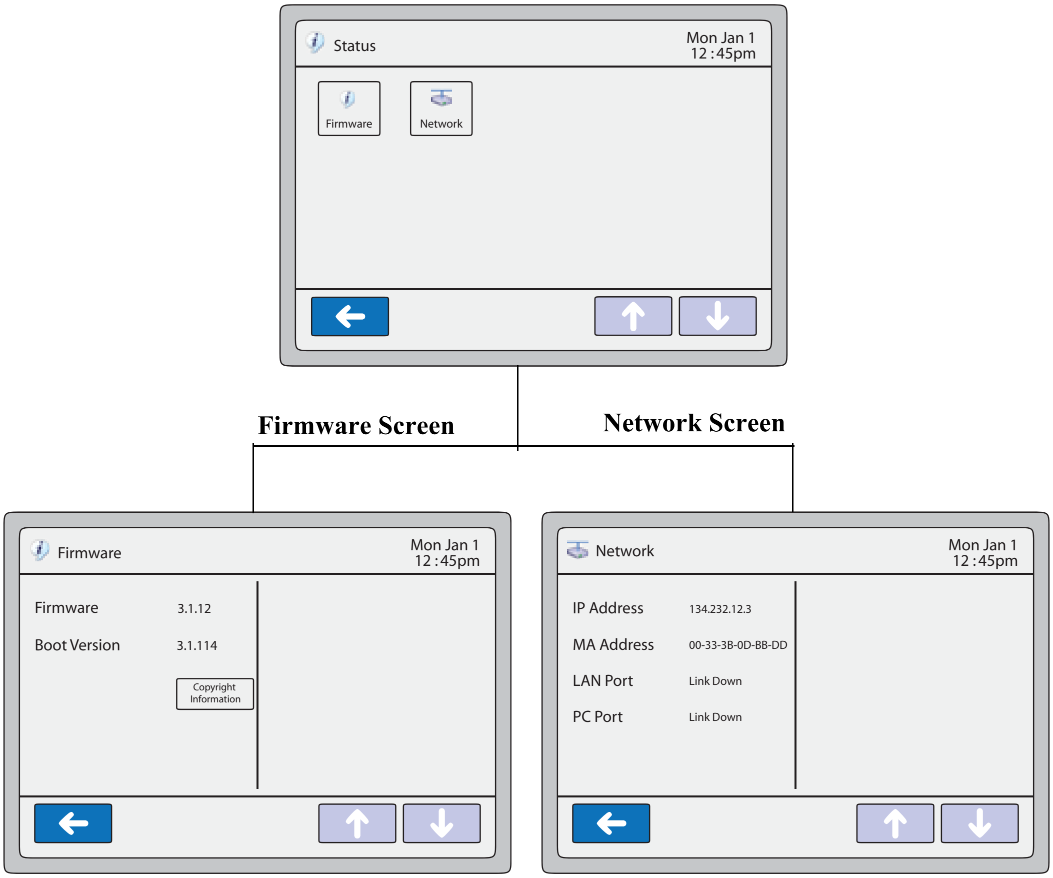
Phone Status Screen for 3-Line LCD Display Phones



Phone Status Screen for 8 and 11-Line LCD Display Phones



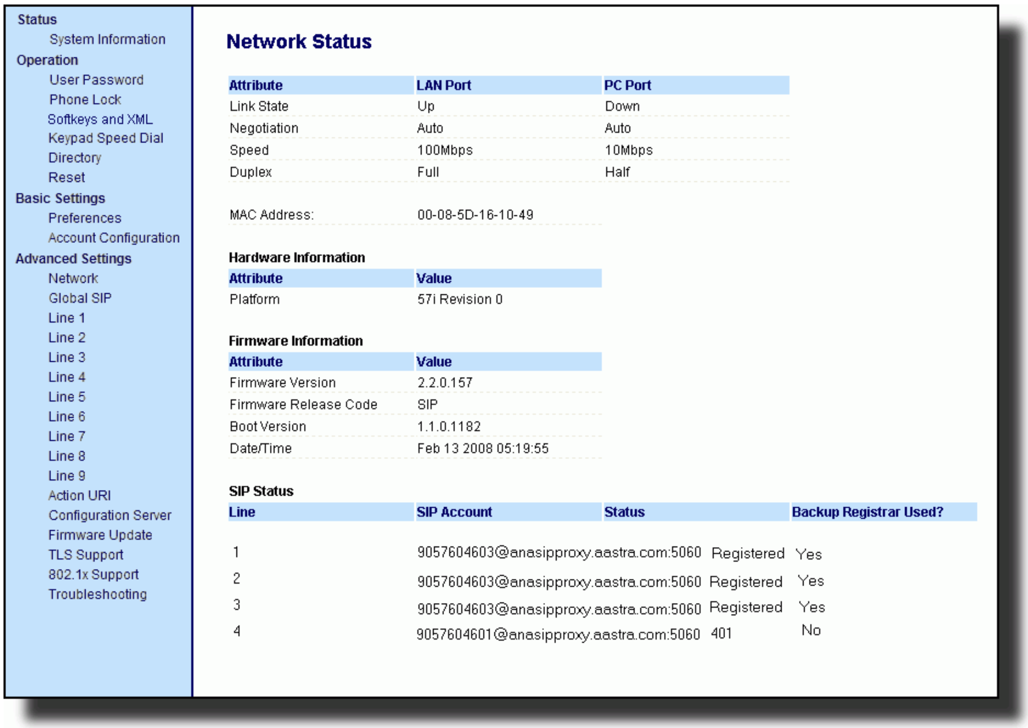
Phone Status for 6739i Phone



- **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 and Boot version that is currently installed on the IP phone.

Phone Status via Aastra Web UI

The first screen that displays after logging into the Aastra Web UI for a phone is the Network Status screen. This screen also displays when selecting **Status->System Information**. The information on this screen is available to the user and the administrator as read-only.



The following is a description of the information on the Network Status screen:

- **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.

- **SIP Status**
  Displays information about the SIP registration status of the phone. If there are accounts configured on the IP Phone, their SIP status displays in this field. All model phones display the status of up to 9 lines.

The following table describes the status conditions that can display for an account(s).

Status Condition	Description

Registered	Displays this status on accounts that HAVE been registered with the SIP proxy server.
	
	Example:
	`Line  SIP Account  Status  Backup Registrar Used?`
	`1  9057604603@anasipproxy.aastra.com:5060Registered Yes`
	
	where
	Account Number is “1”
	SIP Account is “9057604603@anasipproxy.aastra.com” on port “5060”
	Status is “Registered”
	Backup registrar is used (“Yes”)

SIP Error Number	Displays on accounts when registration fails with the SIP proxy server.
	
	Example:
	`Line  SIP Account  Status  Backup Registrar Used?`
	`4  9057604601@anasipproxy.aastra.com:5060401  No`
	
	where
	Account Number is “4”
	SIP Account is “9057604601@anasipproxy.aastra.com” on port “5060”
	Status is “401” - Unregistered if SIP registration fails.
	Backup registrar is used (“No”)

Note: The IP Phones can register with multiple server using the same user name. So the SIP Status information on the Network Status screen may display the same account with different registrar and proxy IP addresses. For more information, see

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.

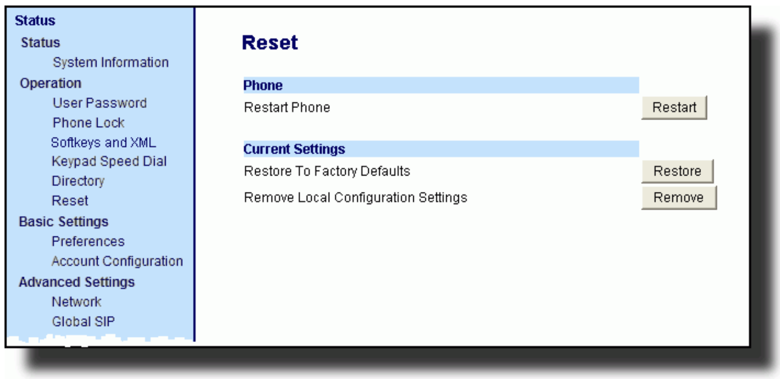
You can restart the phone using the IP Phone UI or the Aastra Web UI.

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 3-Line LCD Displays:*** Press **#** to confirm. **Note:** To cancel the Restart, press the 3 key. ***For 8 and 11-Line LCD Displays:*** Press **Restart**. **Note:** To cancel the Restart, press **Cancel**.
For the 6739i:	
1	Press the **Options** key on the phone to enter the Options List.
2	Press the **Restart** button. A “Restart the Phone?” prompt displays.
3	Press **Yes** to restart the phone, or **No** to cancel the restart function..

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

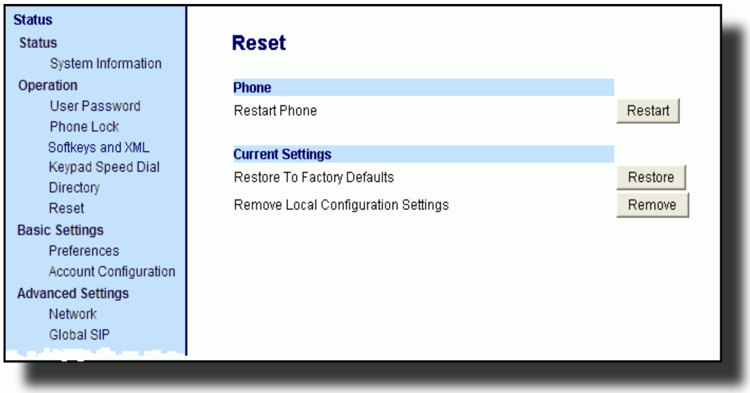
Factory default settings are the settings that reside on the phone after it has left the factory. The factory default settings on the phone sets the factory defaults for all of the settings in the *aastra.cfg*, *<mac>.cfg,* and local configuration. Performing this action results in losing all user-modified 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 the **Options** key 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 3-Line LCD Displays::*** The “*Restore Defaults?*” prompt displays. Press **#** to confirm. ***For 8 and 11-Line LCD Phones:*** The “*Reset phone to factory defaults?*” prompt displays. Press **Default** to confirm.
For the 6739i:	
1	Press the **Options** key on the phone to enter the Options List.
2	Press the **Advanced** button. A keyboard displays.
3	Enter the Administrator password using the keyboard, and press <**Enter**>. Default is “**22222**”.
4	Press the **Reset** button. A “Reset Configuration?” prompt displays.
5	Press the **Factory Default** button. The phone immediately resets to factory defaults and the phone reboots.

Settings Factory Defaults Using the Aastra Web UI

Aastra Web UI

Step	Action
1	Click on **Operation->Reset->Current Settings.** 
2	In the "**Restore to Factory Defaults**" field, click  . This restores all factory defaults, and removes any saved configuration and directory list files.

Erasing the Phone's Local Configuration

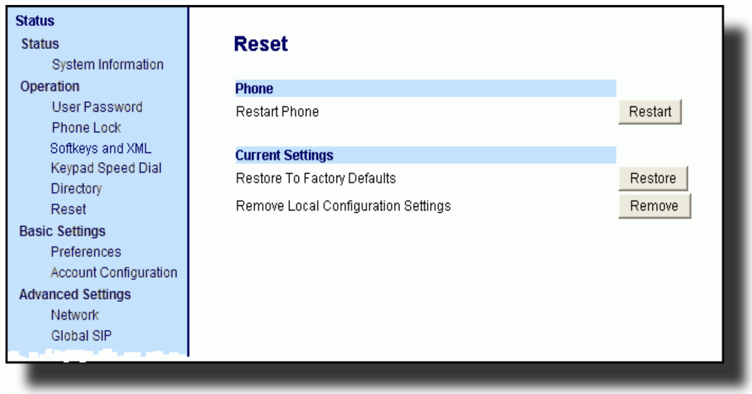
You can reset the IP Phone's local configuration if required. The local configuration is the last updated configuration you performed using the IP Phone UI or the Aastra Web UI. Performing this action results in losing all recently user-modified settings. For more information about local configuration, see Chapter 1, the section, "Configuration File Precedence" on page 1-37.

Erasing the Phone's Local Configuration 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 **Administrator Menu** and enter your Administrator Password (default is **22222**).
3	Select **Erase Local Config.**
4	***For 3-Line LCD Displays::*** The "*Erase local config?*" prompt displays. Press **#** to confirm. ***For 8 and 11-Line LCD Displays::*** The "*Erase local config?*" prompt displays. Press **Erase** to confirm.
For the 6739i:	
1	Press the **Options** key on the phone to enter the Options List.
2	Press the **Advanced** button. A keyboard displays.
3	Enter the Administrator password using the keyboard, and press <**Enter**>. Default is "**22222**".
4	Press the **Reset** button. A "Reset Configuration?" prompt displays.
5	Press the **Erase Local Cfg.** button. The phone immediately erases the local configuration on the phone and the phone reboots.

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

Aastra Web UI

Step	Action
1	Click on **Operation->Reset->Current Settings.** 
2	In the "**Remove Local Configuration Settings**" field, click  . This removes the last customized configuration settings made on the phone.

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. Dial Plan field accepts up to 512 characters. For more information on this feature, see "Local Dial Plan" on page 5-58.
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 5-60.
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 5-60.
Park Call **Note:** This option can be set by both Users and Administrators.	sip lineN park pickup config	The parking of a live call to a specific extension. **This feature on the Basic Preferences screen is available on all phones EXCEPT the 9143i, and 6753i.** To configure the Park feature on a key, see Chapter 5, the section, "Park/Pick Up Softkey" on page 5-207.
Pick Up Parked Call **Note:** This option can be set by both Users and Administrators.	sip lineN park pickup config	Picking up a parked call at the specified extension. **This feature on the Basic Preferences screen is available on all phones EXCEPT the 9143i, and 6753i.** To configure the Pickup feature on a key, see Chapter 5, the section, "Park/Pick Up Softkey" on page 5-207.
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 5-62.

Parameter in Aastra Web UI	Parameter in Configuration Files	Description
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 5-64.
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" on page 5-66.
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 Tone" on page 5-68.
Call Waiting Tone Period **Note:** This option can be set by an Administrator only.	call waiting tone period	Specifies the time period, in seconds, that the call waiting tone is audible on an active call when another call comes in. When enabled, the call waiting tone plays at regular intervals for the amount of time set for this parameter. For example, if set to "30" the call waiting tone plays every 30 seconds. When set to "0", the call waiting tone is audible only once on the active call. For more information on this feature, see "Call Waiting Tone Period" on page 5-70.
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 5-71.
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 5-72.
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 5-73.
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 5-74.

Parameter in Aastra Web UI	Parameter in Configuration Files	Description
Switch Focus to Ringing Line **Note:** This option can be set by both Users and Administrators.	switch focus to ringing line	Enables or disables whether or not the UI focus is switched to a ringing line while the phone is in the connected state. For more information on this feature, see “Switch Focus to Ringing Line” on page 5-77.
Call Hold Reminder During Active Calls **Note:** This option can be set by both Users and Administrators.	call hold reminder during active calls	Enables or disables the ability for the phone to initiate a continuous reminder tone on the active call when another call is on hold. When this feature is disabled, a ring splash is heard when the active call hangs up and there is still a call on hold. For more information on this feature, see “Call Hold Reminder During Active Calls” on page 5-79.
Call Hold Reminder **Note:** This option can be set by both Users and Administrators.	call hold reminder	Enables or disables the reminder ring splash timer to start as soon as you put a call on hold (even when no other calls are active on the phone). When enabled, the phone initiates a reminder ring splash periodically for the single call on hold. When disabled, no reminder ring splash is audible. For more information on this feature, see “Call Hold Reminder (on single hold)” on page 5-81.
	call hold reminder timer **Note:** This option can be set by an Administrator only.	Specifies the time delay, in seconds, that a ring splash is heard on an active call when another call was placed on hold. For example, if a call comes into Line 1, and then a call comes into Line 2 and you answer Line 2, Line 1 is automatically placed on hold. While on the active Line 2, after 7 seconds, a ring splash audio sounds on the line reminding you that the call on Line 1 is still on hold. This timer begins to increment after Line 2 is answered. **Notes**: **1.** This parameter is used with the “call hold reminder frequency” parameter. **2.** You must enable this “call hold reminder timer” parameter for it to work. **3.** A value of “0” disables the call hold reminder feature. For more information on this feature, see “Call Hold Reminder Timer & Frequency” on page 5-83.
	call hold reminder frequency **Note:** This option can be set by an Administrator only.	Specifies the time interval, in seconds, between each ring splash sound on the active line. For example, if a call comes into Line 1, and then a call comes into Line 2 and you answer Line 2, Line 1 is automatically placed on hold. While on the active Line 2, after 7 seconds, a ring splash audio sounds on the line reminding you that the call on Line 1 is still on hold (determined by the “call hold reminder timer” parameter), and then the ring splash is heard again after 60 seconds (determined by this parameter). **Notes**: **1.** You must enable the “call hold reminder” and/or “call hold reminder during active calls” parameter(s), and the “call hold reminder timer” parameter, for this parameter to work. **2.** A value of “0” prevents additional rings. For more information on this feature, see “Call Hold Reminder Timer & Frequency” on page 5-83.

Parameter in Aastra Web UI	Parameter in Configuration Files	Description
Preferred Line **Note:** This option can be set by both Users and Administrators.	preferred line	Specifies the preferred line to switch focus back to when incoming or outgoing calls end on the phone. For more information on this feature, see "Preferred Line and Preferred Line Timeout" on page 5-84.
Preferred Line Timeout (seconds) **Note:** This option can be set by both Users and Administrators.	preferred line timeout	Specifies the time, in seconds, that the phone switches back to the preferred line after a call (incoming or outgoing) ends on the phone, or after a duration of inactivity on an active line. For more information on this feature, see "Preferred Line and Preferred Line Timeout" on page 5-84.
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 5-87.
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 5-90.
DND Key Mode **Note:** This option can be set by both Users and Administrators.	dnd key mode	Allows you to configure the DND mode to use on the phone (Account, Phone, Custom) when the DND key is pressed. You can configure DND for all accounts or a specific account. For more information on this feature, see "DND Key Mode" on page 5-92. Also see Chapter 5, the section, "Do Not Disturb (DND)" on page 5-182.
Call Forward Key Mode **Note:** This option can be set by both Users and Administrators.	call forward key mode	Allows you to configure the Call Forward mode to use on the phone (Account, Phone, or Custom). You can configure Call Forward for all accounts or a specific account. For more information on this feature, see "Call Forward Mode" on page 5-94. Also see Chapter 5, the section, "Call Forwarding" on page 5-218.
Use LLDP ELIN **Note:** This option can be set by both Users and Administrators.	use lldp elin	Enables or disables the use of an Emergency Location Identification Number (ELIN) received from LLDP as a caller ID for emergency numbers. **Caution:** In Release 2.3 and later, LLDP is enabled by default. If LLDP is enabled on your network, the phones may come up with different network settings. For more information on this feature, see "Link Layer Discovery Protocol for Media Endpoint Devices (LLDP-MED) and Emergency Location Identification Number (ELIN)" on page 5-97.

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 5-101.
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 5-101.
Play Warning Tone **Note:** This option can be set by both Users and Administrators.	sip intercom 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 5-101.
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 5-101.
Outgoing Intercom Settings (8 and 11-Line LCD phones)		
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 5-101.

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 5-101.
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 5-101.

Group Paging RTP Settings

Parameter in Aastra Web UI	Parameter in Configuration Files	Description
Paging listen addresses **Note:** This option can be set by both Users and Administrators.	paging group listening	Allows you to specify up to 5 listening multicast addresses to send/ receive a Real Time Transport Protocol (RTP) stream to/from these pre-configured multicast address(es) without involving SIP signaling. For more information on this feature, see "Group Paging RTP Settings" on page 5-105.

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 speed dial 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 speed dialing on the 9480i CT or 6757i CT Base Stations, the **Redial** key on the 9480i CT and 6757i CT handsets retain their original functionality. The **Redial** key on the handset is not configured for speed dial. For more information on this feature, see "Speed DialKey Mapping" on page 5-109.
Map Conf Key To	map conf key to	Sets the **Conf** key as a speed dial 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 speed dialing on the 9480i CT or 6757i CT Base Stations, the **Conf** key on the 9480i CT and 6757i CT handsets retain their original functionality. The **Conf** key on the handset is not configured for speed dial. For more information on this feature, see "Speed DialKey Mapping" on page 5-109.

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 5-112.
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 5-112.
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 5-112.

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 5-116.
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 5-116.
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 5-116.
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 5-116.

Parameter in Aastra Web UI	Parameter in Configuration Files	Description
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 5-116.
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 5-116.
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 5-116.
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 5-116.

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 5-122.
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 5-122.
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 5-122.

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 5-165.
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 5-165.

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 5-19.
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 5-19.
Time Zone **Note:** This option can be set by both Users and Administrators.	N/A	time zone name **Custom Parameters**: • time zone minutes • dst minutes • dst [start|end] relative date • dst start month • dst end month • dst start week • dst end week • dst start day • dst end day • dst start hour • dst end hour	This parameter allows you to set the time zone code or customize the time zone for their area as required. For more information on this feature, see "Time Zone & DST" on page 5-20.
Time Servers **Note:** This option can be set by both Users and Administrators.	NTP Time Servers	time server disabled	This parameter allows you to enable or disable the Network Time Server (NTP) to set the time on the phone. For more information on this feature, see "Time Servers" on page 5-29.

Parameter in IP Phone UI	Parameter in Aastra Web UI	Parameters in Configuration Files	Description
Time Server 1 **Note:** This option can be set by both Users and Administrators.	Time Server 1	time server1	This parameter allows you to set the IP address of Time Server 1 in dotted decimal format. For more information on this feature, see "Time Servers" on page 5-29.
Time Server 2 **Note:** This option can be set by both Users and Administrators.	Time Server 2	time server2	This parameter allows you to set the IP address of Time Server 2 in dotted decimal format. For more information on this feature, see "Time Servers" on page 5-29.
Time Server 3 **Note:** This option can be set by both Users and Administrators.	Time Server 3	time server3	This parameter allows you to set the IP address of Time Server 3 in dotted decimal format. For more information on this feature, see "Time Servers" on page 5-29.

Live Dialpad

Parameter in IP Phone UI	Parameter in Aastra Web UI	Parameters in Configuration Files	Description
Live Dialpad **Note:** This option can be set by a User via the IP Phone UI and by an Administrator via the IP Phone UI and the configuration files.	N/A	live dialpad	This parameter turns the "Live Dialpad" feature ON or OFF. For more information on this feature, see "Live Dial Pad*" on page 5-39.

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 are: 0 (English) is default 1-4 The values 1-4 are dependent on the "language N" parameter. For example, if "language 1: lang_fr.txt", then "language: 1" would set the webpage language to French. **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 5-40. For more information on specifying a language to use on the IP Phone, see "Specifying the Screen Language to Use" on page 5-43.

Parameter in Aastra Web UI	Parameter in Configuration Files	Description
Input Language **Note:** This option can be set by both Users and Administrators.	input language	Allows you to specify the language to use for inputs on the IP Phone. Entering a language value for this parameter allows users to enter text and characters in the IP Phone UI, Aastra Web UI, and in XML applications via the keypad on the phone, in the language(s) specified. Valid values are: • English • French • Français • German • Deutsch • Italian • Italiano • Spanish • Español • Portuguese • Português • Russian • Русский • Nordic For more information on this feature, see "Specifying the Input Language to Use" on page 5-46.
Language 1 thru 4	language N	The language pack you want to load to the IP phone. Valid values are: lang_de.txt (German) lang_dk.txt (Danish) lang_es.txt (Spanish) lang_es_mx.txt (Mexican Spanish) lang_fi.txt (Finnish) lang_fr.txt (French) lang_fr_ca.txt (Canadian French) lang_it.txt (Italian) lang_no.txt (Norwegian) lang_pt.txt (Portuguese) lang_pt_br.txt (Brazillian Portuguese) lang_ru.txt (Russian) lang_sv.txt (Swedish) **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 on this feature, see "Loading Language Packs" on page 5-40.

Account Configuration

The IP phones have a DND and CFWD feature that allows an Administrator and User to configure "do not disturb" and "call forwarding" by account. You can set specific modes for the way you want the phone to handle DND and CFWD. The three modes you can set on the phone for these features are:

- **Account**
- **Phone**
- **Custom**

You can set the modes for DND and CFWD in the Aastra Web UI at the path *Basic Settings->Preferences->General*, or using the following parameters in the configurations files:

- **dnd key mode**
- **call forward key mode**

The following table describes the behavior of the mode settings for DND and CFWD.

Modes	DND	CFWD
Account	Sets DND for a specific account. A pre-configured DND key toggles the account in focus on the IP Phone UI, to ON or OFF.	Sets CFWD on a per account basis. Pressing a pre-configured CFWD key applies to the account in focus
Phone	Sets DND ON for all accounts on the phone. A pre-configured DND key toggles all accounts on the phone to ON or OFF.	Sets the same CFWD configuration for all accounts (**All**, **Busy**, and/or **No Answer**). When you configure the initial account, the phone applies the configuration to all other accounts. (In the Aastra Web UI, only the account you configured is enabled. All other accounts are grayed out but set to the same configuration.) Using the Aastra Web UI, if you make changes to that initial account, the changes apply to all accounts on the phone.
Custom	Sets the phone to display custom screens after pressing a pre-configured DND key, that list the account(s) on the phone. The user can select a specific account for DND, turn DND ON for all accounts, or turn DND OFF for all accounts	Sets CFWD for a specific account or all accounts. You can configure a specific mode (**All**, **Busy**, and/or **No Answer**) for each account independently or all accounts. On the 3-line LCD phones, you can set all accounts to **ALL On** or **ALL Off**. On the 8 and 11-line LCD phones, you can set all accounts to **All On**, **All Off**, or copy the configuration for the account in focus to all other accounts using a **CopytoAll** softkey.

References

For more information about account configuration of DND and CFWD on the IP Phones, see Chapter 5, the sections:

For DND:

- "DND Key Mode" on page 5-92.
- "Do Not Disturb (DND)" on page 5-182.

For CFWD:

- "Call Forward Mode" on page 5-94.
- "Call Forwarding" on page 5-218.

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.

Notification When Incorrect Network Settings Entered

If an Administrator enters incorrect network settings while configuring the network parameters, the IP Phone UI AND the Aastra Web UI immediately notify the Administrator that an incorrect value was entered. This notification applies to the following network parameters:

- A 0.0.0.0 entered as values for the **IP Address**, **Subnet Mask**, and **Gateway** parameters
- **IP Address** and **Gateway** IP address parameter values entered exactly the same
- **Gateway** IP address and the **IP address** parameter values configured on the same subnet

If you configure the Gateway parameter and the IP Address parameter on the same subnet, the following error message displays:

"Gateway IP address and the IP address parameter values configured are not on the same subnet"

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, 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. The IP phones also support Option 60 and 43. For more information, see "DHCP" on page 4-3.
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 4-23.
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 4-23.
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 4-23.

Parameter In IP Phone UI	Parameter in Aastra Web UI	Parameters in Configuration Files	Description
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 4-23.
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 4-23.
Hostname	Hostname	hostname	Specifies the hostname DHCP Option 12 that the phone sends with the DHCP Request packet. For more information, see "Using Option 12 Hostname on the IP Phone" on page 4-10.
Ethernet LAN Port Link PC Port Link PC Port Enabled (3-Line LCD Phones)) Enable PassThru Port (8 and 11-Line LCD Phones)	N/A LAN Port PC Port PC Port PassThru Enable/Disable (3-Line LCD Phonesi) PC Port PassThru Enable/Disable (8 and 11-Line LCD Phones)	 ethernet port 0 ethernet port 1 pc port passthrough enabled	The send (TX) and receive (RX) negotiation to use on the Ethernet LAN Port and Ethernet PC Port for transmitting and receiving data over the LAN or to/from your PC, respectively. For more information on configuring the LAN and PC port negotiation, see "Configuring LAN and PC Port Negotiation" on page 4-23. **Note:** The PC Port parameters are not applicable to the 6730i IP Phone.

Advanced Network Settings

Parameter In IP Phone UI	Parameter in Aastra Web UI	Parameters in Configuration Files	Description
DHCP User Class	DHCP User Class	dhcp userclass	Specifies the User Class DHCP Option 77 that the phone sends to the configuration server with the DHCP Request packet. **Note:** If you specify a value for this parameter, you must restart your phone for the change to take affect. Any change in its value during start-up results in an automatic reboot. For more information, see "Using Option 77 User Class on the IP Phone" on page 4-13.
Download Options	DHCP Download Options	dhcp config option override	The value specified for this parameter overrides the precedence order for determining a configuration server. Valid values are: -1 (Disabled - ignores all DHCP configuration options). 0 (Any) 43 66 159 160 **Notes**: **1.** If the DHCP server supplies Options 159 and 160, the phones will attempt to contact the configuration server given in these options. **2.** You must restart the IP Phone for this parameter to take affect. For more information, see "Using Options 159 and 160 on the IP Phone" on page 4-16. For more information about setting DHCP download preference, see "Configuration Server Download Precedence" on page 4-19.
LLDP Support	LLDP	lldp	Enables or disables Link Layer Discovery Protocol for Media Endpoint Devices (LLDP-MED) on the IP Phone. **Caution:** In Release 2.3 and later, LLDP is enabled by default. If LLDP is enabled on your network, the phones may come up with different network settings. For more information on this feature, see "Link Layer Discovery Protocol for Media Endpoint Devices (LLDP-MED) and Emergency Location Identification Number (ELIN)" on page 5-97.

Parameter In IP Phone UI	Parameter in Aastra Web UI	Parameters in Configuration Files	Description
N/A	LLDP Packet Interval	lldp interval	The amount of time, in seconds, between the transmission of LLDP Data Unit (LLDPDU) packets. The value of zero (0) disables this parameter. **Caution:** In Release 2.3 and later, LLDP is enabled by default. If LLDP is enabled on your network, the phones may come up with different network settings. For more information on this feature, see "Link Layer Discovery Protocol for Media Endpoint Devices (LLDP-MED) and Emergency Location Identification Number (ELIN)" on page 5-97.
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 4-31.
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 4-31.
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 4-31.
N/A	Rport (RFC 3581)	sip rport	Allows you to enable (1) or disable (0) the use of Rport on the IP phone. "Rport" in RFC 3581, allows a client to request that the server send the response back to the source IP address **and** the port from which the request came. For more information, see Chapter 4, "RPORT" on page 4-61.
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 4-63.

Parameter In IP Phone UI	Parameter in Aastra Web UI	Parameters in Configuration Files	Description
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 4-63.
N/A	STUN Server	sip stun ip	IP address of the **STUN** server (also know as Simple Traversal of UDP through NAT). **Notes**: **1.** The UPnP and NAT IP configuration parameters take precedence over the STUN and TURN parameters. **2.** STUN does not work if the NAT device is symmetric. For more information, see Chapter 4, "STUN and TURN Protocols" on page 4-36.
N/A	STUN Port	sip stun port	Port number of the **STUN** server (also know as Simple Traversal of UDP through NAT). **Notes**: **1.** The UPnP and NAT IP configuration parameters take precedence over the STUN and TURN parameters. **2.** STUN does not work if the NAT device is symmetric. For more information, see Chapter 4, "STUN and TURN Protocols" on page 4-36.
N/A	TURN Server	sip turn ip	IP address of the **TURN** server (also known as Traversal Using Relay NAT). **Note:** The UPnP and NAT IP configuration parameters take precedence over the STUN and TURN parameters. For more information, see Chapter 4, "STUN and TURN Protocols" on page 4-36.
N/A	TURN Port	sip turn port	Port number of the **TURN** server (also known as Traversal Using Relay NAT). **Note:** The UPnP and NAT IP configuration parameters take precedence over the STUN and TURN parameters. For more information, see Chapter 4, "STUN and TURN Protocols" on page 4-36.

Parameter In IP Phone UI	Parameter in Aastra Web UI	Parameters in Configuration Files	Description
N/A	TURN User ID	sip turn user	Username that a user must enter when accessing an account on the TURN server. **Note:** The UPnP and NAT IP configuration parameters take precedence over the STUN and TURN parameters. For more information, see Chapter 4, "STUN and TURN Protocols" on page 4-36.
N/A	TURN Password	sip turn pass	Password that a user must enter when accessing an account on the TURN server. **Note:** The UPnP and NAT IP configuration parameters take precedence over the STUN and TURN parameters. For more information, see Chapter 4, "STUN and TURN Protocols" on page 4-36.

HTTPS Settings

Advanced Network Settings includes HTTPS settings for the IP Phones.

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 4-41.
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 4-41.
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 4-41.

Parameter In IP Phone UI	Parameter in Aastra Web UI	Parameters in Configuration Files	Description
Cert Validation	Validate Certificates	https validate certificates	Enables or disables the HTTPS validation of certificates on the phone. When this parameter is set to 1, the HTTPS client performs validation on SSL certificates before accepting them. **Notes**: **1.** If you are using HTTPS as a configuration method, and use a self signed certificate, you must set this parameter to "0" (disabled) before upgrading to Release 2.3 or later of the IP Phones. **2.** If you are using HTTPS and the certificates are not valid or are not signed by Verisign, Thawte, or GeoTrust, Comodo, or CyberTrust, the phones fail to download configuration files. For more information, see Chapter 4, "HTTPS Server Certificate Validation" on page 4-45.
Check Expires	Check Certificate Expiration	https validate expires	Enables or disables the HTTPS validation of the expiration of the certificates. When this parameter is set to 1, the HTTPS client verifies whether or not a certificate has expired prior to accepting the certificate. **Note:** If the "**https validate expires**" parameter is set to **enable**, the clock on the phone must be set for the phone to accept the certificates. For more information, see Chapter 4, "HTTPS Server Certificate Validation" on page 4-45.
Check Hostnames	Check Certificate Hostnames	https validate hostname	Enables or disables the HTTPS validation of hostnames on the phone. For more information, see Chapter 4, "HTTPS Server Certificate Validation" on page 4-45.
N/A	Trusted Certificates Filename	https user certificates	Specifies a file name for a .PEM file located on the configuration server. This file contains the User-provided certificates in PEM format. These certificates are used to validate peer certificates. **Note:** You must disable the "**https validate certificates**" parameter in order for the phone to accept the User-provided certificates. For more information, see Chapter 4, "HTTPS Server Certificate Validation" on page 4-45.

Type of Service (ToS), DSCP

Advanced Network Settings include Type of Service (ToS) and Differentiated Services Code Point (DSCP) for the IP phones.

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 4-51.
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 4-51.
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 4-51.

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 4-51.
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 4-51.
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 4-51.
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 4-51.
PC Port Settings (Port 1)			

Parameter In IP Phone UI	Parameter in Aastra Web UI	Parameters in Configuration Files	Description

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.
			
			Example
			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 4-51.
			
			Note: The PC Port parameters are not applicable to the 6730i IP Phone.

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 4-51.

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**
- **Autodial Settings**

Notes:

1. 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.

2. The IP phones allow you to define different SIP lines with the same account information (i.e., same user name) but with different registrar and proxy IP addresses. This feature works with Registration, Subscription, and Notify processing. This feature also works with the following types of calls: incoming, outgoing, Broadsoft Shared Call Appearance (SCA), Bridged Line Appearance (BLA), conference, transfer, blind transfer.

Basic SIP Authentication Settings

Parameter In IP Phone UI	Parameter in Aastra Web UI	Parameters in Configuration Files	Description
Screen Name	Screen Name (Global and Per-Line)	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 SIP Settings" on page 4-66.
N/A	Screen Name 2 (Global and Per-Line)	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 SIP Settings" on page 4-66.
User Name	Phone Number (Global and Per-Line)	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 SIP Settings" on page 4-66.
Display Name	Caller ID (Global and Per-Line)	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 SIP Settings" on page 4-66.
Auth Name	Authentication Name (Global and Per-Line)	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 SIP Settings" on page 4-66.
Password	Password (Global and Per-Line)	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 SIP Settings" on page 4-66.

Parameter In IP Phone UI	Parameter in Aastra Web UI	Parameters in Configuration Files	Description
N/A	BLA Number (Global and Per-Line)	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 SIP Settings" on page 4-66. For more information about BLA, see Chapter 5, the section, "Bridged Line Appearance (BLA)" on page 5-195.
N/A	Line Mode (Global and Per-Line)	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), Reserved for (2), or BLA (3). Default is Generic (0). For more information, see Chapter 4, "Basic SIP Settings" on page 4-66.

Basic SIP Network Settings

Parameter In IP Phone UI	Parameter in Aastra Web UI	Parameters in Configuration Files	Description
Proxy Server	Proxy Server (Global and Per-Line)	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 SIP Settings" on page 4-66.
Proxy Port	Proxy Port (Global and Per-Line)	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 SIP Settings" on page 4-66.
N/A	Backup Proxy Server (Global and Per-Line)	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 SIP Settings" on page 4-66.
N/A	Backup Proxy Port (Global and Per-Line)	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 SIP Settings" on page 4-66.
N/A	Outbound Proxy Server (Global and Per-Line)	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 SIP Settings" on page 4-66.
N/A	Outbound Proxy Port (Global and Per-Line)	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 SIP Settings" on page 4-66.
N/A	N/A	sip backup outbound proxy server (global) sip lineN backup outbound proxy server (per-line)	The IP address or domain name of the backup outbound SIP proxy server for which the IP phone uses when the primary SIP proxy is unavailable. For more information, see Chapter 4, "Backup Outbound Proxy and Failover Support" on page 4-70.

Parameter In IP Phone UI	Parameter in Aastra Web UI	Parameters in Configuration Files	Description
N/A	N/A	sip backup outbound proxy port (global) sip lineN backup outbound proxy port (per-line)	The backup outbound proxy port on the backup outbound proxy server to which the IP phone sends all SIP messages. For more information, see Chapter 4, "Backup Outbound Proxy and Failover Support" on page 4-70.
Registrar Server	Registrar Server (Global and Per-Line)	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 SIP Settings" on page 4-66.
Registrar Port	Registrar Port (Global and Per-Line)	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 SIP Settings" on page 4-66.
N/A	Backup Registrar Server (Global and Per-Line)	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 SIP Settings" on page 4-66.
N/A	Backup Registrar Port (Global and Per-Line)	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 SIP Settings" on page 4-66.
N/A	Registration Period (Global and Per-Line)	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 SIP Settings" on page 4-66.
N/A	Conference Server URI (Global and Per-Line)	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 5-327.

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 4-83.
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 4-83.
Missed Call Summary Subscription (Global and Per-Line)	sip missed call summary subscription (global) sip lineN missed call summary subscription (per-line)	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.
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.
AS-Feature-Event Subscription (Global and Per-Line)	sip as-feature-event subscription (global) sip lineN as-feature-event subscription (per-line)	Enables or disables the specified line with the BroadSoft's server-side DND, CFWD, or ACD features. For more information about this parameter, see Chapter 6, the section, "As-Feature-Event Subscription" on page 6-16.

Parameter in Aastra Web UI	Parameters in Configuration Files	Description
AS-Feature-Event Subscription Period	sip as-feature-event subscription period	Specifies the amount of time, in seconds, between resubscribing. If the phone does not resubscribe in the time specified for this parameter, it loses subscription. For more information about this parameter, see Chapter 6, the section, "As-Feature-Event Subscription" on page 6-16.
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, "TR-069 Support" 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, "TR-069 Support" 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 4-83.
Timer 1 and Timer 2	sip T1 timer sip T2 timer	These timers are SIP transaction layer timers defined in RFC 3261. Timer 1 is an estimate, in milliseconds, of the round-trip time (RTT). Timer 2 represents the amount of time, in milliseconds, a non-INVITE server transaction takes to respond to a request. For more information, see Chapter 4, "Advanced SIP Settings (optional)" on page 4-83.
Transaction Timer	sip transaction timer	The amount of time, in milliseconds that the phone allows the callserver (registrar/proxy) to respond to SIP messages that it sends. If the phone does not receive a response in the amount of time designated for this parameter, the phone assumes the message has timed out. For more information, see Chapter 4, "Advanced SIP Settings (optional)" on page 4-83.

Parameter in Aastra Web UI	Parameters in Configuration Files	Description
Transport Protocol	sip transport protocol	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, see Chapter 4, "Advanced SIP Settings (optional)" on page 4-83.
Local SIP UDP/TCP Port	sip local port	Specifies the local source port (UDP/TCP) from which the phone sends SIP messages. **Notes**: **1.** It is recommended that you avoid the conflict RTP port range in case of a UDP transport. **2.** 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.If symmetric UDP signaling is disabled, the phone sends from random ports but it listens on the configured SIP local port. For more information, see Chapter 4, "SIP and TLS Source Ports for NAT Traversal" on page 4-34.
Local SIP TLS Port	sip local tls port	Specifies the local source port (SIPS/TLS) from which the phone sends SIP messages. **Note:** It is recommended that you avoid the conflict with any TCP ports being used. For example: WebUI HTTP server on 80/tcp and HTTPS on 443/tcp. For more information, see Chapter 4, "SIP and TLS Source Ports for NAT Traversal" on page 4-34.
Registration Failed Retry Timer	sip registration retry timer	Specifies the time, in seconds, that the phone waits between registration attempts when a registration is rejected by the registrar. For more information, see Chapter 4, "Advanced SIP Settings (optional)" on page 4-83.

Parameter in Aastra Web UI	Parameters in Configuration Files	Description
Registration Timeout Retry Timer	sip registration timeout retry timer	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. For more information, see Chapter 4, "Advanced SIP Settings (optional)" on page 4-83.
Registration Renewal Timer	sip registration renewal timer	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. For more information, see Chapter 4, "Advanced SIP Settings (optional)" on page 4-83.
BLF Subscription Period	sip blf subscription period	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. For more information, see Chapter 4, "Advanced SIP Settings (optional)" on page 4-83.
ACD Subscription Period	sip acd subscription period	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. For more information, see Chapter 4, "Advanced SIP Settings (optional)" on page 4-83.
BLA Subscription Period	sip bla subscription period	Specifies the amount of time, in seconds, that the phone waits to receive a BLA subscribe message from the server. If you specify zero (0), the phone uses the value specified for the BLA expiration in the subscribe message received from the server. If no value is specified, the phone uses the default value of 300 seconds. For more information, see Chapter 4, "Advanced SIP Settings (optional)" on page 4-83.
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" on page 6-22.

Parameter in Aastra Web UI	Parameters in Configuration Files	Description
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 6-24.
XML SIP Notify	sip xml notify event	Enables or disables the phone to accept or reject an aastra-xml SIP NOTIFY message. **Note:** To ensure the SIP NOTIFY is coming from a trusted source, it is recommended that you enable the Whitelist feature (**Whitelist Proxy** parameter) on the IP phone. If enabled, and the phone receives a SIP NOTIFY from a server that is NOT on the whitelist (i.e. untrusted server), the phone rejects the message. For more information about XML SIP Notify see Chapter 6, the section, "XML SIP Notify Events" on page 5-310.

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 4-90.
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 4-90.
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 RFC2833. 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 4-90.
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 4-90.
N/A	DTMF Method (Global and Per-Line)	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 4-90.

Parameter In IP Phone UI	Parameter in Aastra Web UI	Parameters in Configuration Files	Description
N/A	RTP Encryption (Global and Per-Line)	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 4-90.
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 4-90.

Autodial Settings

You can configure the following Autodial settings.

Parameter In IP Phone UI	Parameter in Aastra Web UI	Parameters in Configuration Files	Description
N/A	Autodial Number (Global and Per-Line)	sip autodial number sip lineN autodial number	Globally or on a per-line basis, 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. For more information, see Chapter 4, "Autodial Settings" on page 4-101.
N/A	Autodial Timeout (Global and Per-Line)	sip autodial timeout sip lineN autodial timeout	Globally or on a per-line basis, 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). For more information, see Chapter 4, "Autodial Settings" on page 4-101.
N/A	Use Global Settings (Per-line configurations only)	N/A	For each line, this parameter specifies to use the global autodial settings of "Autodial Number" and "Autodial Timeout". For more information, see Chapter 4, "Autodial Settings" on page 4-101.

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
9143i	9
9480i	9
9480i CT	9
6730i	6
6731i	6
6739i	9
6753i	9
6755i	9
6757i	9
6757i 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** (On a per-line basis, you can also enable/disable the "**Use Global Settings**" parameter.)

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 4-66
- "Advanced SIP Settings (optional)" on page 4-83
- "Real-time Transport Protocol (RTP) Settings" on page 4-90
- "Autodial Settings" on page 4-101

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
9143i	-	Not Applicable	7
9480i	6	Not Applicable	-
9480i CT	6	Not Applicable	-
6730i	-	Not Applicable	8
6731i	-	Not Applicable	8
6739i	55	36 to 108* (Model M670i) 60 to 180** (Model M675i)	-
6753i	-	36 to 108* (Model M670i)	6
6755i	6	36 to 108* (Model M670i) 60 to 180** (Model M675i)	6
6757i	12	36 to 108* (Model M670i) 60 to 180** (Model M675i)	-
6757i CT	12	36 to 108* on Base Station (Model M670i) 60 to 180** on Base Station (Model M675i)	-

*The M670i expansion module consists of 36 softkeys. You can have up to 3 expansion modules on an IP phone totaling 108 softkeys. Valid for 6739i, 6753i, 6755i, 6757i, and 6757i CT phones.

**The M675i expansion module consists of 60 softkeys. You can have up to 3 expansion modules on an IP phone totaling 180 softkeys. Valid for 6739i, 6755i, 6757i, and 6757i 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-202.

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 5-130.

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 Action URI feature prevents the phones from hanging if the Action URIs should fail. The phones also support transparent, non-blocking, XML post execute item URIs.

The IP phone XML events that support this feature are defined in the following table.

Action URI	Description
Startup	Specifies the URI for which the phone executes a GET on when a startup event occurs.
Successful Registration	Specifies the URI for which the phone executes a GET on when a successful registration event occurs.
Registration Event	Specifies the URI for when registration events occur or when there are registration state changes. **Note:** This action URI is not called when the same event is repeated (for example, a timeout occurs again when registration is already in a timeout state.)
Incoming Call	Specifies the URI for which the phone executes a GET on when an incoming call event occurs.
Outgoing Call	Specifies the URI for which the phone executes a GET on when an outgoing call event occurs.
Offhook	Specifies the URI for which the phone executes a GET on when an offhook event occurs.
Onhook	Specifies the URI for which the phone executes a GET on when an onhook event occurs.
Connected	Specifies the URI for which the phone executes an HTTP GET when it goes into the “connected” state. This includes regular phone calls, intercom calls, paging calls, RTP streaming, and the playing of a WAV file. It is also triggered when the phone establishes the second leg of a 3-way call.
Disconnected	Specifies the URI that the phone executes a GET on, when it transitions from the incoming, outgoing, calling, or connected state into the idle state.
XML SIP Notify	Specifies the URI to be called when an empty XML SIP NOTIFY is received by the phone.
Poll	Specifies the URI to be called every "action uri poll interval" seconds.
Poll Interval	Specifies the interval, in seconds, between calls from the phone to the "action uri poll".

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 “Action URIs” on page 5-296.

XML SIP Notify Events and Action URIs

In order for an XML push to bypass the NAT/firewall, the phone supports a proprietary SIP NOTIFY event (aastra-xml) with or without XML content.

If XML content is provided in the SIP NOTIFY, it is processed directly by the phone as it is done for an XML PUSH.

Reference

For more information about enabling the XML SIP Notify on the IP Phones, see Chapter 5, the section, "XML SIP Notify Events" on page 5-310.

Polling Action URIs

Another way to reach a phone behind a NAT/firewall is to have the phone make an XML call at periodic intervals. An Administrator can use the **action uri poll** to command the phone to perform an XML call at configurable intervals.

An Administrator can specify the URI to be called and specify the interval between polls. Configuration of this feature is dynamic (no reboot required).

Reference

For more information about configuring the polling and polling interval of Action URIs, see "Polling Action URIs" on page 5-303.

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 **Note:** For DHCP to automatically populate the IP address or domain name for the download servers, your DHCP server must support Option 66. The IP phones also support Option 60 and 43. For more information, see Chapter 4, the section, "DHCP" on page 4-3. For more information about download protocols on the IP Phone, see Chapter 4, "Configuration Server Protocol" on page 4-104.
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 4-104.
Primary TFTP Path	TFTP Path	tftp path	Specifies the path name for which the configuration files reside on the TFTP server for downloading to the IP Phone. If the IP phone's configuration and firmware files are located in a sub-directory beneath the server's root directory, the relative path to that sub-directory should be entered in this field. **Note:** Enter the path name in the form *folderX\folderX\folderX*. For example, ***ipphone\6757i\configfiles***. For more information, see Chapter 4, "Configuration Server Protocol" on page 4-104.

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 4-104.
Alternate TFTP Path	Alternate TFTP Path	alternate tftp path	Specifies the path name for which the configuration files reside on the Alternate TFTP server for downloading to the IP Phone. If the IP phone's configuration and firmware files are located in a sub-directory beneath the server's 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 4-104.
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 4-104.
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 4-104.
FTP Path	FTP Path	ftp path	Specifies the path name for which the configuration files reside on the FTP server for downloading to the IP Phone. If the IP phone's configuration and firmware files are located in a sub-directory beneath the server's 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 4-104.

Parameter In IP Phone UI	Parameter in Aastra Web UI	Parameters in Configuration Files	Description
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 4-104.
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 4-104.
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 4-104.
HTTP Path	HTTP Path	http path	Specifies the path name for which the configuration files reside on the HTTP server for downloading to the IP Phone. If the IP phone’s configuration and firmware files are located in a sub-directory beneath the server’s 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 4-104.
HTTP Port	HTTP Port	http port	Specifies the HTTP port that the server uses to load the configuration to the phone over HTTP. For more information, see Chapter 4, “Configuration Server Protocol” on page 4-104.
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 4-104.

Parameter In IP Phone UI	Parameter in Aastra Web UI	Parameters in Configuration Files	Description
Download Path	HTTPS Path	https path	Specifies the path name for which the configuration files reside on the HTTPS server for downloading to the IP Phone. If the IP phone's configuration and firmware files are located in a sub-directory beneath the server's 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 4-104.
Download Port	HTTPS Port	https port	Specifies the HTTP port that the server uses to load the configuration to the phone over HTTP. For more information, see Chapter 4, "Configuration Server Protocol" on page 4-104.
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, "Using the Auto-Resync Feature" on page 8-7.

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). For more information, see Chapter 8, the section, "Using the Auto-Resync Feature" on page 8-7.
N/A	Maximum Delay	auto resync max delay	Specifies the maximum time, in minutes, the phone waits past the scheduled time before starting a checksync. For more information, see Chapter 8, the section, "Using the Auto-Resync Feature" on page 8-7.
N/A	Days	auto resync days	Specifies the amount of days that the phone waits between checksync operations. **Note:** A value of 0 causes the phone to checksync every time the clock reads the proper time. A value of 1 forces the phone to wait 24 hours prior to doing the first checksync. For more information, see Chapter 8, the section, "Using the Auto-Resync Feature" on page 8-7.
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 5-288.

Firmware Update Features

The IP phones support the protocols, TFTP, FTP, HTTP or HTTPS to download configuration files and upgrade firmware to the phones from a configuration server.

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

- Using the "**Firmware Update**" page in the Aastra Web UI at the location ***Advanced Settings->Firmware Update***.
- Using the IP Phone UI or the Aastra Web UI to **restart** the phone. The phone automatically looks for firmware updates and configuration files during the boot process.
- Setting an **Auto-Resync** feature to automatically update the firmware, configuration files, or both at a specific time in a 24-hour period). (Feature can be enabled using the configuration files or the Aastra Web UI).

Reference

For more information about firmware update, see Chapter 8, "Upgrading the Firmware."

TLS Support

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. An Administrator can configure the following parameters for TLS Support.

Parameter In IP Phone UI	Parameter in Aastra Web UI	Parameters in Configuration Files	Description
N/A	Transport Protocol	sip transport protocol	Specifies the protocol that the RTP port on the IP phone uses to send out SIP signaling packets. Default is USP. **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, see Chapter 6, the section, "Transport Layer Security (TLS)" on page 6-26.
N/A	N/A	sips persistent tls	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. For more information, see Chapter 6, the section, "Transport Layer Security (TLS)" on page 6-26.

Parameter In IP Phone UI	Parameter in Aastra Web UI	Parameters in Configuration Files	Description
N/A	Root and Intermediate Certificates Filename	sips root and intermediate certificates	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. For more information, see Chapter 6, the section, "Transport Layer Security (TLS)" on page 6-26.
N/A	Local Certificate Filename	sips local certificate	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. For more information, see Chapter 6, the section, "Transport Layer Security (TLS)" on page 6-26.

Parameter In IP Phone UI	Parameter in Aastra Web UI	Parameters in Configuration Files	Description
N/A	Private Key Filename	sips private key	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. For more information, see Chapter 6, the section, "Transport Layer Security (TLS)" on page 6-26.
N/A	Trusted Certificates Filename	sips trusted certificates	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. For more information, see Chapter 6, the section, "Transport Layer Security (TLS)" on page 6-26.

802.1x Support

The IP phones support the IEEE 802.1x Protocol. The 802.1x Protocol is a standard for passing Extensible Authentication Protocol (EAP) over a wired or wireless Local Area Network (LAN).

The 802.1x Protocol on the IP phone facilitates media-level access control, and offers the capability to permit or deny network connectivity, control LAN access, and apply traffic policy, based on user or endpoint identity. This feature supports both the EAP-MD5 and EAP-TLS Protocols.

Note: If configuring 802.1x using the IP Phone UI, the certificates and private keys must already be configured and stored on the phone. Use the configuration files or the Aastra Web UI to load certificates and private keys.

An Administrator can configure the following parameters for the 802.1x Protocol.

Parameter In IP Phone UI	Parameter in Aastra Web UI	Parameters in Configuration Files	Description
EAP-MD5			
802.1x Mode	EAP Type	eap-type	Specifies the type of authentication to use on the IP Phone. For more information, see Chapter 6, the section, "802.1x Support" on page 6-30.
Identity	Identity	identity	Specifies the identity or username used for authenticating the phone. **Note:** The value you enter for this parameter also displays in the Aastra Web UI at the path Advanced Settings-> 802.1x Support->General->Identity For more information, see Chapter 6, the section, "802.1x Support" on page 6-30.
MD5 Password	MD5 Password	md5 password	Specifies the password used for the MD5 authentication of the phone. **Note:** The value you enter for this parameter also displays in the Aastra Web UI at the path Advanced Settings-> 802.1x Support->EAP-MD5 Settings->MD5 Password. The password displays as "*******". For more information, see Chapter 6, the section, "802.1x Support" on page 6-30.
EAP-TLS			
802.1x Mode	EAP Type	eap type	Specifies the type of authentication to use on the IP Phone. For more information, see Chapter 6, the section, "802.1x Support" on page 6-30.

Parameter In IP Phone UI	Parameter in Aastra Web UI	Parameters in Configuration Files	Description
Identity	Identity	identity	Specifies the identity or username used for authenticating the phone. **Note:** The value you enter for this parameter also displays in the Aastra Web UI at the path Advanced Settings-> 802.1x Support->General->Identity For more information, see Chapter 6, the section, "802.1x Support" on page 6-30.
N/A	Root and Intermediate Certificates Filename	802.1x root and intermediate certificates	Specifies the file name that contains the root and intermediate certificates related to the local certificate. For more information, see Chapter 6, the section, "802.1x Support" on page 6-30.
N/A	Local Certificate Filename	802.1x local certificate	Specifies the file name that contains the local certificate. For more information, see Chapter 6, the section, "802.1x Support" on page 6-30.
N/A	Private Key Filename	802.1x private key	Specifies the file name that contains the private key. For more information, see Chapter 6, the section, "802.1x Support" on page 6-30.
N/A	Trusted Certificates Filename	802.1x trusted certificates	Specifies the file name that contains the trusted certificates. For more information, see Chapter 6, the section, "802.1x Support" on page 6-30.

Troubleshooting

The Troubleshooting feature in the Aastra Web UI provides tasks that a system administrator can perform on the IP phones for troubleshooting purposes. Using this feature, 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 (including “Free Memory”and “Maximum Memory Block Size”)
- Enable/disable a WatchDog task
- View Error Messages
- Enable/disable the uploading of configuration and crash file information to a pre-defined server

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

Reference

For more information about troubleshooting on the IP Phones, see Chapter 9, “Troubleshooting.”

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 9480i, 9480i CT, 6739i, 6755i, 6757i, and/or 6757i CT when configuring Administrator Options. For information on using the 9143i, 6730i, 6731i, and 6753i keys to configure the Administrator Options, see Chapter 2, the section, "Using the Options Key" on page 2-5.

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-14.

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 B, "Configuring the IP Phone at the Asterisk IP PBX."

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

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
- Hostname
- LAN Port
- PC Port Pass Thru Enable/Disable
- 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)
- Network Time Protocol Server
- IP Address
- TFTP Server or Alternate TFTP Server if enabled on the phone
- TFTP Path or Alternate TFTP Path if enabled on the phone
- FTP Server
- FTP Path
- HTTP Server
- HTTP Path
- HTTP Port
- HTTPS Server
- HTTPS Path

- HTTPS Port

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.

Enabling/Disabling DHCP Using the Configuration Files

Use the following procedure to enable/disable DHCP on the phone using the configuration files.

Configuration Files
For specific parameters you can set in the configuration files, see Appendix A, the section, "Network Settings" on page A-7.

Enabling/Disabling DHCP Using the IP Phone UI

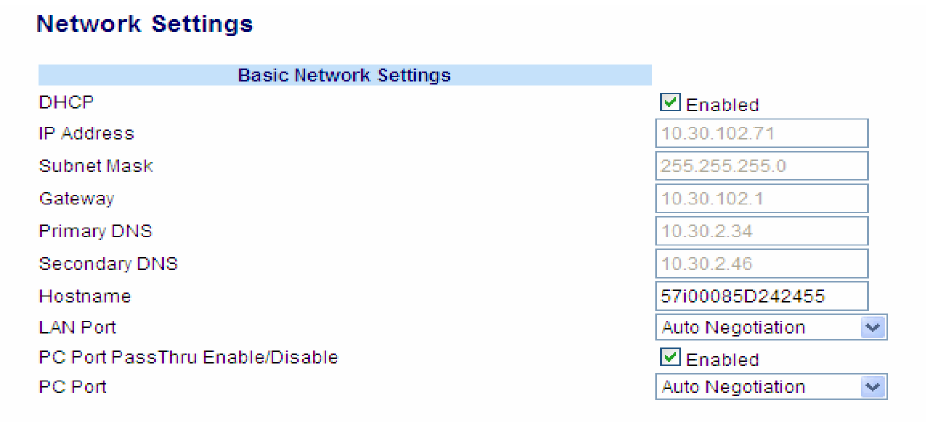
Use the following procedure to enable/disable DHCP on the phone using the IP Phone UI.

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.
For the 6739i:	
1	Press the **Options** key on the phone to enter the Options List.
2	Press **Advanced**. A keyboard displays.
3	Enter the Administrator password using the keyboard. Default is "**22222**".
4	Press **Network**.
5	Press **DHCP Settings**.
6	In the "**Use DHCP?**" field, select "**Enabled**" to enable DHCP. or Press "**Disabled**" to disable DHCP.
7	Press the to return to the previous screen.
8	Press the button or the button at any time to return to the idle screen.

Enabling/Disabling DHCP Using the Aastra Web UI

Use the following procedure to enable/disable DHCP using the Aastra Web UI.

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 Save Settings to save your settings.

DHCP Options 60, 66, and 43 Server Configurations

Option 66

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 your applicable configuration server into your IP phone configuration.

Options 60 and 43

The Aastra phones also support Option 60 and Option 43 as per RFC 2132.

Option 60 (Vendor Class Identifier) provides the DHCP server with a unique identifier for each phone model. This enables a system administrator to send the phone a customized Server Configuration in option 43.

The table below lists the identifier values for each phone model.

Model	Identifier Value
9143i	AastraIPPhone9143i
9480i	AastraIPPhone9480i
9480i CT	AastraIPPhone9480iCT
6730i	AastraIPPhone6730i
6731i	AastraIPPhone6731i
6739i	AastraIPPhone6739i
6753i	AastraIPPhone53i
6755i	AastraIPPhone55i
6757i	AastraIPPhone57i
6757i CT	AastraIPPhone57iCT

The System administrator can use the Vendor Class Identifier to send the phone a customized Server Configuration in option 43 (Vendor-Specific information).

Note: If Aastra IP Phones receive the server configuration from both DHCP Option 66 and DHCP Option 43, then Option 43 takes precedence over Option 66.

Using Option 43 to Customize the IP Phone

A System Administrator can customize the IP Phone(s) in the network by entering a text string in the phone’s configuration files. The following is an Option 43 example using Linux.

On the startup of the phones, when the DHCP server receives the request with the information in this example, it allows the 6757i phones to use FTP and the 6757i CT phones to use TFTP from the same server.

Linux Example

A System Administrator can enter the following in the Aastra IP Phone configuration file:

```
option space AastraIPPhone57i;
option AastraIPPhone57i.cfg-server-name code 02 = text;

option space AastraIPPhone57iCT;
option AastraIPPhone57iCT.cfg-server-name code 02 = text;

Subnet 192.168.1.0 netmask 255.255.255.0 {

  class "vendor-class-57i" {
    match if option vendor-class-identifier="AastraIPPhone57i";
    vendor-option-space AastraIPPhone57i;
    option AastraIPPhone57i.cfg-server-name "ftp://
username:password@10.10.10.1";
  }
  class "vendor-class-57iCT" {
    match if option vendor-class-identifier="AastraIPPhone57iCT";
    vendor-option-space AastraIPPhone57iCT;
    option AastraIPPhone57iCT.cfg-server-name "tftp://10.10.10.1";
  }
}
```

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”;
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”;

Option 43 Redirection and Configuration Server (RCS) Bypass

DHCP Option 43 includes the ability to bypass contacting Aastra's Redirection and Configuration Server (RCS), in addition to the previous support of setting the configuration server to contact.

A sub-option code 3 uses a boolean value (true or false) that controls whether or not the phone should contact the RCS after a factory default. If this value is set to false, the the RCS is not contacted. If it is set to true or is missing, then the RCS is contacted as per previous releases. This can be used in conjunction with the existing code 2 sub-option to set the configuration server.

Configuring RCS Bypass via Option 43 on a Linux DHCP Server

The following example illustrates how to configure RCS bypass via Option 43 on a Linux DHCP server.

```
option space AastraIPPhone;
option AastraIPPhone.cfg-server-name code 02 = text;
option AastraIPPhone.contact-rcs code 03 = boolean;

Subnet 192.168.1.0 netmask 255.255.255.0 {

#The 6757i phones do not contact the RCS but use the defined FTP server for
configuration files.
  class "vendor-class-57i" {
    match if option vendor-class-identifier="AastraIPPhone57i";
    vendor-option-space AastraIPPhone;
    option AastraIPPhone.cfg-server-name "ftp://username:password@10.10.10.1";
    option AastraIPPhone.contact-rcs false;
  }
#The 6757iCT phones do not contact the RCS.
class "vendor-class-57iCT" {
    match if option vendor-class-identifier="AastraIPPhone57iCT";
    vendor-option-space AastraIPPhone;
    option AastraIPPhone.contact-rcs false;
```

Using Option 12 Hostname on the IP Phone

If you set the phone to use DHCP Option 12, the phone automatically sends this option to the configuration server. This option specifies the hostname (name of the client). The name may or may not be qualified with the local domain name (based on RFC 2132). See RFC 1035 for character set restrictions.

Notes:

1. The hostname of [<model><MAC address>] automatically populates the field on initial startup of the phone. For example, for a 6753i, the "Hostname" field is automatically populated as "53i00085D164435", where the model number is "6753i" and the MAC address is "00085D164435".

2. If the configuration server sends the hostname back to the phone in a DHCP Reply Packet, the hostname is ignored.

An Administrator can change the "Hostname" for the DHCP Option 12 via the configuration files, the IP Phone UI, and the Aastra Web UI.

Configuring DHCP Option 12 Hostname on the IP Phone

Use the following procedures to configure DHCP Option 12 Hostname on the IP Phone.

Configuration Files

For specific parameters you can set in the configuration files, see Appendix A, the section, "DHCP Option Settings" on page A-12.

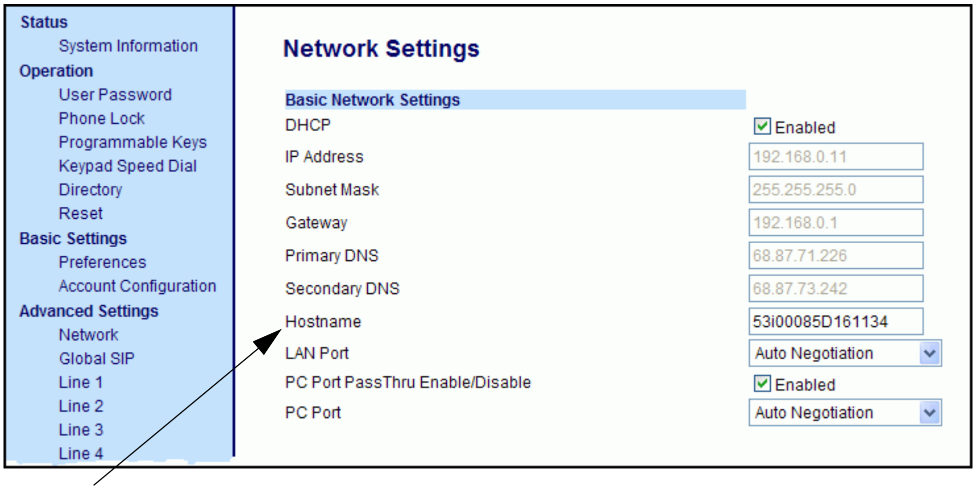
Aastra IP Phone UI

Step	Action
1	Press **Options**, and then select **Administrator Menu**.
2	Select **Network Settings**.
3	Select **Hostname**.
4	By default, the "**Hostname**" field is automatically populated with [<Model><MAC address>] of your phone (for example, 53i00085D164435). If you want to change the hostname, enter a hostname for your phone in the "**Hostname**" field, then press **DONE**. Valid values are up to 64 alpha-numeric characters. You can use a fully qualified domain name if required.
5	Restart the phone for the change to take affect.

Aastra IP Phone UI	
Step	**Action**
For the 6739i:	
1	Press the **Options** key on the phone to enter the Options List.
2	Press **Advanced**. A keyboard displays.
3	Enter the Administrator password using the keyboard. Default is "**22222**".
4	Press **Network**.
5	Press to scroll to the next screen.
6	By default, the "**Hostname**" field is automatically populated with [<Model><MAC address>] of your phone (for example, 53i00085D164435). If you want to change the hostname, press the button in the "**Hostname**" field, and enter a hostname for your phone in the text box that displays.. Valid values are up to 64 alpha-numeric characters. You can use a fully qualified domain name if required.
7	Press until the Options List screen displays.
8	Press **Restart** to restart the phone for the change to take affect.

Aastra Web UI

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



Hostname</td></tr>
<tr><td>2</td><td>By default, the “Hostname” field is automatically populated with [<Model><MAC address>] of your phone (for example, 53i00085D164435).

If you want to change the hostname, enter a hostname for your phone in the “Hostname” field. Valid values are up to 64 alpha-numeric characters. You can use a fully qualified domain name if required.</td></tr>
<tr><td>3</td><td>Click 

to save your changes.

Note: Changing the “Hostname” field requires a reboot of your phone.</td></tr>
<tr><td>4</td><td>Click on Operation->Reset, and click RESTART.</td></tr>
</table>

Using Option 77 User Class on the IP Phone

DHCP Option 77 User Class is sent in DHCP request packets from the phone to the configuration server. This Option 77 defines specific User Class identifiers to convey information about a phone's software configuration or about its user's preferences. For example, you can use the User Class option to configure all phones in the Accounting Department with different user preferences than the phones in the Marketing Department. A DHCP server uses the User Class option to choose the address pool for which it allocates an address from, and/or to select any other configuration option.

Notes:

1. If the User Class is not specified (left blank) in the DHCP request packet, the phone does not send the User Class DHCP Option 77.

2. Multiple User Classes inside a DHCP Option 77 are not supported.

3. DHCP Option 77 may affect the precedence of DHCP Options, dependent on the DHCP Server.

An Administrator can configure the DHCP Option 77 User Class via the configuration files, the IP Phone UI, and the Aastra Web UI.

Configuring DHCP Option 77 User Class on the IP Phone

Use the following procedures to configure DHCP Option 77 User Class on the IP Phone.

Configuration Files

For specific parameters you can set in the configuration files, see Appendix A, the section, "DHCP Option Settings" on page A-12.

Aastra IP Phone UI

Step	Action
1	Press **Options**, and then select **Administrator Menu**.
2	Select **Network Settings**.
3	Select **DHCP Settings**.
4	Select **DHCP User Class**.
5	In the "**DHCP User Class**" field, enter a DHCP User Class to apply to your phones, then press **DONE**. Valid values are up to 64 alpha-numeric characters. For example, "**admin**".
6	Restart the phone for the change to take affect.

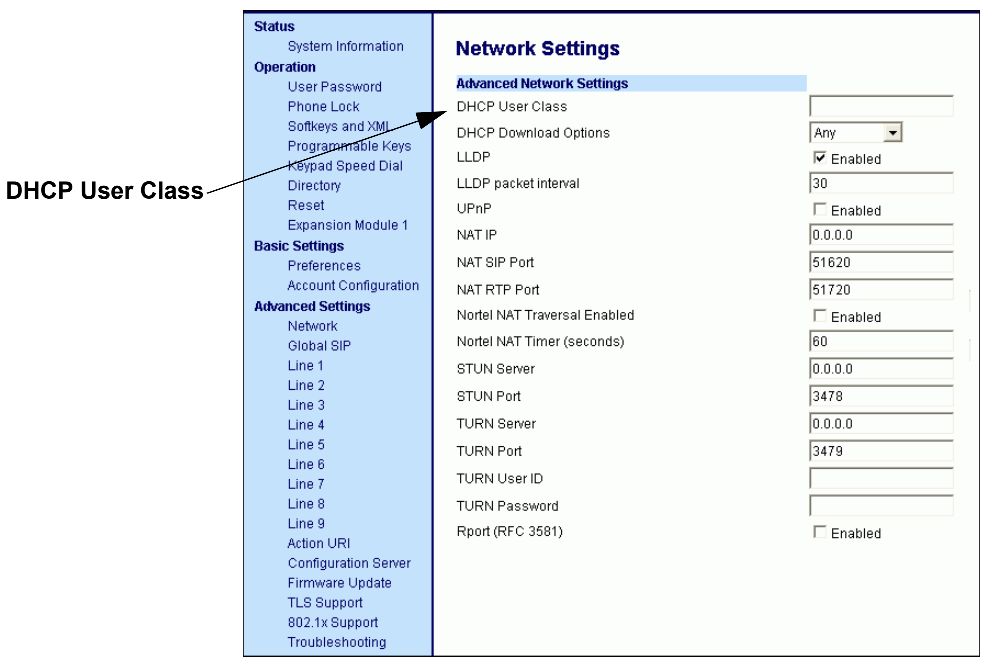
Aastra IP Phone UI

Step	Action
For the 6739i:	
1	Press the **Options** key on the phone to enter the Options List.
2	Press **Advanced**. A keyboard displays.
3	Enter the Administrator password using the keyboard. Default is “**22222**”.
4	Press **Network**.
5	Press **DHCP Settings**.
6	Press **DHCP Download Options**.
7	Select and press the “**Option 77**” value.
8	Press  until the Options List screen displays.
9	Press **Restart** to restart the phone for the change to take affect.

.

Aastra Web UI

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



2	In the “**DHCP User Class**” field, enter a DHCP User Class to apply to your phones. For example, “**admin**”. Valid values are up to 64 alpha-numeric characters.

Aastra Web UI	
3	Click Save Settings to save your changes. **Note:** Entering a value in the “**DHCP User Class**” field requires a reboot of your phone.
4	Click on **Operation->Reset**, and click **RESTART**.

Using Options 159 and 160 on the IP Phone

In addition to DHCP options 43 and 66 already supported on the IP Phones, the phones also support DHCP Options 159 and 160. The IP Phones use the following order of precedence when deriving the configuration server parameters: 43, 160, 159, 66.

In addition, an administrator can override this order of precedence by setting a configuration parameter called, **dhcp config option override** (configuration files), **DHCP Download Options** (Aastra Web UI), or **Download Options** (IP Phone UI). Setting this parameter results in the phone only using the chosen DHCP option and ignoring the other options.

For more information about setting DHCP download preference, see "Configuration Server Download Precedence" on page 4-19..

Warning: Administrators should review the updated IP phone DHCP option precedence order and configuration options to avoid potential impact to existing Aastra IP phone deployments.

Configuring DHCP Download Options on the IP Phones

Use the following procedures to configure DHCP Option Override on the IP Phone.

Configuration Files

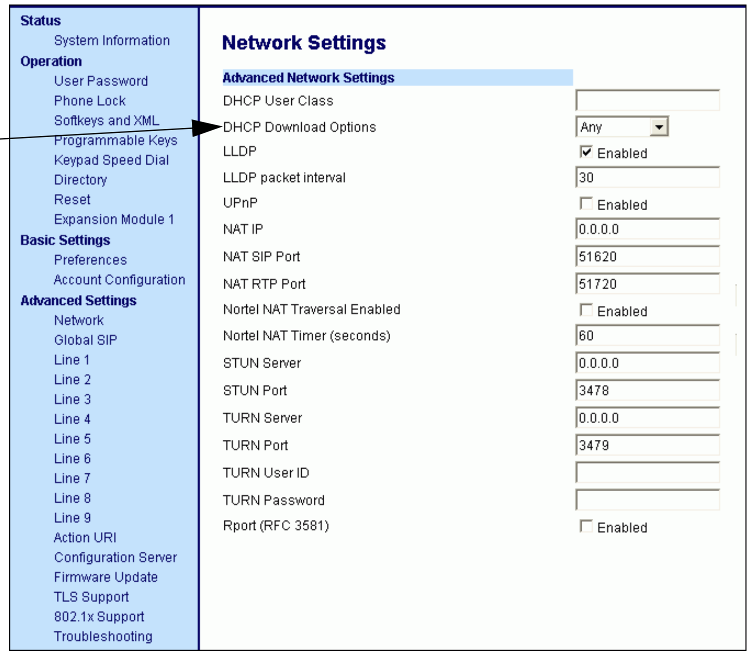
For specific parameters you can set in the configuration files, see Appendix A, the section, "DHCP Option Settings" on page A-12.

Aastra IP Phone UI

Step	Action
1	Press **Options**, and then select **Administrator Menu**.
2	Enter you Administrator password and press **Enter**.
3	Select **Network Settings**.
4	Select **DHCP Settings**.
5	Select **Download Options**. The following list displays: • **Any** (default) - no override - uses normal precedence order of 43, 160, 159, 66. • **Option 43** • **Option 66** • **Option 159** • **Option 160** • **Disabled** (Ignores all DHCP configuration options (43, 66, 159, 160))

Aastra IP Phone UI

Step	Action
6	Choose an option that you want to use to override the DHCP normal precedence order, and press **DONE**.
7	Restart the phone for the selection to take affect.
For the 6739i:	
1	Press the **Options** key on the phone to enter the Options List.
2	Press **Advanced**. A keyboard displays.
3	Enter the Administrator password using the keyboard. Default is “**22222**”.
4	Press **Network**.
5	Press **DHCP Settings**.
6	Press **DHCP Download Options**.
7	Select and press the “**Option 159**” or “**Option 160**” value.
8	Press  until the Options List screen displays.
9	Press **Restart** to restart the phone for the change to take affect.

Aastra Web UI	
1	Click on **Advanced Settings->Network->Advanced Network Settings.** **DHCP Download Options Parameter** 
2	In the "**DHCP Download Options**" field, select an option to use to override the normal precedence order. Valid values are: • **Any** (default) - no override - uses normal precedence order of 43, 160, 159, 66. • **Option 43** • **Option 66** • **Option 159** • **Option 160** • **Disabled** (Ignores all DHCP configuration options (43, 66, 159, 160))
3	Click **Save Settings** to save your changes.
4	Click on **Operation->Reset**, and restart the phone for the changes to take affect.

Configuration Server Download Precedence

An Administrator can set the phone's download precedence to ignore DHCP, **(only during the boot when the remote configuration server is contacted)** and use the following precedence instead:

1. Configuration URI,
2. DHCP, and then
3. Direct configuration.

To configure the download precedence, you use the option value (-1) as the value for the "**dhcp config option override**" parameter in the configuration files. Setting this parameter to "-1" causes all DHCP configuration options to be ignored.

Configuring a Download Precedence Using the Configuration Files

Use the following procedure to configure the DHCP download precedence using the configuration files.

Configuration Files
For specific parameters you can set in the configuration files, see Appendix A, the section, "DHCP Option Settings" on page A-12.

Configuring a Download Precedence Using the IP Phone UI

Use the following procedure to configure a download precedence using the IP Phone UI.

Aastra IP Phone UI

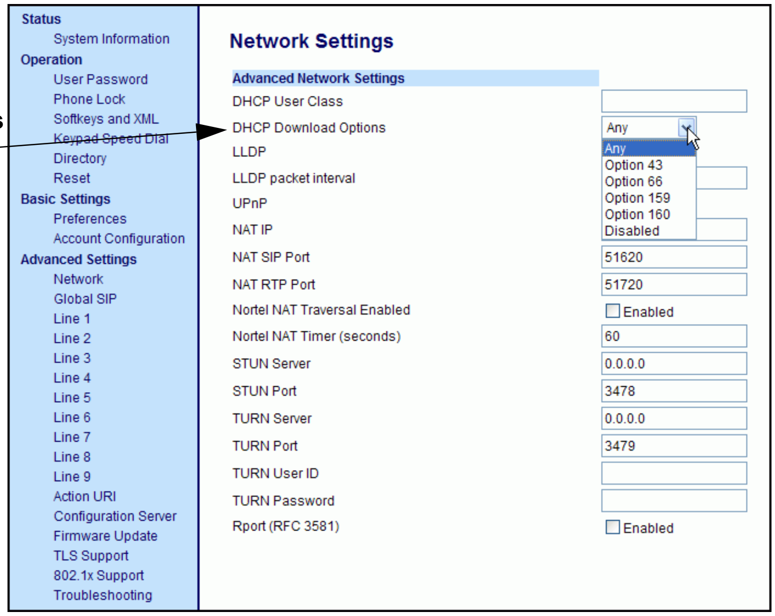
Step	Action
1	Press **Options**, and then select **Administrator Menu**.
2	Enter you Administrator password and press **Enter**.
3	Select **Network Settings**.
4	Select **DHCP Settings**.
5	Select **Download Options**. **Note:** Disabled (Ignores all DHCP configuration options (43, 66, 159, 160))
6	Select the **Disabled** option and press **Enter**. **Note:** The "**Disabled**" download option performs the same function as the "-1" in the configuration files (ignores DHCP options).
7	Restart the phone for the selection to take affect.
For the 6739i:	

Aastra IP Phone UI

Step	Action
1	Press the **Options** key on the phone to enter the Options List.
2	Press **Advanced**. A keyboard displays.
3	Enter the Administrator password using the keyboard. Default is “**22222**”.
4	Press **Network**.
5	Press **DHCP Settings**.
6	Press **DHCP Download Options**.
7	Select and press the “**Disabled**” value. **Note:** Disabled (Ignores all DHCP configuration options (43, 66, 159, 160). This option also performs the same function as the “-1” in the configuration files (ignores DHCP options).
8	Press  until the Options List screen displays.
9	Press **Restart** to restart the phone for the change to take affect.

Configuring a Download Precedence Using the Aastra Web UI

Use the following procedure to configure a download precedence using the Aastra Web UI.

	Aastra Web UI
1	Click on **Advanced Settings->Network->Advanced Network Settings.** **DHCP Download Options Parameter** 
2	In the "**DHCP Download Options**" field, select "**Disabled**" from the list of options. **Note:** In the Aastra Web UI, the "**Disabled**" download option performs the same function as the "-1" in the configuration files (ignores DHCP options).
3	Click [Save Settings] to save your setting.
4	Select **Operation->Reset**, and click **Restart** to reboot the phone.

Multiple DHCP Servers

The IP Phones can receive messages from multiple DHCP servers.

After the phone receives its first DHCP message, it listens for a specific time period, for more DHCP messages. If the first DHCP offer contains configuration server information (Options 43, 66, 159 or 160), then the phone times out and continues using the first DHCP offer, without listening for more DHCP offers. If the first DHCP message contains no configuration server information, the phone continues to listen for other DHCP messages. If the second DHCP message contains configuration server information and other conditions, the phone chooses the second DHCP message over the initial DHCP message.

Note: If the **DHCP Download Options** parameter is enabled with a value (Option 43, Option 66, Option 159, or Option 160), the phone checks the override option setting before timing out.

IMPORTANT NOTE

Users currently using multiple DHCP servers on a single network could be affected by this feature.

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 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 4-104.

You can configure the network settings using the configuration files, the IP phone UI, or the Aastra Web UI.

Errors Messages Display when Incorrect Network Settings Entered

The IP Phone UI AND the Aastra Web UI immediately notify the Administrator if an incorrect value is being entered for the following network parameters in the IP Phone UI and the Aastra Web UI. :

- A 0.0.0.0 entered as values for the **IP Address**, **Subnet Mask** and **Gateway** parameters
- **IP Address** and **Gateway** IP address parameter values entered exactly the same
- **Gateway** IP address and the **IP address** parameter values configured on the same subnet

If you configure the Gateway parameter and the IP Address parameter on the same subnet, the following error message displays:

"*Gateway IP address and the IP address parameter values configured are not on the same subnet*".

Configuration Files

For specific parameters you can set in the configuration files, see Appendix A, the section, "Network Settings" on page A-7.

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 **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.
For the 6739i:	
Note: To manually configure DHCP parameters, DHCP must be disabled on the phone.	
1	Press the **Options** key on the phone to enter the Options List.
2	Press **Advanced**. A keyboard displays.
3	Enter the Administrator password using the keyboard. Default is "**22222**".
4	Press **Network**.
5	Press **IP Address**, enter the IP address of your phone in the text box and press . The IP Address must be entered in the fomat 0.0.0.0; for example, 192.168.0.7.
6	Press the Subnet Mask button, enter the subnet mask address and press . For example, 255.255.0.0.
7	Press **Gateway**, enter the IP address of your gateway in the text box and press . The Gateway must be entered in the fomat 0.0.0.0; for example, 192.168.0.1.
8	If required, press the Primary DNS and/or Secondary DNS buttons, enter the IP address of these servers as applicable and press . The IP addresses must be entered in the format 0.0.0.0. The IP Phone is manually configured.

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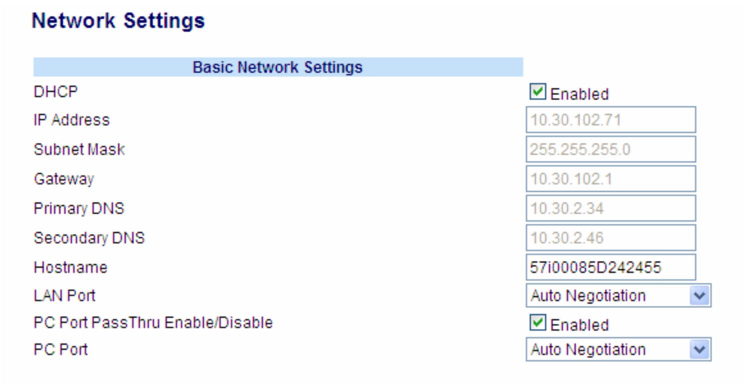
Aastra Web UI

Step	Action
1	Click on **Advanced Settings->Network->Basic Network Settings.** 
2	Enter an IP address of the phone in the **IP Address** field.
3	Enter a subnet mask in the **Subnet Mask** field.
4	Enter a gateway address in the **Gateway** field.
5	Enter a Primary DNS in the **Primary DNS** field, and/or a secondary DNS in the **Secondary DNS** field.
6	Click  to save your settings. The IP phone is manually configured.

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)

➡ **Note:** The PC Port parameters are not applicable to the 6730i IP Phone.

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.

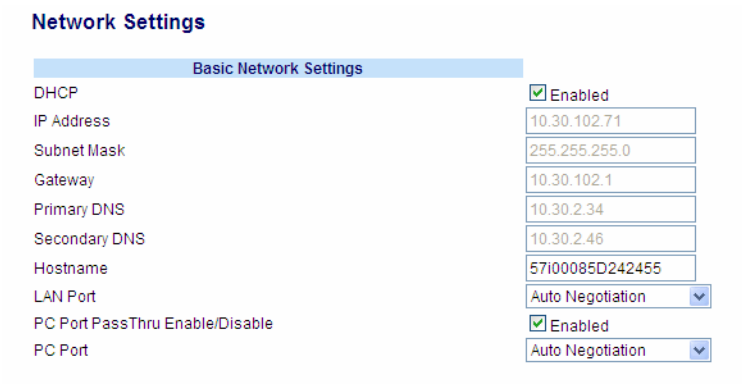
Configuration Files
For specific parameters you can set in the configuration files, see Appendix A, the section, "Network Settings" on page A-7.

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 **Ethernet**.
6	Select **LAN Port Link.**
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**. **Note:** PC Port Link parameters are not applicable to the 6730i IP Phone.
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**.
For the 6739i:	
1	Press the **Options** key on the phone to enter the Options List.
2	Press **Advanced**. A keyboard displays.
3	Enter the Administrator password using the keyboard. Default is "**22222**".

IP Phone UI

Step	Action
4	Press **Network**.
5	Press [down arrow button] to scroll to the next screen.
6	Press **Ethernet & VLAN**.
7	Press **LAN Port**.
8	Select a negotiation method to use on port 0. Valid values are: • Auto • Full 10Mbps • Full 100Mbps • Half 10Mbps • Half 100Mbps Default is **Auto.**
9	Press [back button]
10	Press **PC Port**.
11	Select a negotiation method to use on port 1. Valid values are: • Auto • Full 10Mbps • Full 100Mbps • Half 10Mbps • Half 100Mbps Default is **Auto.**
12	Press [back button] until the Options List screen displays.
13	Press **Restart** to restart the phone for the change to take affect.

Aastra Web UI	
Step	**Action**
1	Click on **Advanced Settings->Network->Basic Network Settings.** 
2	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.**
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.

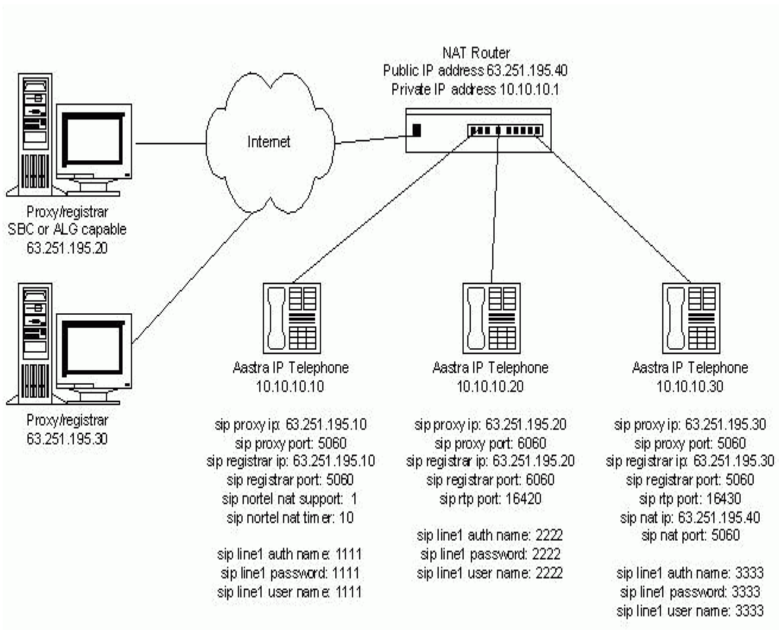
Advanced Network Settings

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

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



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.

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.

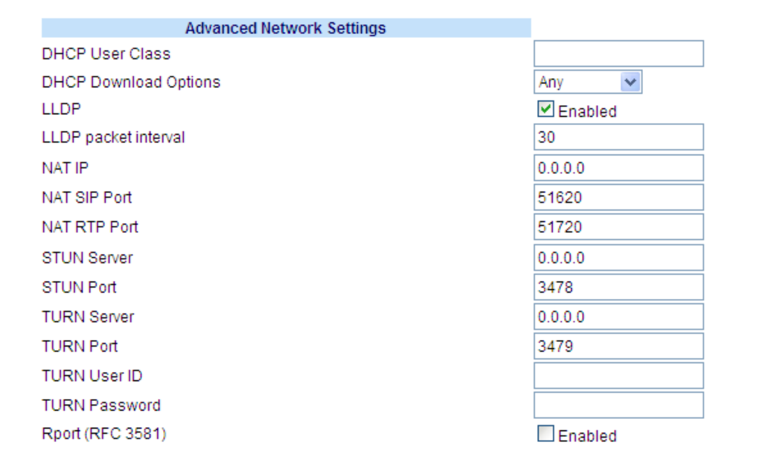
Configuration Files
For specific parameters you can set in the configuration files, see Appendix A, the section, "Network Address Translation (NAT) Settings" on page A-30.

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**.

IP Phone UI	
Step	**Action**
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**.
For the 6739i:	
Note: NAT is disabled by default. Use this procedure to enable NAT if required.	
1	Press the **Options** key on the phone to enter the Options List.
2	Press **Advanced**. A keyboard displays.
3	Enter the Administrator password using the keyboard. Default is "**22222**".
4	Press **Network**.
5	Press to scroll to the next screen.
6	Press the **NAT Settings** button.
7	Press **Static NAT**, enter a public IP address of your NAT device, in dotted-decimal format, in the text box, and press .
8	Press **NAT SIP Port**, and enter the public SIP signalling port number of your NAT device. Default is **51620**.
9	Press **NAT RTP Port**, and enter the RTP Port number of your NAT device. Default is **51720**.
10	Press until the Options List screen displays.
11	Press **Restart** to restart the phone for the change to take affect.

Aastra Web UI

Step	Action
1	Click on **Advanced Settings->Network->Advanced Network Settings.** 
2	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.
3	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.
4	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.
5	Click Save Settings to save your settings.

SIP and TLS Source Ports for NAT Traversal

A System Administrator can configure the SIP and TLS source ports on the IP Phone. Previously, the IP phone used default values (**5060** for UDP/TCP and **5061** for TLS). The two new parameters for configuring the SIP and TLS source ports are:

- **sip local port**
- **sip local tls port**

You can configure the SIP and TLS source ports using the configuration files or the Aastra Web UI. **After configuring these parameters, you must reboot the phone.**

If NAT is disabled, the port number also shows in the VIA and Contact SIP headers.

If you enable NAT, the phone uses the NAT port number (and NAT IP address) in the Via and Contact SIP headers of SIP messages, but still use the configured source port.

Note: 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.If symmetric UDP signaling is disabled, the phone sends from random ports but it listens on the configured SIP local port.

Configuring SIP and TLS Source Ports Using the Configuration Files

You use the following parameters to configure SIP and TLS ports for NAT traversal:

- **sip local port**
- **sip local tls port**

Configuration Files

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

- "Local SIP UDP/TCP Port Setting" on page A-32.
- "Local SIP TLS Port" on page A-32.

Configuring SIP and TLS Source Ports Using the Aastra Web UI

Use the following procedure to configure SIP and TLS source ports using the Aastra Web UI.

Aastra Web UI

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

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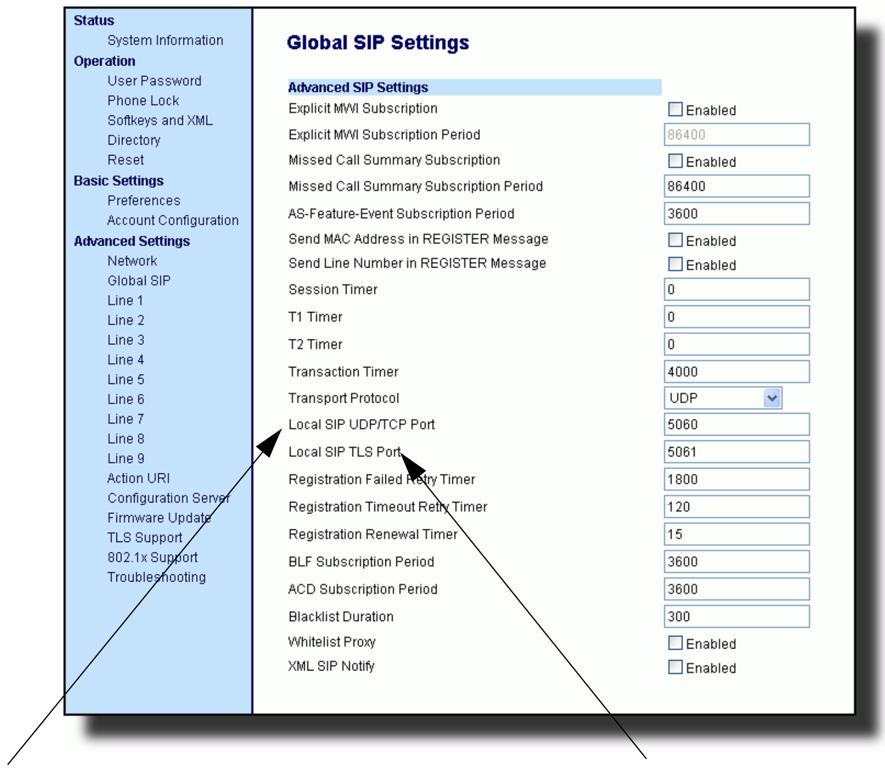

“Local SIP UDP/TCP Port” Parameter **“Local SIP TLS Port” Parameter**

2 The “**Local SIP UDP/TCP Port**” field has a default value of **5060**. Change this value if required to a value greater than 1024 and less than 65535.

Note: It is recommended that you avoid the conflict RTP port range in case of a UDP transport.

3 The “**Local SIP TLS Port**” field has a default value of **5061**. Change this value if required to a value greater than 1024 and less than 65535.

Note: It is recommended that you avoid the conflict with any TCP ports being used. For example: WebUI HTTP server on 80/tcp and HTTPS on 443/tcp.

4 Click Save Settings to save your changes.

STUN and TURN Protocols

The IP phones support the following audio-path NAT Traversal features:

- **Simple Traversal of User Datagram Protocol (UDP) through Network Address Translation (NAT)** or also known as **STUN** (RFC 3489)
  and
- **Traversal Using Relay NAT** or also known as **TURN**

STUN is a protocol that allows the IP phones on a network to discover the presence and types of NATs and firewalls between them and the public Internet. It also provides the ability for the phones to determine the public IP addresses allocated to them by the NAT. STUN works with many existing NATs, and does not require any special behavior from them. As a result, it allows the phones to work through existing NAT infrastructures.

TURN is a protocol that governs the reception of data over a connection by a single communications device operating behind a NAT or firewall. A TURN server relays packets from an external IP address towards the IP phone only if that phone has previously sent a packet through the same TURN server to that particular external IP address.

SIP NAT IP configurations takes precedence over the STUN/TURN configurations. Typically, the STUN/TURN configuration is only used for media (RTP traffic) - not for signaling. (For signaling, you need to enable "Rport" if the NAT device does not recognize SIP. For more information about "Rport", see the section, "RPORT" on page 4-61 of this release note.

The STUN/TURN configuration applies globally on the phone. If you configure both STUN and TURN on the phone, it discovers what type of NAT device is between the phone and the public network. If the NAT device is full cone, restricted cone, or port restricted cone, the phone uses STUN. If the NAT device is symmetric, the phone uses TURN.

If you configure STUN only, the phone uses STUN without the NAT discovery process.

Note: STUN does not work if the NAT device is symmetric.

If you configure NAT only, the phone uses NAT and does not perform the NAT discovery process during startup. TURN is compatible with all types of NAT devices.

Limitations to Using STUN and TURN

- The Firewall type discovery process on the phone is limited to 20 seconds. If the discovery process fails, the STUN server may not be configured correctly.
- When making a new phone call, the phone limits obtaining the port from the STUN/TURN server to 5 seconds. If the call does not go through in 5 seconds, the phone makes the call using the Session Description Protocol (SDP) with a local IP:port.

An Administrator can configure a STUN and/or TURN server on the IP Phones using the configuration files or the Aastra Web UI.

Configuring STUN and TURN Servers Using the Configuration Files

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Note: The NAT IP configuration parameter takes precedence over the STUN and TURN parameters.

Use the following parameters to configure STUN and TURN servers in your network.

- **sip stun ip**
- **sip stun port**
- **sip turn ip**
- **sip turn port**
- **sip turn user**
- **sip turn pass**

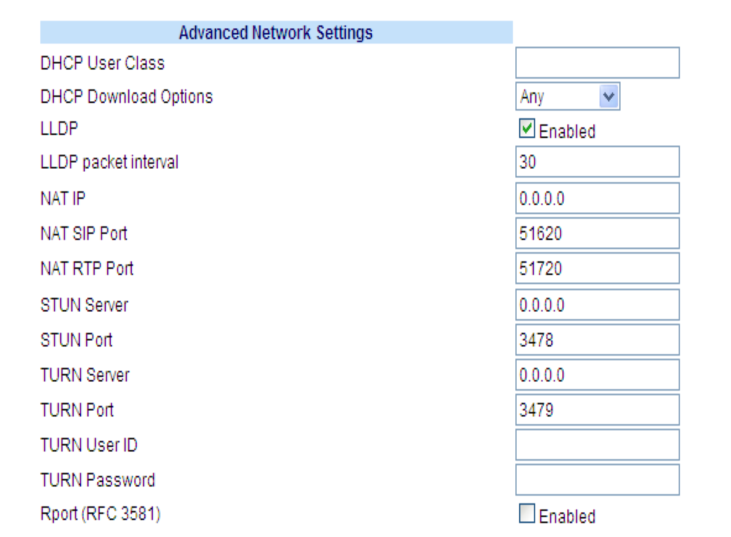
Configuration Files

For specific parameters you can set in the configuration files, see Appendix A, the sections: "SIP STUN Parameters" on page A-33 and "SIP TURN Parameters" on page A-34.

Configuring STUN and TURN Servers Using the Aastra Web UI

Note: The NAT IP configuration parameter takes precedence over the STUN and TURN parameters.

Use the following procedure to configure STUN and TURN servers in your network using the Aastra Web UI..

	Aastra Web UI
1	Click on **Advanced Settings->Network->Advanced Network Settings**. 
Configure STUN on the IP Phone	
2	In the "**STUN Server**" field, enter the IP address and/or qualified domain name of the STUN server in your network. You can enter up to 2 values separated by a comma. The first value is the primary and the second value is the backup. For example: **10.50.103.12, stunbackup.aastra.com.** The STUN configuration is only used for media (RTP traffic) that goes through the server - not for signaling. (For signaling, you need to enable Rport if the NAT device does not recognized SIP). The STUN configuration applies globally to each phone. If you configure both STUN and TURN on the phone, it discovers what type of NAT device is between the phone and the public network. If the NAT device is full cone, restricted cone, or port restricted cone, the phone uses STUN. If the NAT device is symmetric, the phone uses TURN. If you configure STUN only, the phone uses STUN without the NAT discovery process. **Notes**: **1.** The NAT IP configuration parameter takes precedence over the STUN parameters. **2.** STUN does not work if the NAT device is symmetric.

	Aastra Web UI
3	In the "**STUN Port**" field, enter the port number of the STUN server. You can enter up to 2 values separated by a comma. The first value is the primary and the second value is the backup. For example: **3478,3479** Default is **3478**. Range of values are 0 to 65535.
4	Click Save Settings to save your changes.
Configure TURN on the IP Phone	
5	In the "**TURN Server**" field, enter the IP address and/or qualified domain name of the TURN server in your network. You can enter up to 2 values separated by a comma. The first value is the primary and the second value is the backup. For example: **10.50.103.12, turnbackup.aastra.com.** The TURN configuration is only used for media (RTP traffic) that goes through the server - not for signaling. (For signaling, you need to enable Rport if the NAT device does not recognized SIP). The TURN configuration applies globally to each phone. If you configure both STUN and TURN on the phone, it discovers what type of NAT device is between the phone and the public network. If the NAT device is full cone, restricted cone, or port restricted cone, the phone uses STUN. If the NAT device is symmetric, the phone uses TURN. If you configure TURN only, the phone uses TURN with the NAT discovery process. TURN is compatible with all types of NAT devices but can be costly since all traffic goes through a media relay (which can be slow, can exchange more messages, and requires the TURN server to allocate bandwidth for calls). **Note:** The NAT IP configuration parameter takes precedence over the STUN and TURN parameters.
6	In the "**TURN Port**" field, enter the port number of the TURN server. You can enter up to 2 values separated by a comma. The first value is the primary and the second value is the backup. For example: **3479,3480** Default is **3479**. Range of values are 0 to 65535.
7	(Optional) In the "**TURN User ID**" field, enter the username that a user must enter when accessing an account on the TURN server. For example, **0412919146**. Valid values are up to 63 alphanumeric characters.
8	(Optional) In the "**TURN Password**" field, enter the password that a user must enter when accessing an account on the TURN server. For example, **42447208233b8b8b8a234**. Valid values are up to 63 alphanumeric characters.
9	Click Save Settings to save your changes.

Interactive Connectivity Establishment (ICE) Support

The phones now support the Interactive Connectivity Establishment (ICE) Protocol. ICE makes use of the Session Traversal Utilities for NAT (STUN) protocol and its extension, Traversal Using Relay NAT (TURN).

In an ICE environment, two agent endpoints (or two phones communicating at different locations) are able to communicate via the SIP Protocol by exchanging Session Description Protocol (SDP) messages. At the beginning of the ICE process, the agents are ignorant of their own topologies. In particular, they might or might not be behind a NAT (or multiple tiers of NATs).

ICE allows the agents to discover enough information about their topologies to potentially find one or more paths by which they can communicate.

The ICE Protocol is automatically enabled if both STUN and TUNR servers are configured in the network. The following occurs when ICE is used on the phone:

- The TURN address/port is always used as the preferred media address in initial INVITES.
- Media is sent through the TURN server prior to the completion of the ICE connectivity check.
- A Re-INVITE is used to adjust media (if required) after the ICE connectivity check is complete.
- When ICE is enabled, call hold is performed via the sendonly attribute instead of changing the media address to 0.0.0.0.
- The ICE Protocol supports the RTCP SDP attribute (RFC 3605)

STUN and TURN can be enabled using the Aastra Web UI or the configuration files. This automatically enables ICE.

Reference

To configure STUN and TURN (which automatically enables ICE), see "Configuring STUN and TURN Servers Using the Configuration Files" on page 4-37 and "Configuring STUN and TURN Servers Using the Aastra Web UI" on page 4-38.

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 4-104.

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-36.

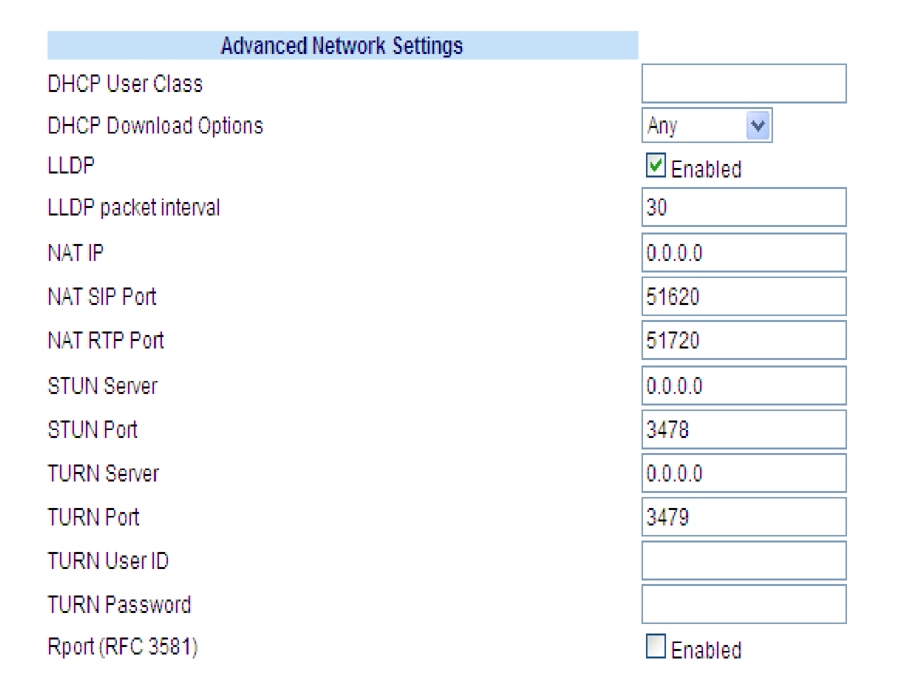
IP Phone UI	
Step	**Action**
1	Press on the phone to enter the Options List.
2	Select **Administrator Menu.**
3	Enter your Administrator password. **Note:** The IP Phones accept numeric passwords only.
4	Select **Configuration Server.**
5	Select **HTTPS Settings.**
Configure HTTPS Client	
6	Select **HTTPS 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**.

For 6739i:

IP Phone UI	
1	Press the **Options** key on the phone to enter the Options List.
2	Press **Advanced**. A keyboard displays.
3	Enter the Administrator password using the keyboard. Default is “**22222**”.
4	Press **Cfg. Svr.**.
5	Press **HTTPS**.
Configure HTTPS Client	

6	Press **HTTPS Client Method.**
7	Press a client method value to use for HTTPS. Valid values are: • **SSL 3.0 (default)** • **TLS 1.0**
8	Press **Done** to save the changes.
Configure HTTPS Server	
9	Press **HTTPS Server.**
10	Enter the IP address of the HTTPS server in the text box. Enabling this feature redirects the HTTP protocol to HTTPS.
11	Press  until the Options List screen displays.
12	Press **Restart** to restart the phone for the change to take affect.

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.

Aastra Web UI	
Step	**Action**
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.

HTTPS Server Certificate Validation

The HTTPS client on the IP Phones support validation of HTTPS certificates. This feature supports the following:

- Verisign, GeoTrust, Thawte, Comodo, CyberTrust signed certificates
- User-provided certificates
- Checking of hostnames
- Checking of certificate expiration
- Ability to disable any or all of the validation steps
- Phone displays a message when a certificate is rejected (except on check-sync operations)

All validation options are enabled by default.

Certificate Management

Aastra Provided Certificates

The phones come with root certificates from Verisign, GeoTrust, Thawte, Comodo, and CyberTrust pre-loaded.

User Provided Certificates

The administrator has the option to upload their own certificates onto the phone. The phone downloads these certificates in a file of .PEM format during boot time after configuration downloads. The user-provided certificates are saved on the phone between firmware upgrades but are deleted during a factory default. The download of the User-provided certificates are based on a filename specified in the configuration parameter, **https user certificates** (**Trusted Certificates Filename** in the Aastra Web UI; User-provided certificates are not configurable via the IP Phone UI).

Note: Certificates that are signed by providers other than Verisign, GeoTrust or Thwate do not verify on the phone by default. The user can overcome this by adding the root certificate of their certificate provider to the use-provided certificate .PEM file.

Certificate Validation

Certificate validation is enabled by default. Validation occurs by checking that the certificates are well formed and signed by one of the certificates in the trusted certificate set. It then checks the expiration date on the certificate, and finally, compares the name in the certificate with the address for which it was connected.

If any of these validation steps fail, the connection is rejected. Certificate validation is controlled by three parameters which you can configure via the configuration files, the IP Phone UI, or the Aastra Web UI:

- **https validate certificates -** Enables/disables validation
- **https validate hostname -** Enables/disables the checking of the certificate commonName against the server name.
- **https validate expires -** Enables/disables the checking of the expiration date on the certificate.

User Interface

Certificate Rejection

When the phone rejects a certificate, it displays, ***"Bad Certificate"*** on the LCD.

An Administrator can configure HTTPS Server Certificate Validation using the configuration files, the IP Phone UI, or the Aastra Web UI.

Configuring HTTPS Server Certificate Validation

Use the following procedures to configure the HTTPS server certificate validation on the IP phones.

Configuration Files
For specific parameters you can set in the configuration files, see Appendix A, the section, "HTTPS Server Certificate Validation Settings" on page A-37.

Aastra IP Phone UI

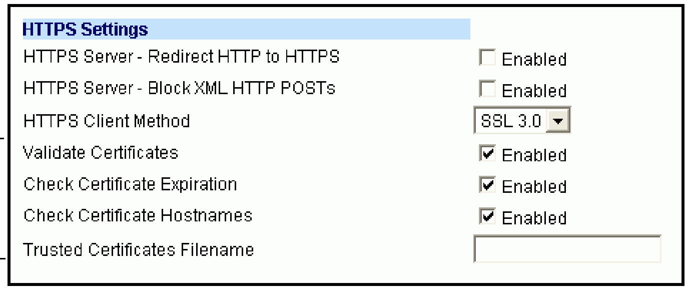
Step	Action
1	Press **Options**, and then select **Administrator Menu**.
2	Select **Configuration Server**.

Aastra IP Phone UI

Step	Action
3	Select **HTTPS Settings->Cert. Validation**. The following list displays: • **Enable** • **Check Expires** • **Check Hostnames**
Enable/Disable HTTPS Server Certificate Validation	
4	Select **Enable**.
5	Press **Change** to toggle the **“Enable”** field to **“Yes” or “No”.** **Note:** If you are using HTTPS as a configuration method, and use a self signed certificate, you must set this field to “**No**” before upgrading to Release 2.3 of the IP Phones.
6	Press **DONE** to save the change and return to the Certificates screen. **Note:** This change is immediately applied after pressing **DONE**.
Enable/Disable HTTPS Validate Certificate Expiration	
7	Select **Check Expires**.
8	Press **Change** to toggle the **“Check Expires”** field to **“Yes” or “No”.** **Notes**: **1.** This change is immediately applied after pressing **DONE**. **2.** If the “**Check Expires**” parameter is set to **Yes**, the clock on the phone must be set for the phone to accept the certificates.
9	Press **DONE** to save the change and return to the Certificates screen. **Note:** This change is immediately applied after pressing **DONE**.
Enable/Disable HTTPS Validate Hostname	
10	Select **Check Hostnames**.
11	Press **Change** to toggle the **“Check Hostnames”** field to **“Yes” or “No”.**
12	Press **DONE** to save the change and return to the Certificates screen. **Note:** This change is immediately applied after pressing **DONE**.
13	Press to exit the Options Menu and return to the idle screen.

For 6739i:

Aastra IP Phone UI	
Step	**Action**
1	Press the **Options** key on the phone to enter the Options List.
2	Press **Advanced**. A keyboard displays.
3	Enter the Administrator password using the keyboard. Default is "**22222**".
4	Press **Cfg. Svr.**.
5	Press **HTTPS**.
Enable/Disable HTTPS Server Certificate Validation	
6	Press **Cert. Validation**, and select **Enable**.
7	Press  to scroll to the next screen.
Enable/Disable HTTPS Validate Certificate Expiration	
8	Press **Check Expires**, and select **Enable**.
Enable/Disable HTTPS Validate Hostname	
9	Press **Check Hostnames**, and select **Enable**.
10	Press  until the Options List screen displays.
11	Press **Restart** to restart the phone for the change to take affect.

	Aastra Web UI
1	Click on **Advanced Settings->Network->HTTPS Settings.** **HTTPS Validation Certificate Parameters** 
2	The **"Validate Certificates"** field is enabled by default. To disable validation of certificates, click the check mark in the box to clear the check mark. When this parameter is enabled, the HTTPS client performs validation on SSL certificates before accepting them. **Notes**: **1.** This parameter is immediately applied after clicking the **SAVE SETTINGS** button. **2. If you are using HTTPS as a configuration method, and use a self signed certificate, you must disable (uncheck) this field before upgrading to Release 2.3 of the IP Phones**.
3	The **"Check Certificate Expiration**" field is enabled by default. To disable validation of certificate expiration, click the check mark in the box to clear the check mark. When this parameter is enabled, the HTTPS client verifies whether or not a certificate has expired prior to accepting the certificate. **Notes**: **1.** This parameter is immediately applied after clicking the **SAVE SETTINGS** button. **2.** If the "**Check Certificates Expiration**" parameter is set to **Yes**, the clock on the phone must be set for the phone to accept the certificates.
4	The **"Check Certificate Hostnames**" field is enabled by default. To disable validation of hostnames, click the check mark in the box to clear the check mark. **Note:** This parameter is immediately applied after clicking the **SAVE SETTINGS** button.

Aastra Web UI	
5	If you require the download of User-provided certificates in a .PEM formatted file, enter the file name in the format <filename.pem> in the "**Trusted Certificates Filename**" field. For example: **trustedCerts.pem** This parameter specifies a file name for a .PEM file located on the configuration server. This file contains the User-provided certificates in PEM format. These certificates are used to validate peer certificates. **Notes**: **1.** You must disable the "**Validate Certificates**" field in order for the phone to accept the User-provided certificates. **2.** This parameter requires you restart the phone in order for it to take affect.
6	Click Save Settings to save your changes.
7	If you entered a filename in the "**Trusted Certificates Filename**" field, click on **Operation->Reset**, and restart the phone for the changes to take affect.

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 pa rticular 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

Parameters in Configuration Files	Parameters in Aastra Web UI
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 4-55. For configuring this feature using the configuration files, see Appendix A, the section, "Virtual Local Area Network (VLAN) Settings" on page A-40.

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 26, 46, and 46 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 4-53).

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
sip	26
rtp	46
rtcp	46

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 26, 46, and 46, respectively. Use the following procedures to change these settings if required.

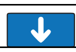
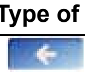
Configuration Files

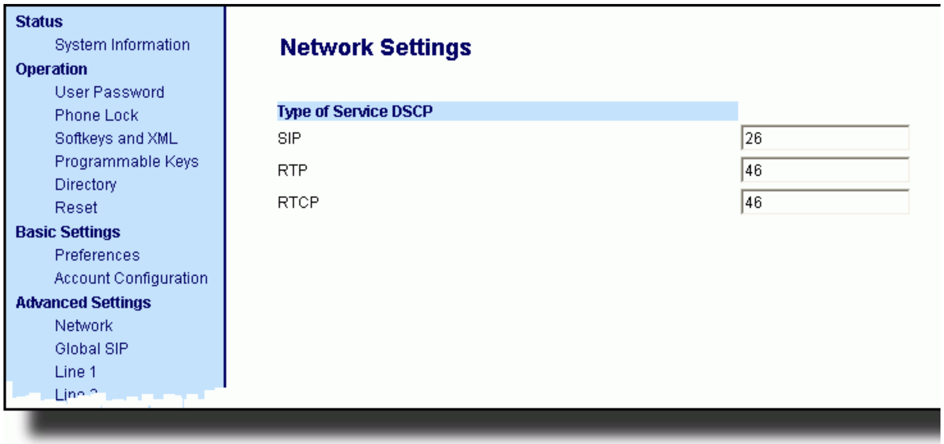
For specific parameters you can set in the configuration files, see Appendix A, the section, "Type of Service (ToS)/DSCP Settings" on page A-47.

IP Phone UI

Step	Action
1	Press on the phone to enter the Options List.
2	Select **Administrator Menu.**

IP Phone UI

Step	Action
3	Enter your Administrator password. **Note:** The IP Phones accept numeric passwords only.
4	Select **Network Settings.**
5	Select **Type of Service DSCP.**
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 4-53 For more information, see the section "Configuring VLAN (optional)" on page 4-55.
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.**
For the 6739i:	
1	Press the **Options** key on the phone to enter the Options List.
2	Press **Advanced**. A keyboard displays.
3	Enter the Administrator password using the keyboard. Default is "**22222**".
4	Press **Network**.
5	Press  to scroll to the next screen.
6	Press **Type of Service DSCP**.
7	Press **Type of Service SIP,** and then enter a value for ToS. Default is **26**.
8	Press **Type of Service RTP**, and then enter a value for ToS. Default is **46**.
9	Press **Type of Service RTCP**, and then enter a value for ToS. Default is **46**.
10	Press  until the Options List screen displays.
11	Press **Restart** to restart the phone for the change to take affect.

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 4-55 For more information, see the section “Configuring VLAN (optional)” on page 4-55.
4	Click Save Settings to save your settings.

Configuring VLAN (optional)

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

Notes:
1. 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.
2. PC Port parameters are not applicable to the 6730i IP Phone.

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-40.

IP Phone UI

Step	Action
1	Press [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 **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.
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

IP Phone UI	
Step	**Action**
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**. **Note:** PC Port parameters are not applicable to the 6730i IP Phone.
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 **4095**. **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). **<u>Example</u>** 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`**
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.**

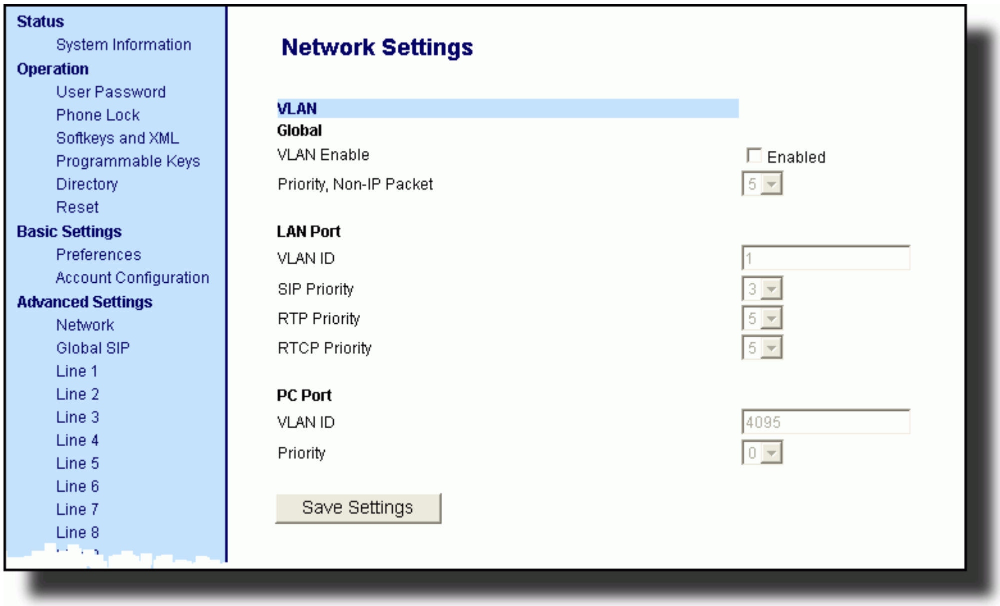
For 6739i:

IP Phone UI	
Step	**Action**
1	Press the **Options** key on the phone to enter the Options List.
2	Press **Advanced**. A keyboard displays.
3	Enter the Administrator password using the keyboard. Default is “**22222**”.
4	Press **Network**.
5	Press ↓ to scroll to the next screen.
6	Press **Ethernet & VLAN.**
To globally enable/disable VLAN and set priority for non-IP packets:	

IP Phone UI

Step	Action
7	Press **VLAN Settings**.
8	Press **VLAN**, and then press **Enable**.
To set VLAN ID and priority for LAN Port (Port 0):	
9	Press **LAN Port VLAN.**.
10	Press **LAN Port VLAN ID** and then enter a value from **1** to **4094** to specify the VLAN ID for the LAN Port. Default is **1**.
11	Press **SIP Priority** and then enter a value from **0** to **7** to specify the SIP priority for the LAN Port. Default is **3**.
12	Press **RTP Priority** and then enter a value from **0** to **7** to specify the RTP priority for the LAN Port. Default is **5**.
13	Press **RTCP Priority** and then enter a value from **0** to **7** to specify the RTCP priority for the LAN Port. Default is **5**.
14	Press  to return to the VLAN screen.
To set VLAN ID and priority for PC Port (Port 1):	
15	Press **PC Port VLAN.**.
16	**Note:** Press **PC Port VLAN ID** and then enter a value from **1** to **4095** for the PC Port VLAN ID. Default is **4095**. 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). **<u>Example</u>** 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`**
17	Press **Priority** and then enter a value from **0** to **7** to specify the PC Port VLAN priority. Default is **0**.
To globally enable/disable VLAN and set priority for non-IP packets:	
18	On the **Network** screen, press the **Ethernet & VLAN** button.
19	Press **VLAN Settings.**
20	Press **LAN Port VLAN.**
21	Press **Other Priority** and enter a value from **0** to **7** to specify the global setting for the LAN port. This value is also applicable to non-IP packets. Default is **5**.
22	Press  until the Options List screen displays.
23	Press **Restart** to restart the phone for the change to take affect.

Aastra Web UI

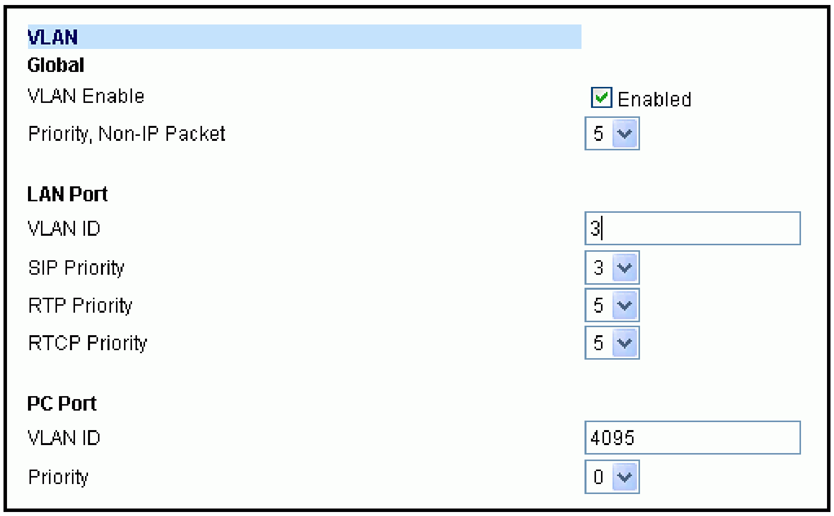
<table>
<tr><th>Step</th><th>Action</th></tr>
<tr><td>1</td><td>Click on Advanced Settings->Network->VLAN.


</td></tr>
<tr><td colspan="2">To globally enable/disable VLAN and set priority for non-IP packets:</td></tr>
<tr><td>2</td><td>Enable VLAN by checking the VLAN Enable field check box. (Disable this field by unchecking the check box).</td></tr>
<tr><td>3</td><td>With VLAN enabled, select the priority (0 to 7) for non-IP packets in the Priority, Non-IP Packet field.</td></tr>
<tr><td colspan="2">To set VLAN ID and priority for the LAN Port (Port 0):</td></tr>
<tr><td>4</td><td>Enter a VLAN ID value from 1 to 4094 in the VLAN ID field. Default is 1.</td></tr>
<tr><td>5</td><td>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</td></tr>
</table>

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 1.

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 

to save your settings.</td></tr>
</table>

RPORT

The Session Initiation Protocol (SIP) operates over UDP and TCP. When used with UDP, responses to requests are returned to the source address from which the request came, and returned to the port written into the topmost “Via” header of the request. However, this behavior is not desirable when the client is behind a Network Address Translation (NAT) or firewall.

A parameter created for the “Via” header called “**Rport**” in RFC 3581, allows a client to request that the server send the response back to the source IP address **and** the port from which the request came.

When you enable “Rport, the phone always uses symmetric signaling (listens on the port used for sending requests.)

Note: Configuring the Rport parameter is recommened for clients behind a Network Address Translation (NAT) or firewall since this parameter allows a client to request that the server send the response back to the source IP address and the port from which thre request came.

An Administrator can configure “**Rport**” using the configuration files or the Aastra Web UI.

Configuring Rport Using the Configuration Files

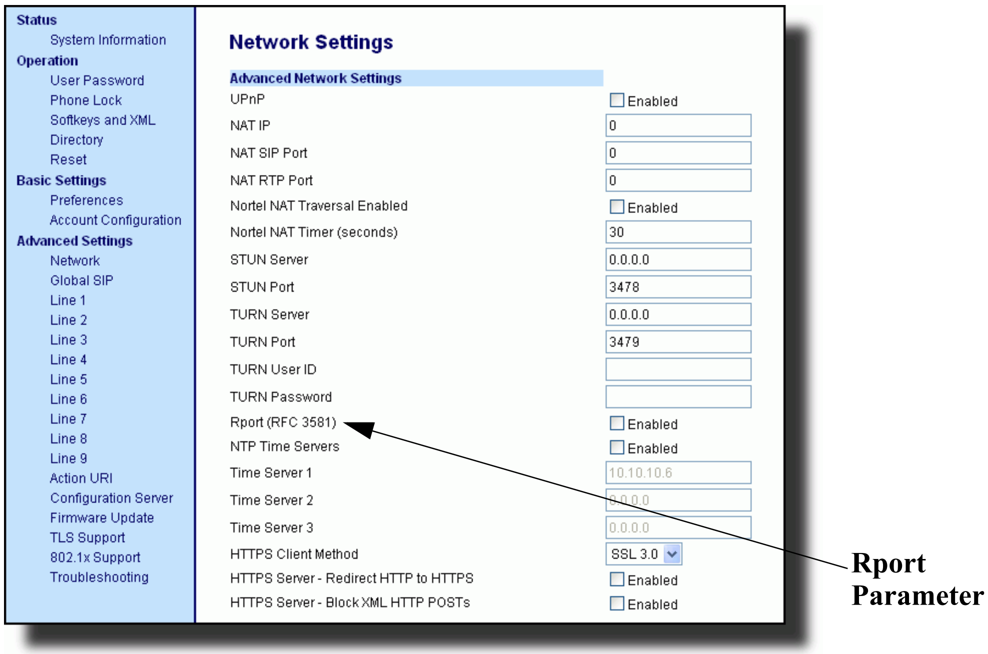
Use the following procedures to configure Rport on your phone.

Configuration Files
For specific parameters you can set in the configuration files, see Appendix A, the section, “Rport Setting” on page A-31.

Configuring Rport Using the Aastra Web UI

Use the following procedure to configure Rport on your phone using the Aastra Web UI.

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



</td></tr>
<tr><td>2</td><td>In the "Advanced Network Settings" section, enable the "Rport (RFC3581" field by checking the check box. (Disable Rport by unchecking the box).

“Rport” in RFC 3581, allows a client to request that the server send the response back to the source IP address and the port from which the request came.</td></tr>
<tr><td>3</td><td>Click 

to save your changes.</td></tr>
</table>

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.

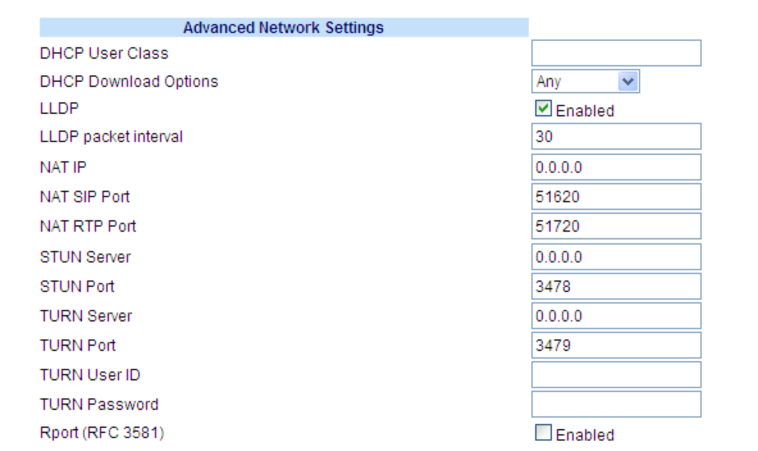
Configuration Files
For specific parameters you can set in the configuration files, see Appendix A, the section, "Time Server Settings" on page A-55.

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

IP Phone UI

Step	**Action**
1	Press 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.)
8	Enter the IP Address (in dotted decimal) or qualified domain name for the Time Server.
9	Press **Done** to save the change.
For the 6739i:	
1	Press the **Options** key on the phone to enter the Options List.
1	Press **Set Time**.
2	Press **Timer Server 1, Time Server 2,** or **Time Server 3**. **Note:** The Time Servers are disabled by default.
3	Enter the IP Address (in dotted decimal) or qualified domain name for the Time Server.
4	Press to return to the idle screen.

Use the following procedure to enable/disable and configure the NTP Servers 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 Advanced Settings->Network->Advanced Network Settings.

</td></tr>
<tr><td>2</td><td>Enable the "NTP Time Servers" field by checking the check box.
(Disable this field by unchecking the box). Default is disabled.</td></tr>
<tr><td>3</td><td>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.</td></tr>
<tr><td>4</td><td>Click Save Settings to save your changes.</td></tr>
</table>

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 4-66.

For more information bout Advanced SIP Settings, see "Advanced SIP Settings (optional)" on page 4-83.

Basic SIP Settings

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.

The IP phones allow you to define different SIP lines with the same account information (i.e., same user name) but with different registrar and proxy IP addresses. This feature works with Registration, Subscription, and Notify processing. This feature also works with the following types of calls: incoming, outgoing, Broadsoft Shared Call Appearance (SCA), Bridged Line Appearance (BLA), conference, transfer, blind transfer.

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 • User Name • Display Name • Authentication Name • Password	• Screen Name • Screen Name 2 • Phone Number • Caller ID • Authentication Name • Password • BLA Number • Line Mode • Call Waiting (see Chapter 5)	• sip screen name • sip screen name 2 • sip user name • sip display name • sip auth name • sip password • sip bla number • sip mode • call waiting (see Chapter 5 • sip vmail
SIP Global Network Parameters		
• Proxy Server • Proxy Port • Registrar Server • Registrar Port	• Proxy Server • Proxy Port • Backup Proxy Server • Backup Proxy Port • Outbound Proxy Server • Outbound Proxy Port • Backup Outbound Proxy • Backup 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 backup outbound proxy • sip backup 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 5-327.

SIP Per-Line Parameters

IP Phone UI Parameters	Aastra Web UI Parameters	Configuration File Parameters
SIP Per-Line Authentication Parameters		
• Screen Name • User Name • Display Name • Auth Name • Password	• Screen Name • Screen Name 2 • Phone Number • Caller ID • Authentication Name • Password • BLA Number • Line Mode • Call Waiting (see Chapter 5	• 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 call waiting (see Chapter 5 • sip lineN vmail
SIP Per-Line Network Parameters		
• Proxy Server • Proxy Port • Registrar Server • Registrar Port	• Proxy Server • Proxy Port • Backup Proxy Server • Backup Proxy Port • Outbound Proxy Server • Outbound Proxy Port • Backup Outbound Proxy Server • Backup 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 backup outbound proxy • sip lineN backup 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 5-327.

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" on page 5-282.

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 Outbound Support

The IP Phones support draft-ietf-sip-outbound-15. That specification describes how a SIP User Agent (UA) behind a firewall, reuses an existing connection (usually the REGISTER outbound connection) for the inbound request if the proxy supports it. The UA uses keep-alive packets to monitor the connection status.

An Administrator can enable or disable this feature using the following parameter in the configuration files:

- **sip outbound support**

Note: TLS and TCP both support this feature. If the Global SIP parameter "Persistent TLS" is set on the phone, then only one TLS persistent connection can be established since the phone uses the local port 5061 for connection. If the Global SIP parameter "TLS" is set on the phone, more than one connection can be setup since the phone uses a random local port for connection.

Enablling/Disabling SIP Outbound Draft 15 Support.

Use the following procedure to enable/disable SIP outbound Draft 15 support.

Configuration Files

For the specific parameter you can set in the configuration files, see Appendix A, the section, "SIP Outbound Support" on page A-72.

Limitations

The following are limitations to this feature:

- SIP outbound on the phones is only supported for TLS and TCP based messages. UDP is not supported.
- The "keep-alive" only works with REGISTER connections.

Backup Outbound Proxy and Failover Support

The IP phones support a backup outbound proxy and failover. This feature provides the following:

- The ability to specify a backup outbound proxy.
- The ability to support SIP outbound on all connection types.
- The ability to configure the SIP outbound keep alive timer.
- The ability to reestablish failed outbound connections in the background.
- The ability to support the DNS Cache Time-to-Live (TTL) requirements

Using this feature depends on the SIP network settings on your phone. The following table identifies network configuration scenarios, and the method by which this specific feature works in each scenario.

IF	THEN
SIP Outbound Disabled and	
backup proxy and backup registrar configured,	• All invite, register, and subscribe requests attempt to use the primary proxy/registrar first • If the primary registrar fails, the phone registers to the backup proxy. • If the backup proxy fails, the phone registers using the Address of Record (AOR) of the backup proxy, and moves all subscriptions to the backup proxy. • When the primary registrar comes back online, the phone registers to it using the currently active AOR. • When the primary proxy comes back online the phone registers with the primary AOR to the currently active registrar and moves all subscriptions to the primary proxy.
backup proxy, backup registrar, and backup outbound proxy configured,	• All invite, register, and subscribe requests attempt to use the primary proxy/registrar first. • If any connection fails, the phone registers the backup AOR on the backup registrar. It moves all subscriptions to the backup proxy. • When the primary is functional again, registration and subscriptions are moved back to the primary proxy/registrar.
backup outbound proxy configured only,	• All invite, register and subscribe requests are sent through the primary outbound proxy first. • If the primary proxy fails, the phone performs registration and subscriptions through the backup outbound proxy. • When the primary proxy comes back online, the registrations and subscriptions are performed again through the primary outbound proxy.

SIP Outbound Enabled and	
backup proxy and backup registrar configured,	• Establishes flow to the primary proxy and registrar. • If the flow to the primary registrar fails, the phone: – establishes flow to the backup registrar. – registers to the backup registrar. – attempts to reestablish flow to the primary registrar in the background. – When the primary registrar comes back up, the phone unregisters from the backup and registers with the primary. • If the flow to the primary proxy fails, the phone: – establishes flow to the backup proxy. – registers the new AOR with the active registrar. – moves subscriptions to the backup proxy. – attempts to reestablish the flow to the primary proxy in the background. • When the flow to the primary proxy is reestablished, the phone: – registers the primary AOR to the active registrar. – moves subscriptions to the primary proxy. – unregisters/unsubscribes from the backup proxy/registrar.
backup proxy, backup registrar, and backup outbound proxy configured, **Note**: This configuration assumes that the outbound proxy is maintaining its own outbound connections to the proxy/registrar.	• Establishes a flow to the primary outbound proxy. • If the flow fails, the phone: – establishes the flow to the backup proxy. – registers the backup AOR to the backup registrar. – moves subscriptions to the backup proxy. – attempts to reestablish connection to the primary outbound proxy in the background. • When the flow to the primary proxy is reestablished, the phone: – registers the primary AOR to the primary registrar. – moves the subscriptions to the primary proxy. – unregisters/unsubscribes from the backup proxy/registrar.
backup outbound proxy configured only,	• Establishes a flow to the primary outbound proxy. • If the flow fails, the phone: – establishes the flow to the backup proxy. – registers the backup AOR to the backup registrar. – moves subscriptions to the backup proxy.

Configuring a Backup Outbound Proxy

To configure this feature an Administrator can set the following parameters in the configuration files or the Aastra Web UI:

Parameter	**Aastra Web UI Configuration**	**Configuration File Configuration**
sip outbound support	-	✔
sip symmetric udp signaling	-	✔
sip transport protocol (	✔	✔
Global Parameters		
sip backup outbound proxy	✔	✔
sip backup outbound proxy port	✔	✔
Per-Line Parameters		
sip lineN backup outbound proxy	✔	✔
sip lineN backup outbound proxy port	✔	✔

Note: The "sip outbound support", "sip symmetric udp signaling", and "sip transport protocol" parameters are existing parameters on the phone. For more information about these parameters, see Appendix A, "SIP Outbound Support", "Symmetric UDP Signaling Setting", and "Advanced SIP Settings".

Use the following procedure to configure backup outbound proxies..

Configuration Files

For the specific parameter you can set in the configuration files, see Appendix A, the section, "Backup Outbound Proxy (Global Settings)" on page A-85 and "Backup Outbound Proxy (Per-line Settings)" on page A-96.

Limitations

The following are limitations with this feature:

- Keep-alive mechanisms shall be limited to IPv4 only.
- Per M5T, RFC5686 is not fully supported although the draft upon which it was based (draft-ietf-sip-outbound-15) is supported.

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**

Contact Header Matching

When sending SIP packets, the IP Phones observe the Contact header by matching the username, domain name, port, and transport per SIP RFC 3261.This is called "strict SIP Contact header matching." However, in specific networks (such as behind some SOHO routers), the phone registers with its private address in the Contact, but when the response is sent back, the router maintains the public side IP address in the Contact header. This causes a non-matching Contact header and the phone does not accept the new registration expiry timer.

You can set the parameter, "**sip contact matching**", which allows the Administrator to specify the method used by the phone to match the Contact Header. This parameter is available via the configuration files only.

Enabling/Disabling the "Contact Header Matching" Feature

Use the following procedure to enable/disable the "Contact Header Matching" feature.

Configuration Files
For the specific parameter you can set in the configuration files, see Appendix A, the section, "Contact Header Matching" on page A-73.

Configuring Basic SIP Authentication Settings

You can configure SIP authentication 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-74 or "SIP Basic, Per-Line Settings" on page A-86. For specific parameters you can set in the configuration files for call waiting, see the section, "Call Waiting Settings" on page A-78 or "SIP Per-Line Call Waiting Setting" on page A-91.

Reference

For more information about setting the call waiting parameters, see Chapter 5, the section, "Call Waiting" on page 5-66. Call Waiting cannot be set via the IP Phone UI.

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

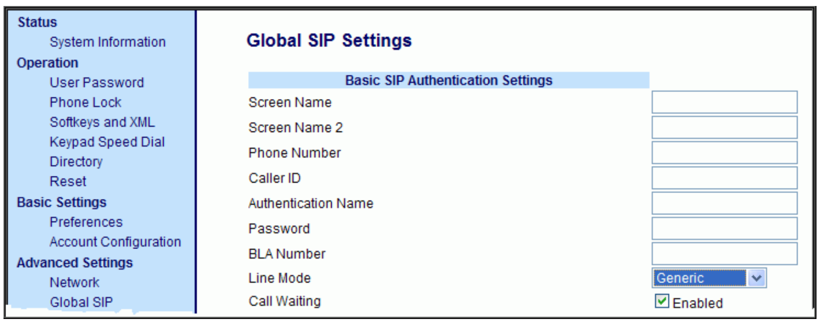
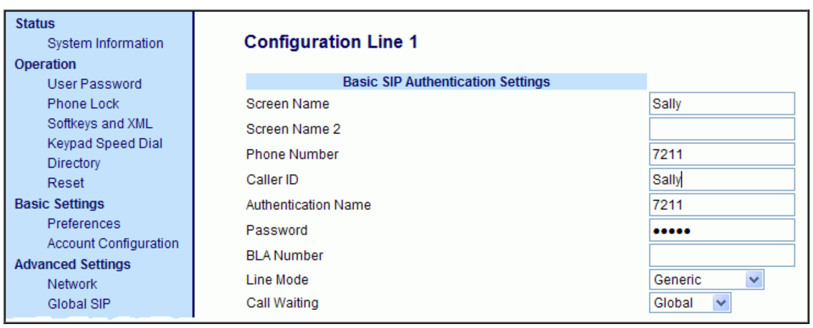
IP Phone UI

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 **User Name** to enter the username that appears in the name field of the SIP URI. This user name is also used for registering the phone at the registrar. **Note:** The IP phones allow usernames containing dots ("."). You can also enter the same user name for different registrar and proxy IP addresses.
6	Press **Done** to save the changes.
7	Select **Display Name** to enter the name used in the display name field of the "From SIP" header field.
8	Press **Done** to save the changes.
9	Select **Screen Name** and enter the name to display on the idle screen.
10	Press **Done** to save the changes.
11	Select **Authentication Name** to enter the authorization name used in the username field of the Authorization header field of the SIP REGISTER request.
12	Press **Done** to save the changes.
13	Select **Password** to enter the password used to register the IP phone with the SIP proxy**.** **Note:** The IP phones accept numeric passwords only.
14	Press **Done** (3 times) to save the changes. **Note:** The session prompts you to restart the IP phone to apply the configuration settings
15	Select **Restart.**
For the 6739i:	
1	Press the **Options** key on the phone to enter the Options List.
2	Press **Advanced**. A keyboard displays.
3	Enter the Administrator password using the keyboard. Default is "**22222**".
4	Press **SIP**.
5	Press **User Name** and enter the username that appears in the name field of the SIP URI. This user name is also used for registering the phone at the registrar. **Note:** The IP phones allow usernames containing dots ("."). You can also enter the same user name for different registrar and proxy IP addresses.
6	Press to scroll to the next screen.
7	Press **Display Name** and enter the name used in the display name field of the "From SIP" header.
8	Press **Screen Name** and enter the name to display on the idle screen.
9	Press **Auth. Name** and enter the authorization name used in the username field of the Authorization header field of the SIP REGISTER request.
10	Press **Password** and enter the password used to register the IP phone with the SIP proxy**.** **Note:** The IP phones accept numeric passwords only.

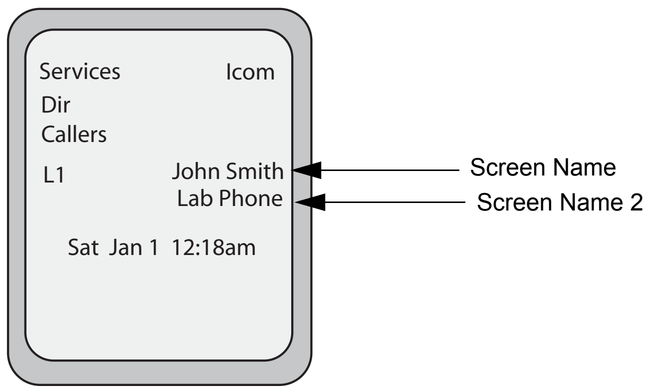
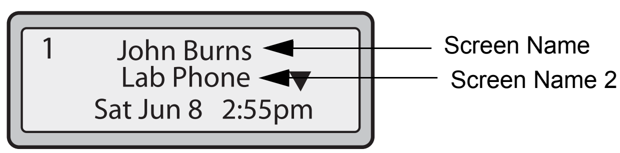
IP Phone UI

Step	Action
11	Press  until the Options List screen displays.
12	Press **Restart** to restart the phone for the change to take affect.

Aastra Web UI

Step	Action
1	For global configuration, click on **Advanced Settings->Global SIP->Basic SIP Authentication Settings.** 
	Or, for per-line configuration, click on **Advanced Settings->Line *N* (1-9)0->Basic SIP Authentication Settings**. 
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** **8 and 11-Line** **LCD Displays**  **Figure 2** **3-Line** **LCD Displays** 
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. For more information about setting the BLA on the phone, see Chapter 5, the section, “Bridged Line Appearance (BLA)” on page 5-195.
9	In the "**Line Mode**" field, select "Generic" for normal mode, "BroadSoft SCA" for a BroadWorks network.
Configure Global Call Waiting	
10	The "**Call Waiting**" field is enabled by default. To disable call waiting on a global basis, uncheck this box. For more information about setting the call waiting parameters, see Chapter 5, the section, “Call Waiting” on page 5-66.

Aastra Web UI	
Step	**Action**
Configure Per-Line Call Waiting	
11	The "**Call Waiting**" field is set to "**Global**" by default. To enable call waiting for a specific line, select "**enabled**" from the list in this field. To disable call waiting for a specific line, select "**disabled**" from the list in this field. For more information about setting the call waiting parameters, see Chapter 5, the section, "Call Waiting" on page 5-66.
12	Click Save Settings to save your changes.

Configuring Basic SIP Network Settings (optional)

You can configure SIP network 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-74 or "SIP Basic, Per-Line Settings" on page A-86.

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

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.**
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**.

IP Phone UI

Step	Action
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. **Note:** The session prompts you to restart the IP phone to apply the configuration settings.
16	Select **Restart.**

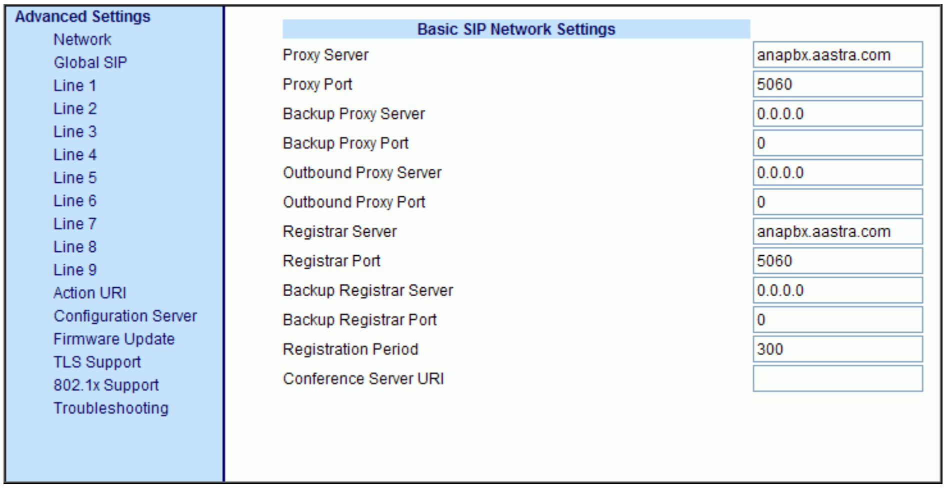
For the 6739i:

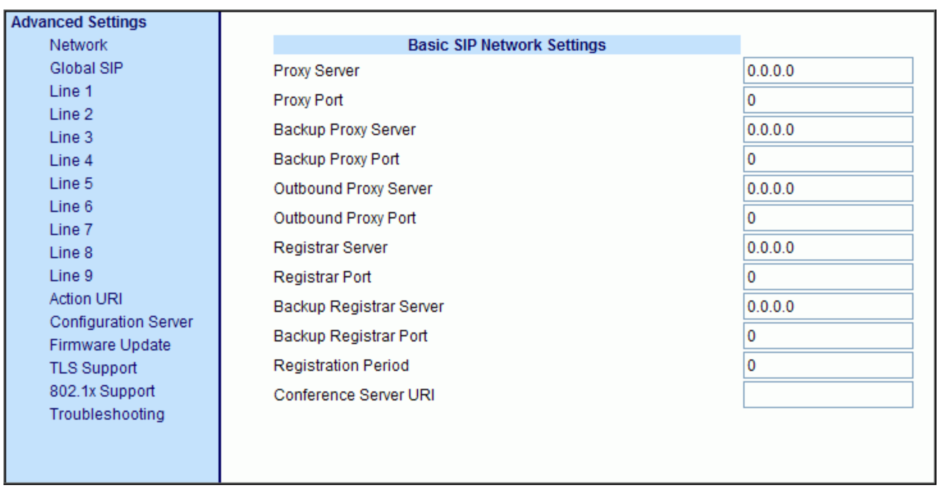
Note: You can set global configuration only using the IP Phone UI.

IP Phone UI

Step	Action
1	Press the **Options** key on the phone to enter the Options List.
2	Press the **Advanced** button. A keyboard displays.
3	Enter the Administrator password using the keyboard. Default is "**22222**".
4	Press the **SIP** button.
Configuring Proxy IP and Proxy Port	
5	Press **Proxy Server** and enter an IP address or fully qualified host name in the **Proxy Server** field. Default is **0.0.0.0**.
6	Press **Proxy Port** and enter a Proxy Port number in the **Proxy Port** field for accessing the SIP proxy server. For example, **5060**. Default is **0**.
Configuring Registrar IP and Registrar Port	
7	Press **Registrar Server** and 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.
8	Press **Registrar Port** and enter a Registrar Port number in the **Registrar Port** field for accessing the SIP registrar server. For example, **5060**. Default is **0**.
9	Press  until the Options List screen displays.
10	Press **Restart** to restart the phone for the change to take affect.

Aastra Web UI

<table>
<tr><th>Step</th><th>Action</th></tr>
<tr><td>1</td><td>For global configuration, click on Advanced Settings->Global SIP->Basic SIP Network Settings.


</td></tr>
<tr><td></td><td>Or, for per-line configuration, click on Advanced Settings->Line N (1-9)->Basic SIP Network Settings.


</td></tr>
<tr><td>2</td><td>In the "Proxy Server" field, enter an IP address or fully qualified host name of the SIP proxy server.</td></tr>
<tr><td>3</td><td>In the "Proxy Port" field, enter a port number for accessing the SIP proxy server.</td></tr>
<tr><td>4</td><td>In the "Backup Proxy Server" field, enter an IP address or fully qualified host name for the backup proxy server.</td></tr>
<tr><td>5</td><td>In the "Backup Proxy Port" field, enter a port number for accessing the backup proxy server.</td></tr>
</table>

Aastra Web UI

Step	Action
6	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.
7	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.
8	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.
9	In the "**Registrar Port**" field, enter the port number associated with the Registrar.
10	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.
11	In the "**Backup Registrar Port**" field, enter the port number associated with the backup registrar.
12	In the "**Registration Period**" field, enter the requested registration period, in seconds, from the registrar.
13	To enter a value in the "**Conference Server URI**" field, see Chapter 5, the section, "Centralized Conferencing (for Sylantro and Broadsoft Servers)" on page 5-327.
14	Click Save Settings to save your changes.

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
MWI for BLA Account	sip mwi for bla account (see Chapter 5)
Missed Call Summary Subscription (global) Missed Call Summary Subscription (per-line) (see Chapter 6) Missed Call Summary Subscription Period (see Chapter 6)	sip missed call summary subscription (global) sip lineN missed call summary subscription (per-line) (see Chapter 6) sip missed call summary subscription period (see Chapter 6)
AS-Feature-Event Subscription (global) AS-Feature-Event Subscription (per-line) (see Chapter 6) AS-Feature Event Subscription Period (see Chapter 6)	sip as-feature-event subscription (global) sip lineN as-feature-event subscription (per-line) (see Chapter 6) sip as-feature-event subscription period (see Chapter 6)
Send MAC Address in REGISTER Message (see Chapter 6) Send Line Number in REGISTER Message (see Chapter 6)	sip send mac (see Chapter 6) sip send line (see Chapter 6)
Session Timer	sip session timer
T1 Timer T2 Timer	sip T1 timer sip T2 timer
Transaction Timer	sip transaction timer
Transport Protocol	sip transport protocol
Local SIP UDP/TCP Port (see page 4-34)	sip local port (see page 4-34)
Local SIP TLS Port (see page 4-34)	sip local tls port (see page 4-34)
Registration Failed Retry Timer Registration Timeout Retry Timer Registration Renewal Timer	sip registration retry timer sip registration timeout retry 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)
BLA 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)
XML SIP Notify (see Chapter 6)	XML SIP Notify (see Chapter 6)

Reference

Refer to Appendix A,"Advanced SIP Settings" on page A-103 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."

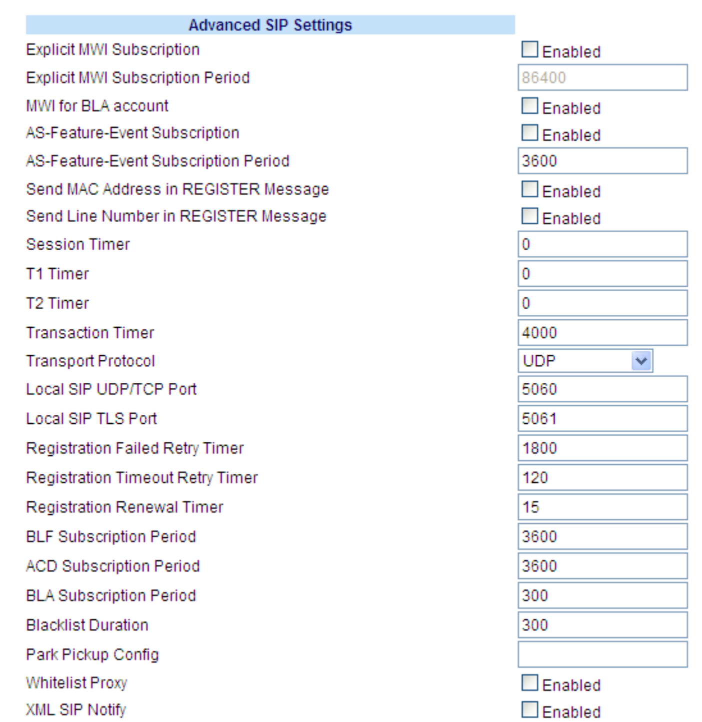
Configuring Advanced SIP Settings

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

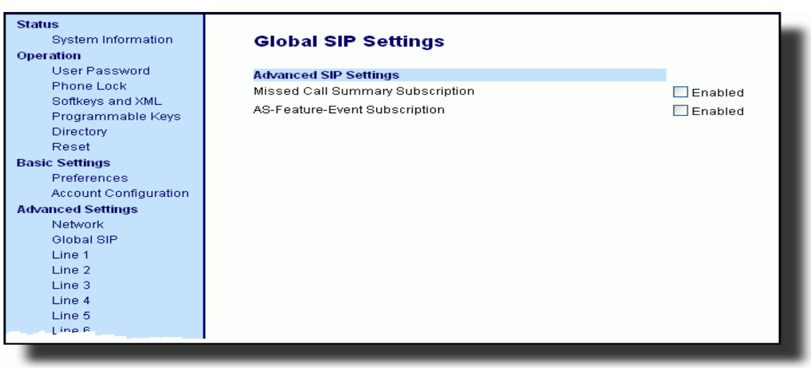
Configuration Files
For specific parameters you can set in the configuration files, see Appendix A, the section, "Advanced SIP Settings" on page A-103.

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.



</td></tr>
<tr><td>2</td><td>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.</td></tr>
<tr><td>3</td><td>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.</td></tr>
</table>

Aastra Web UI	
Step	**Action**
4	Enable the "**MWI for BLA Account**" to enable or disable a BLA configured line to send an MWI SUBSCRIBE message for the BLA account. **Notes**: 1. If you change the setting on this parameter, you must reboot the phone for it to take affect. 2. Both the "sip explicit mwi subscription" and "sip mwi for bla account" parameters must be enabled in order for the MWI subscription for BLA to occur. 3. The MWI re-subscription for the BLA account uses the value set for the "sip explicit mwi subscription period" parameter to re-subscribe. 4. Whether or not the "sip mwi for bla account" parameter is enabled, the priority for displaying MWI does not change.
5	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, the section, "Missed Call Summary Subscription" on page 6-13. **Note:** The "Missed Call Summary Subscription" feature is configurable on a global or per-line basis.
6	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, the section, "Missed Call Summary Subscription" on page 6-13. **Note:** The "Missed Call Summary Subscription Period" is configurable on a global basis only.
7	Enable the "**AS-Feature-Event Subscription**" field by checking the check box. (Disable this field by unchecking the check box. Default is disabled). This feature enables or disables the specified line with the BroadSoft's server-side DND, CFWD, or ACD features. For more information about this feature, see Chapter 6, the section, "As-Feature-Event Subscription" on page 6-16. **Note:** The "AS-Feature-Event Subscription " feature is configurable on a global or per-line basis.
8	If you enable the "**AS-Feature-Event Subscription**" field, then in the "**AS-Feature-Event Subscription Period**" field, enter the amount of time, in seconds, between resubscribing. If the phone does not resubscribe in the time specified for this parameter, it loses subscription. Default is 3600. For more information about this feature, see Chapter 6, the section, "As-Feature-Event Subscription" on page 6-16.

Aastra Web UI

Step	Action
9	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, the section, "TR-069 Support" on page 6-5. **Note:** The "AS-Feature-Event Subscription Period" feature is configurable on a global basis only
10	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.
11	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.
12	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.
13	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 4-94 and Chapter 6, the section, "Transport Layer Security (TLS)" on page 6-26.
14	In the "**Local SIP UDP/TCP Port"** field, specify the local source port (UDP/TCP) from which the phone sends SIP messages. Default is 5060. For more information about this feature, see the section, "SIP and TLS Source Ports for NAT Traversal" on page 4-34.
15	In the "**Local SIP TLS Port"** field, specify the local source port (SIPS/TLS) from which the phone sends SIP messages. Default is 5061. For more information about this feature, see the section, "SIP and TLS Source Ports for NAT Traversal" on page 4-34.
16	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.
17	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.

Aastra Web UI

Step	Action
18	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.
19	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 5-155.
20	(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 5-173.
21	The "**BLA Subscription Period**" field is enabled by default with a value of 300 seconds. This feature sets the amount of time, in seconds, that the phone waits to receive a BLA subscribe message from the server. If you specify zero (0), the phone uses the value specified for the BLA expiration in the subscribe message received from the server. If no value is specified, the phone uses the default value of 300 seconds. For information about setting the "BLA Subscription Period", see Chapter 5, the section, "BLA Subscription Period" on page 5-173.
22	(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 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. For information about setting the "Blacklist Duration", see Chapter 6, the section, "Blacklist Duration" on page 6-22.
23	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 6-24.

Aastra Web UI

Step	Action
24	Enable the "**XML SIP Notify**" field by checking the check box. (Disable this field by unchecking the check box. Default is disabled). Enabling this parameter allows the phone to accept or reject an aastra-xml SIP NOTIFY message. **Note:** To ensure the SIP NOTIFY is coming from a trusted source, it is recommended that you enable the Whitelist feature (**Whitelist Proxy** parameter) on the IP phone. If enabled, and the phone receives a SIP NOTIFY from a server that is NOT on the whitelist (i.e. untrusted server), the phone rejects the message. For information about setting this feature, see Chapter 5, the section, "XML SIP Notify Events" on page 5-310.
25	Click  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. The phones support decoding and playing out DTMF tones sent in SIP INFO requests. The following DTMF tones are supported:
- Support signals 0-9, #, *
- Support durations up to 5 seconds

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	Packetizatio n 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 configure a custom set of codecs and attributes instead of using the defaults (see “Customized Codec Preference List” below).

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. The default payload setting is to allow all codecs. You can set payload to use only basic codecs (G.711 u-Law, G.711 a-Law, G.729), or select from up to 14 codecs for the phones AND customize a codec preference list of up to 10 codecs. In the Aastra Web UI, codecs 2 through 10 can be set to “None” if required (no codecs).

Note:
In the Aastra Web UI:
- setting Codec 1 to “All” ignores the packetization interval (ptime). The packetization interval setting defaults to 30, which is the default for all codecs.
- setting Codec 1 to “All” automatically sets all other codec preference fields 2 through 10 to “None”.
- setting Codec 1 to “Basic” and all other codec preferences in 2 through 10 to “None”, forces the phone to use only the basic codecs as in previous releases (G.711 u-law, G.711 a-law, and G.729). If you select an additional codec to use in the codec preferences 2 through 10 fields, those codecs are added to the list of Basic codecs for the phone to use.

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 Codec 1 Codec 2 . . . . . Codec 10 (in Web UI)	**Configuration Files** 0 - G711u/8000 8 - G711a/8000 98 - G726-16/8000 97 - G726-24/8000 115 - G726-32/8000 96 - G726-40/8000 18 - G729/8000 106 - BV16/8000 107 - BV32/16000 110 - G711u/16000 111 - G711a/16000 9 - G722/8000 113 - L16/16000 112 - L16/8000 Leave blank for all codecs	**Web UI** G711u (8K) G711a (8K) G726-16 G726-24 G726-32 G726-40 G729 BV16 (8K) BV32 (16K) G711u (16K) G711a (16K) G722 L16 (16K) L16 (8K) All (Codec 1 only) Basic (Codec 1 only) None (Codecs 2 thru 10 only)
ptime (in milliseconds) Packetization Interval (in Web UI)	5, 10, 15, 20.......90	
silsupp Silence Suppression (in Web UI)	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-128.

IP Phone UI

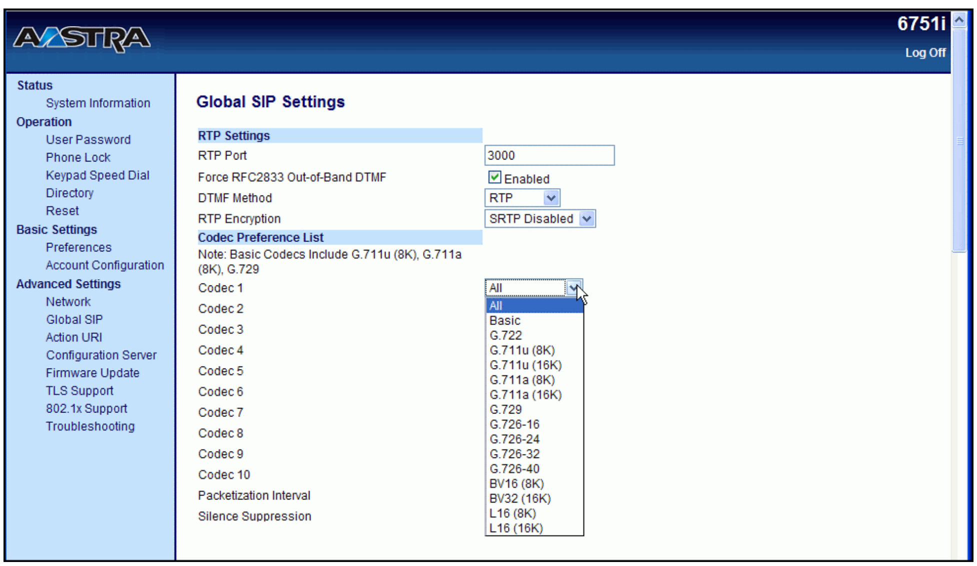
Step	Action
1	Press on the phone to enter the Options List.
2	Select **Administrator Menu.**
3	Enter your Administrator password. **Note:** The IP Phones accept numeric passwords only.
4	Select **SIP Settings.**
5	Select **RTP Port Base** to change the RTP port base setting. Default is **3000**.
6	Press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

IP Phone UI

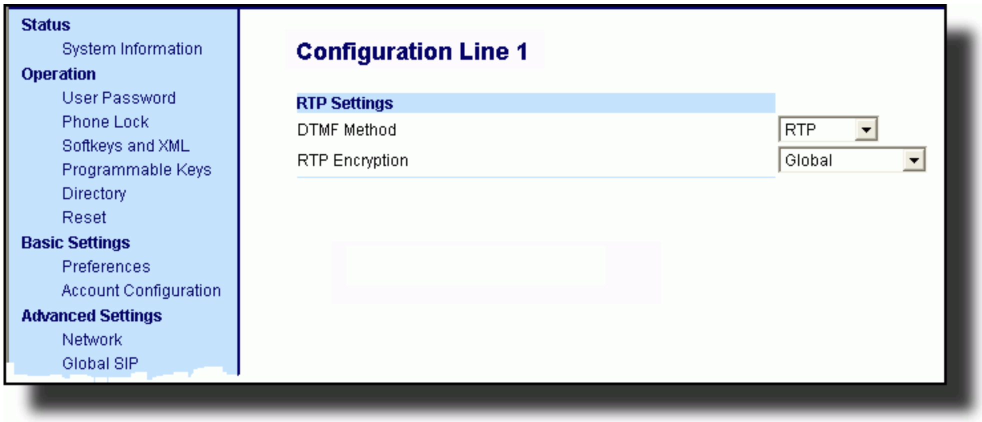
Step	Action
7	Select **Restart.**
For the 6739i:	
1	Press the **Options** key on the phone to enter the Options List.
2	Press the **Advanced** button. A keyboard displays.
3	Enter the Administrator password using the keyboard. Default is “**22222**”.
4	Press the **SIP** button.
5	Press ↓ to scroll to the next screen.
6	Press **RTP Port Base** and enter an RTP port base setting. Default is **3000**.
7	Press ← until the Options List screen displays.
8	Press **Restart** to restart the phone for the change to take affect.

Aastra Web UI

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



Aastra Web UI

Step	Action
	Click on **Advanced Settings->Line <N>->RTP Settings.** ***Per-Line Settings*.** 
2	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. **Note:** The phones support decoding and playing out DTMF tones sent in SIP INFO requests. The following DTMF tones are supported: • Support signals 0-9, #, * • Support durations up to 5 seconds
3	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.
4	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.
5	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.

Aastra Web UI

Step	Action
6	In the **Codec Preference List**, select a codec (with its payload type) you want the phones to use. Valid values are: All Basic (G.711 u-law, G.711 a-law, G.729) G722 G711u/8K G711u/16K G711a/8K G711a/16K G729 G726-16 G726-24 G726-32 G726-40 BV16 (8K) BV32 (16K) L16 (8K) L16 (16K) **Notes**: **1.** Setting **Codec 1** to "**All**" ignores the **packetization interval** (ptime). The **packetization interval** setting defaults to 30, which is the default for all codecs. **2.** Setting **Codec 1** to "**All**" automatically sets all other codec preference fields **2** through **10** to "**None**". **3.** Setting **Codec 1** to "**Basic**" and all other codec preferences in **2** through **10** to "**None**", forces the phone to use only the basic codecs as in previous releases (G.711 u-law, G.711 a-law, and G.729). If you select an additional codec to use in the codec preferences **2** through **10** fields, those codecs are added to the list of Basic codecs for the phone to use.
7	(Optional) In Codec 2 through Codec 10, select a preference of codecs (with its payload type) to use on the phone. Valid values are: None G722 G711u/8K G711u/16K G711a/8K G711a/16K G729 G726-16 G726-24 G726-32 G726-40 BV16 (8K) BV32 (16K) L16 (8K) L16 (16K) **Note:** You can select up to 9 codecs in addition to the codec you selected in step 6.
8	In the "**Packetization Interval**" field, select the time, in milliseconds. Valid values are 5 to 90, in increments of 5 milliseconds.

Aastra Web UI

Step	Action
9	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.
10	Click **Save Settings** to save your changes.

RTCP Summary Reports

The IP phones include the capability of enabling/disabling the generation of RTCP summary reports using the SIP vq-rtcpxr event package. These RTCP summary reports include voice quality statistics according to draft-ietf-sipping-rtcp-summary-05 specifications including packet loss, jitter, and delay statistics, as well as call quality scores. When this feature is enabled, an RTCP summary report is sent at the end of each call via a PUBLISH message to the configuration server.

In addition to enabling/disabling the generation of these reports, you must specify the hostname and port of the entity, known as the collector, receiving the reports. Similar to the other IP Phone SIP account parameters, the RTCP summary report parameters can be set on a global or a per-line basis using the configuration files only.

The RTCP summary report parameters are:

- **sip rtcp summary reports**
- **sip Line*N* rtcp summary reports**
- **sip rtcp summary report collector**
- **sip Line*N* rtcp summary report collector**
- **sip rtcp summary report collector port**
- **sip Line*N* rtcp summary report collector port**

Limitations

The following is a limitation when enabling RTCP summary reports on the phone:

- The call must be at least 5 seconds long in order to generate the RTCP extended reports.

Configuring RTCP Summary Reports

Use the following procedure to configure RTCP summary reports.

Configuration Files

For specific parameters you can set in the configuration files, see Appendix A, the section, "RTCP Summary Reports" on page A-44.

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 speed dial 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:

<u>**Global Configuration**</u>

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

<u>**Per-Line Configuration**</u>

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

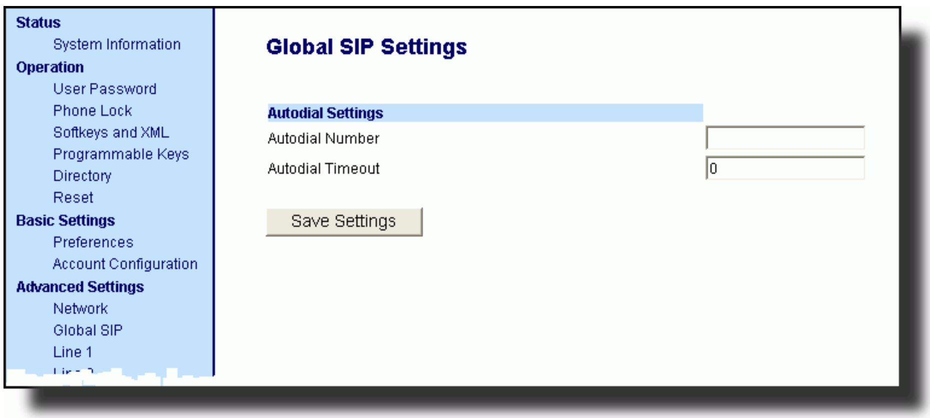
Configuration Files

For specific parameters you can set in the configuration files, see Appendix A, the section, “Autodial Settings” on page A-133.

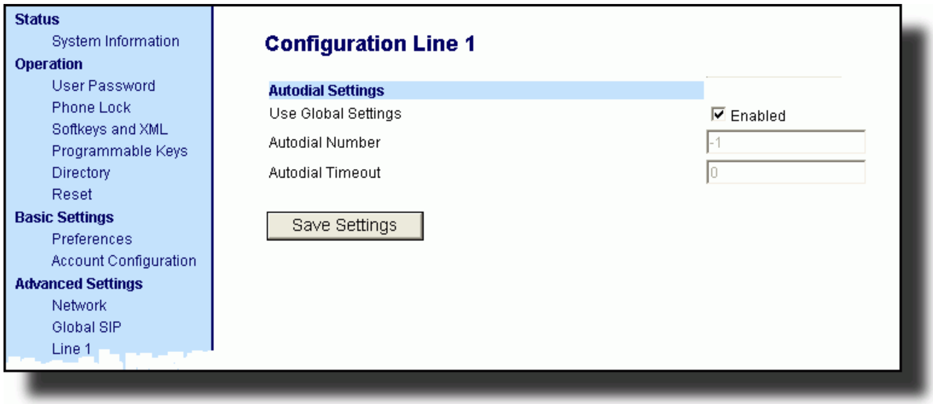
Configuring Autodial Using the Aastra Web UI

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**
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.

<table>
<tr><th colspan="2">Aastra Web UI</th></tr>
<tr><td colspan="2">Per-Line Configuration</td></tr>
<tr><td>1</td><td>Click on Advanced Settings->Line <1 - 9>->Autodial Settings.



</td></tr>
<tr><td>2</td><td>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.</td></tr>
<tr><td>3</td><td>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</td></tr>
<tr><td>4</td><td>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.</td></tr>
<tr><td>5</td><td>Click Save Settings to save your changes.</td></tr>
</table>

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. For each Protocol, you can specify the path for which the configuration files are located on the server. For HTTP and HTTPS, you can also specify the port number to use for downloading the phone configuration. For FTP, you can configure a Username and Password that are authenticated by the FTP server.

The TFTP setting is the default download protocol. You can configure the type of protocol that the IP phones use 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 4-3.

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-18.

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.**

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** (8 and 11-Line LCD phones) or **Set** (3-Line LCD phones) 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 fully qualified domain name of the primary TFTP server. - Press **Done** or **Set** to save the change. - Select **Pri TFTP Path**. - Enter the path name for which the configuration files reside on the TFTP server for downloading to the IP Phone. If the IP phone's files are located in a sub-directory beneath the server's root directory, the relative path to that sub-directory should be entered in this field. Enter the path name in the form *folderX\folderX\folderX*. For example, ***ipphone\6757i\configfiles.*** **Optional**: You can also configure an Alternate TFTP server and Alternate TFTP Path if required by selecting the "Alternate TFTP" and the "Alt TFTP Path" parameters. - From the TFTP Settings menu, select **Alternate TFTP** and press Enter. - Enter the IP address or fully qualified domain name of the alternate TFTP server. - Press **Done** or **Set** to save the change. - Select **Alt TFTP Path**. - Enter the path name for which the configuration files reside on the Alternate TFTP server for downloading to the IP Phone. If the IP phone's files are located in a sub-directory beneath the server's root directory, the relative path to that sub-directory should be entered in this field. Enter the path name in the form *folderX\folderX\folderX*. For example, ***ipphone\6757i\configfiles.*** **FTP Settings** - Select **FTP Server**. - Enter the IP address or fully qualified domain name of the FTP server. - Press **Done** or **Set** to save the change. - Select **FTP Path**. - Enter the path name for which the configuration files reside on the FTP server for downloading to the IP Phone. If the IP phone's files are located in a sub-directory beneath the server's root directory, the relative path to that sub-directory should be entered in this field. Enter the path name in the form *folderX\folderX\folderX*. For example, ***ipphone\6757i\configfiles.*** **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** or **Set**.

IP Phone UI

<table>
<tr><th>Step</th><th>Action</th></tr>
<tr><td>9
(Cont'd)</td><td>
<table>
<tr><th>HTTP Settings</th></tr>
<tr><td>
- Select HTTP Server
- Enter the IP address of the HTTP server.
- Press Done or Set.
- Select HTTP Path.
- Enter the path name for which the configuration files reside on the HTTP server for downloading to the IP Phone. If the IP phone's files are located in a sub-directory beneath the server's root directory, the relative path to that sub-directory should be entered in this field. Enter the path name in the form folderX\folderX\folderX. For example, ipphone\6757i\configfiles.
- Select HTTP Port.
- Enter the HTTP port that the server uses to load the configuration to the phone over HTTP. Default is 80.
- Press Done or Set.
</td></tr>
<tr><th>HTTPS Settings</th></tr>
<tr><td>
- Select HTTP Client.
- Select Download Server.
- Enter the IP address of the HTTPS server.
- Press Done or Set.
- Select Download Path.
Enter the path name for which the configuration files reside on the HTTPS server for downloading to the IP Phone. If the IP phone's files are located in a sub-directory beneath the server's root directory, the relative path to that sub-directory should be entered in this field. Enter the path name in the form folderX\folderX\folderX. For example, ipphone\6757i\configfiles.
- Press Done or Set.
- Select Client Method.
- Select the client method to use for downloading the configuration files (SSl 3.0 or TLS 1.0). For more information about which client method to use, see the section, "HTTPS Client/ Server Configuration" on page 4-41.
- Select Download Port.
- Enter the HTTPS port that the server uses to load the configuration to the phone over HTTPS. Default is 443.
</td></tr>
</table>
</td></tr>
</table>

IP Phone UI

Step	Action
9 (Cont'd)	**HTTPS Settings (Continued)** - Select **HTTPS Server.** - Select **HTTP->HTTPS**. - **For 3-Line LCD Displays:**: Press **Change** to select "**Do not redirect**"or "**Redirect**". Default is "**Do not redirect**". Enabling this feature redirects the HTTP protocol to HTTPS. Press **Set**. - For 8 and 11-Line LCD Displays:: Press **Change** to select "**Yes**" and redirect HTTP to HTTPS. Select "**No**" to not direct HTTPS to HTTPS, Default is "**No**". Enabling this feature redirects the HTTP protocol to HTTPS. Press **Done**. - Select **XML HTTP POSTs**. - For 3-Line LCD Displays:: Press **Change** to select "**Do not block**" or "**Block**". Enabling this feature blocks XML HTTP POSTs from the IP Phone. - For 8 and 11-Line LCD Displays:: Press **Change** to select "**Yes**" and block XML HTTP Posts. Select "**No**" to unblock XML HTTP Posts. 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 4-41.
10	Press **Done** or **Set** repeatedly until the session prompts you to restart the IP phone to apply the configuration settings.
11	Select **Restart.**

For the 6739i:

IP Phone UI

Step	Action
1	Press the **Options** key on the phone to enter the Options List.
2	Press the **Advanced** button. A keyboard displays.
3	Enter the Administrator password using the keyboard. Default is “**22222**”.
4	Press the **SIP** button.
5	Press **Cfg. Svr.**.
6	Pres the **Download Protocol** field, and press the value you want the phone to use for downloading from the configuration server. Valid values are: • **TFTP** (Default) • **FTP** • **HTTP** • **HTTPS** The IP phone uses the protocol you select to download new firmware and configuration files from the configuration server.
7	After selecting the download protocol, you must identify specific parameters for that protocol. On the Cfg. Svr screen, press the button associated with the download protocol you selected in step 6. TFTP FTP HTTP HTTPS

IP Phone UI

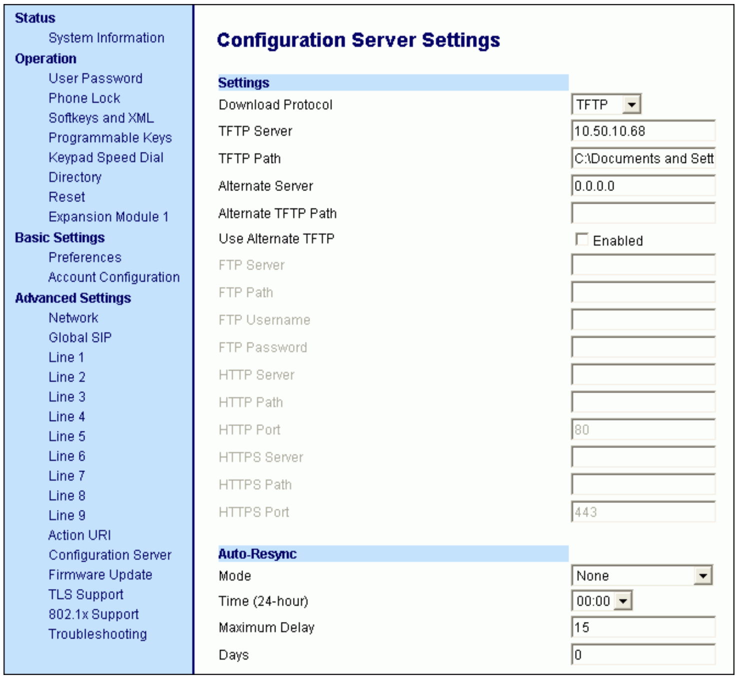
Step	Action
8	Use the following table to configure the applicable server.

TFTP Settings
- Press **TFTP Server.** - Enter the IP address or fully qualified domain name of the primary TFTP server. - Press **TFTP Path**. - Enter the path name for which the configuration files reside on the TFTP server for downloading to the IP Phone. If the IP phone's files are located in a sub-directory beneath the server's root directory, the relative path to that sub-directory should be entered in this field. Enter the path name in the form *folderX\folderX\folderX*. For example, ***ipphone\6757i\configfiles.*** **Optional**: You can also configure an Alternate TFTP server and Alternate TFTP Path if required by selecting the "Alternate TFTP" and the "Alt TFTP Path" parameters. - Press **Use Alternate TFTP,** and press "**Yes**" to use an alternate TFTP server. Default is "**No**". - Press **Alternate TFTP Server.** - Enter the IP address or fully qualified domain name of the alternate TFTP server. - Press **Alt. TFTP Path**. - Enter the path name for which the configuration files reside on the Alternate TFTP server for downloading to the IP Phone. If the IP phone's files are located in a sub-directory beneath the server's root directory, the relative path to that sub-directory should be entered in this field. Enter the path name in the form *folderX\folderX\folderX*. For example, ***ipphone\6757i\configfiles.***
FTP Settings
- Press **FTP Server**. - Enter the IP address or fully qualified domain name of the FTP server. - Press **FTP Path**. - Enter the path name for which the configuration files reside on the FTP server for downloading to the IP Phone. If the IP phone's files are located in a sub-directory beneath the server's root directory, the relative path to that sub-directory should be entered in this field. Enter the path name in the form *folderX\folderX\folderX*. For example, ***ipphone\6757i\configfiles.*** **Optional**: You can enter a username and password for accessing the FTP server if required: - Press **FTP Username**. - Enter a username for accessing the FTP server. - Press **FTP Password**. - Enter a password for accessing the FTP server.

IP Phone UI

Step	Action
9 (Cont'd)	**HTTP Settings** - Press **HTTP Server.** - Enter the IP address of the HTTP server. - Press **HTTP Port**. - Enter the HTTP port that the server uses to load the configuration to the phone over HTTP. Default is **80**. .-Press **HTTP Path**. - Enter the path name for which the configuration files reside on the HTTP server for downloading to the IP Phone. If the IP phone's files are located in a sub-directory beneath the server's root directory, the relative path to that sub-directory should be entered in this field. Enter the path name in the form *folderX\folderX\folderX*. For example, ***ipphone\6757i\configfiles.*** **HTTPS Settings** - Press **HTTPS Server.** - Enter the IP address of the HTTPS server. - Press **HTTPS Port**. - Enter the HTTPS port that the server uses to load the configuration to the phone over HTTPS. Default is **443**. - Press **HTTPS Path.** Enter the path name for which the configuration files reside on the HTTPS server for downloading to the IP Phone. If the IP phone's files are located in a sub-directory beneath the server's root directory, the relative path to that sub-directory should be entered in this field. Enter the path name in the form *folderX\folderX\folderX*. For example, ***ipphone\6757i\configfiles.*** - Press **HTTPS Client Method**. - Select the client method to use for downloading the configuration files (SSl 3.0 or TLS 1.0). For more information about which client method to use, see the section, "HTTPS Client/ Server Configuration" on page 4-41.
9	Press [←] until the Options List screen displays.
10	Press **Restart** to restart the phone for the change to take affect.

Aastra Web UI

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

Aastra Web UI

<table>
<tr><th>Step</th><th>Action</th></tr>
<tr><td>2</td><td>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.

<table>
<tr><th>TFTP</th></tr>
<tr><td>- Enter an IP address or fully qualified domain name in the "TFTP Server" field.
- Enter the path name in the “TFTP Path” field for which the configuration files reside on the TFTP server for downloading to the IP Phone. If the IP phone’s files are located in a sub-directory beneath the server’s root directory, the relative path to that sub-directory should be entered in this field. Enter the path name in the form folderX\folderX\folderX. For example, ipphone\6757i\configfiles.

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. You can also enter a path name for the alternate TFTP server in the “Alternate TFTP Path” field.</td></tr>
<tr><th>FTP</th></tr>
<tr><td>- Enter an IP address or fully qualified domain name in the "FTP Server" fiel.d.
- Enter the path name in the “FTP Path” field for which the configuration files reside on the FTP server for downloading to the IP Phone. If the IP phone’s files are located in a sub-directory beneath the server’s root directory, the relative path to that sub-directory should be entered in this field. Enter the path name in the form folderX\folderX\folderX. For example, ipphone\6757i\configfiles.

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

Aastra Web UI

<table>
<tr><th>Step</th><th>Action</th></tr>
<tr><td>3</td><td>
<table>
<tr><th>HTTP</th></tr>
<tr><td>- Enter an IP address or fully qualified domain name in the "HTTP Server" field.
- Enter the path name in the “HTTP Path” field for which the configuration files reside on the HTTP server for downloading to the IP Phone. If the IP phone’s files are located in a sub-directory beneath the server’s root directory, the relative path to that sub-directory should be entered in this field. Enter the path name in the form folderX\folderX\folderX. For example, ipphone\6757i\configfiles.
- Enter the HTTP port number in the “HTTP Port” field that the server uses to load the configuration to the phone over HTTP.

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.</td></tr>
<tr><th>HTTPS</th></tr>
<tr><td>- Enter an IP address or fully qualified domain name in the "HTTPS Server" field.
- Enter the path name in the “HTTPS Path” field for which the configuration files reside on the HTTP server for downloading to the IP Phone. If the IP phone’s files are located in a sub-directory beneath the server’s root directory, the relative path to that sub-directory should be entered in this field. Enter the path name in the form folderX\folderX\folderX. For example, ipphone\6757i\configfiles.
- Enter the HTTPS port number in the “HTTPS Port” field that the server uses to load the configuration to the phone over HTTPS.

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

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 4-41.</td></tr>
<tr><td>4</td><td>Click Save Settings to save your settings.

Note: The session prompts you to restart the IP phone to apply the configuration settings.</td></tr>
<tr><td>5</td><td>Select Operation->Reset and click Restart.</td></tr>
</table>

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 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.

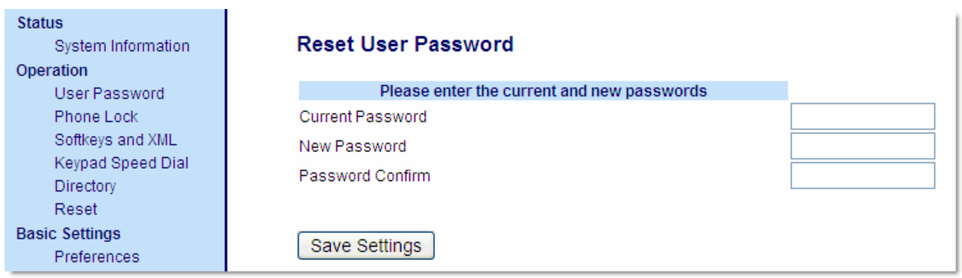
Note: The IP phones support numeric characters only in passwords. If you enter a password with alpha characters, the phone uses the default password instead.

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-14.

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.
For the 6739i:	
1	Press the **Options** key on the phone to enter the Options List.
2	Press **Password**. A keyboard displays.

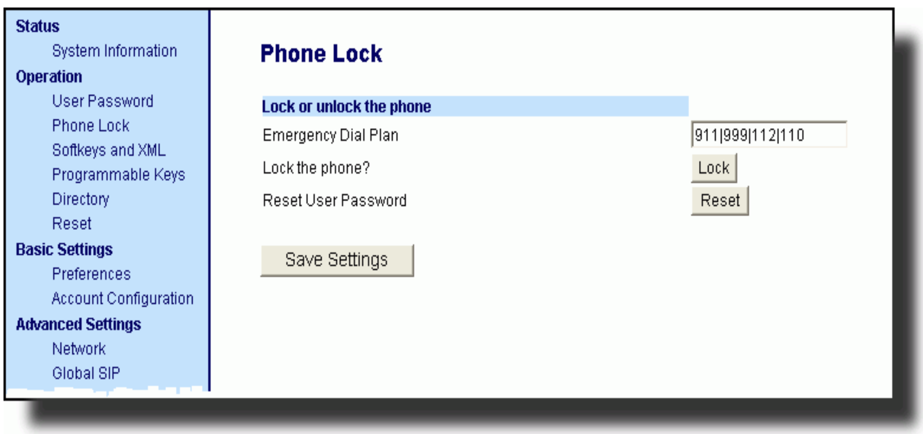
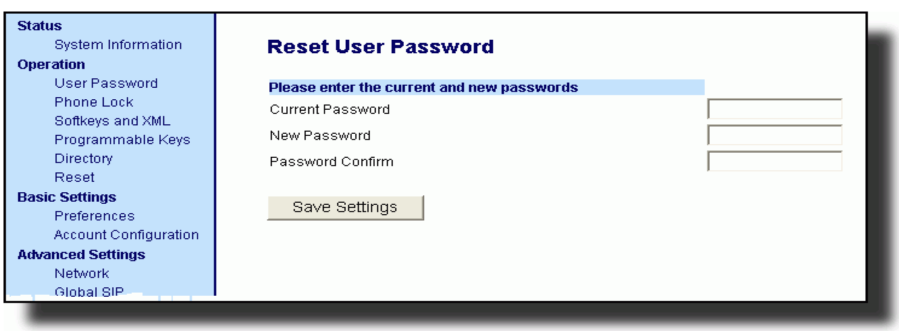
IP Phone UI	
3	Press on the “**Current Password**” text box, and enter the current user password.
4	Press on the “**New Password**” text box, and enter the new user password.
5	Press on the “**Re-enter Password**” text box, and re-enter the user password.
6	Press **Save**. The Options screen displays indicating your changes were saved.

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  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**. 
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.
For the 6739i:	
1	Press the **Options** key on the phone to enter the Options List.

Aastra Web UI	
2	Press **Password**. A keyboard displays.
3	Press on the "**Current Password**" text box, and enter the current user password.
4	Press on the "**New Password**" text box, and enter the new user password.
5	Press on the "**Re-enter Password**" text box, and re-enter the user password.
6	Press **Save**. The Options screen displays indicating your changes were saved.

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-14.

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 at the path *Options->Lock*.
- Using the Aastra Web UI via the path *Operation->Phone Lock*.
- Using the configuration files to configure a softkey as “phonelock”, and then pressing the key to lock/unlock the phone.
- Using the Aastra Web UI to configure a softkey 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 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**. Default is “**22222**”. The phone unlocks.

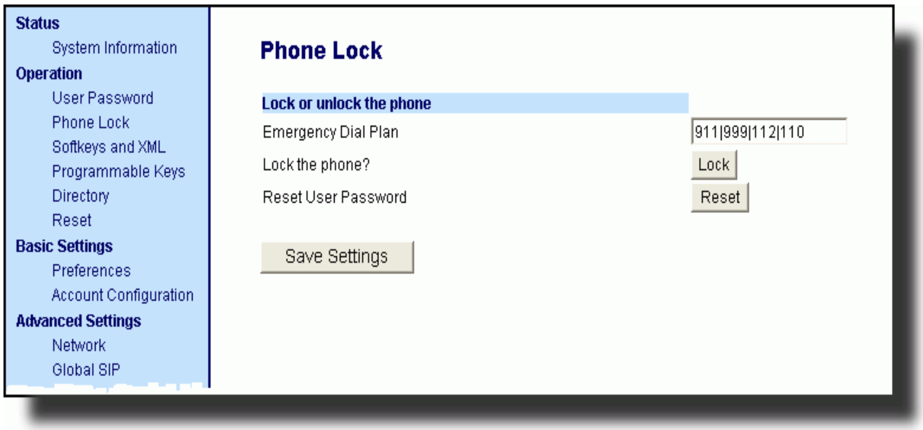
***For the 6739i*:**

IP Phone UI

Step	Action
Lock the phone:	
1	Press on the phone to enter the Options List.
2	Press **Lock.** The prompt, "*Lock the phone*?" displays.
3	Press **Yes** to lock the phone The phone locks.
Unlock the phone:	
1	Press on the phone to enter the Options List. A "*Phone is Locked*" screen displays allowing you to press an "**Unlock the Phone**" button.
2	Press **Unlock the Phone**. A prompt, "*Enter Unlock Password"* displays as well as a keyboard.
3	Enter the user or administrator password and press **Enter**. Default is "**22222**". A prompt "*Unlock the Phone?*" displays.
4	Press **Yes** to unlock the phone. 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-202.

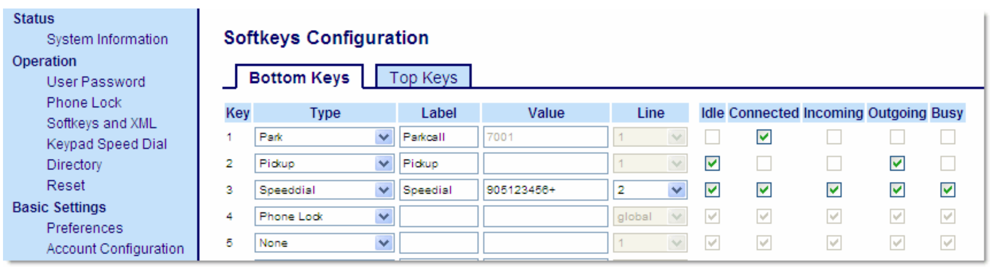
Reference

To use the lock/unlock softkey or programmable key, see "Using the Configured Lock/Unlock Softkey on the IP Phone" on page 5-12.

Configuring a Lock/Unlock Softkey using the Aastra Web UI

Using the Aastra Web UI, you can configure a softkey 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 softky 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 6757i Key Screen is shown as an example. 
2	Select a key you want to configure for locking/unlocking the phone.
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 Softkey 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.

IP Phone UI	
Step	**Action**
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.

For the 6739i:

IP Phone UI	
Step	**Action**
Lock the phone:	
1	Press the **LOCK** softkey. The phone locks. The message "*Phone is Locked*" displays on the screen. The LED for the softkey AND the Message Waiting Lamp illuminate steady ON.
Unlock the phone:	
1	Press the **UNLOCK** key. The "Unlock" key has a steady ON LED. A password prompt displays..
2	Enter your user password and press **ENTER**. The phone unlocks. The LED for the key AND the Message Waiting Lamp go OFF. The message "*Phone is unlocked*" briefly displays on the screen..
3	Enter the user or administrator password and press **Enter**. Default is "**22222**". A prompt "*Unlock the Phone?*" displays.
4	Press **Yes** to unlock the phone. The phone unlocks.

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.

Emergency Dial Plan and Pattern Matching

The IP Phones support emergency dialing using pattern matching and prepend dial plan functionality.

There are two ways to dial a number on the phone:

- dialing digit-by-digit (i.e., select line and dial)
- dialing by string (i.e., pre-dial then go off-hook)

When a user dials digit-by-digit, the phone adds every digit to a dialed string and checks against the dial plan. If the phone is not locked, it checks against the regular dial plan. If the phone is locked, it checks against the emergency dial plan.

When a user dials by string, (pre-dial, speed-dial, etc., and then goes off-hook), and the phone is not locked, it checks to see if the number matches the emergency dial plan. If it doesn't match, it blocks the call from going through. If the phone is locked, and the number matches the emergency dial plan it allows the call to go through.

Adding a pre-pend to a dial plan also works with both dialing digit-by-digit and dialing by string.

Limitation

The following is a limitation for emergency dial plans with pattern matching:

- Secondary dial tone is not supported.

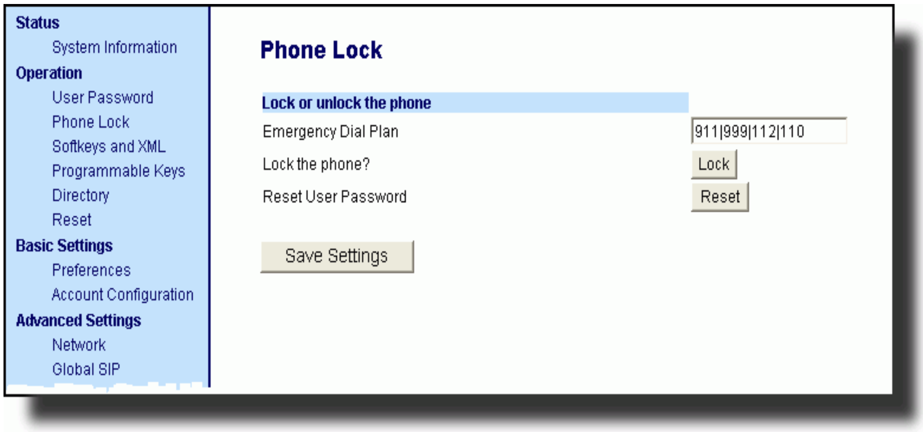
You can set the emergency dial plan via the configuration files or the Aastra Web UI.

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-15.

Aastra Web UI	
1	Click on **Operation->Phone Lock**. 
2	In the "**Emergency Dial Plan**" field, enter the 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 "x+#|xx+*". You can enter up to 512 characters in this field.
3	Click Save Settings to save the emergency dial plan to your phone.

User Dial Plan Setting

The IP Phones have a parameter for configuring a dial plan that distinguishes between calling a PSTN number and an Internet URL.

This parameter is "**sip user parameter dial plan**". Using the configuration files, an Administrator can configure a dial plan (i.e., **6xx|8xxxx|9xxxxxxx**) that the phone checks before sending the SIP packet. If for example, the number that was dialed was 645, the phone checks the dial plan and matches the number to the dial plan (**6xx** in the example above), before sending out the SIP packet. The SIP packet header indicates user=ip (i.e., "To: <sip:645@10.30.102.24:10060;user=ip>"). If the number that was dialed was 9,456-2345, the phone matches the number to the dial plan (**9xxxxxxx** in the example above), before sending out the SIP packet. The SIP packet header in this case indicates user=phone (i.e., "To: <sip:94562345:10060;user=phone>").

Note: Entering a dial plan value for this parameter enables this feature. Entering no value for this parameter in the configuration files, disables this feature.

An Administrator can use the following parameter in the configuration files for this feature:

- **sip user parameter dial plan**

When you enter a value for this parameter in the configuration files, the phone performs in the following manner:

WHEN	THEN
the SIP phone calls a PSTN phone number,	the network uses the "sip user parameter dial plan" value to connect the call to the far-end destination on the PSTN (User = phone). For example, the network would use the parameter if the following dial plan was configured: **sip user parameter dial plan: 6xx | 8xxx** This indicates the destination phone is on the PSTN network and so the parameter is used.
the SIP phone calls a SIP phone,	the network disregards the "sip user parameter dial plan" value and connects the SIP phone to the far-end SIP destination (User = IP). For example, the network would not use the parameter if the following dial plan was configured: **sip user parameter dial plan: 8xxx** This indicates the destination phone is on an IP network and so the parameter is not used.

Note: You can configure the “sip user parameter dial plan” parameter on a global basis only. If it is misconfigured, then the parameter is ignored.

Configuring the SIP User Parameter Dial Plan

Use the following procedure to configure the SIP user parameter dial plan.

Configuration Files

For the specific parameter you can set in the configuration files, see Appendix A, the section, “User Dial Plan Setting” on page A-16.

Limitation

The “sip user parameter dial plan” value is checked AFTER an existing pre-pend dial plan is checked, so the number that the phone dials will have pre-pend digits also.

Time and Date

In addition to enabling/disabling the time server, you can also 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 Format	Configuration Files IP Phone UI Aastra Web 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.

Note: DST can be set on the phones using the IP Phone UI and configuration files only. For more information, see "Time Zone & DST" on page 5-20.

Time Zone & DST

There are two ways you can set the time zone on the IP Phones.

First Method - You can set a time zone using the **Time Zone** option in the IP Phone UI or you can use the "**time zone name"** parameter in the configuration files. Both methods allow you to enter a value from the Time Zone table. The list of time zone names is provided in the table in Appendix A, the section, "Time Zone Name/Time Zone Code Table" on page A-51. The following is an example:

```
time zone name: US-Eastern
```

Second Method - You can use the **Time Zone** option in the IP Phone UI or your can use the "**time zone name"** parameter in the configuration files, and specify a value of "**Custom**" for this parameter (must use initial caps). The "Custom" option allows you to customize the time zone for your area using additional configuration parameters. The following is an example using relative time for EST:

```
time zone name: Custom
```

The following table identifies the additional time zone and DST parameters you can enter in the configuration files..

Custom Configuration File Parameter	Description	Example
time zone minutes	The number of minutes the time zone is offset from UTC (Coordinated Universal Time). This can be positive (West of the Prime Meridian) or negative (East of the Prime Meridian). The default is Eastern Standard Time (EST) with a value of 300 (GMT minus 5 hours).	time zone minutes: 300 **Note:** For additional values for this parameter, see "Custom Time Zone and DST Settings" on page A-57.
dst minutes	The number of minutes to add during Daylight Saving Time. Valid values are a positive integer between 0 to 60.	dst minutes: 60
dst [start|end] relative date	Specifies how to interpret the start and end day, month, and week parameters - absolute (0) or relative (1).	dst [start|end] relative date: 1
Absolute Time (not applicable to Eastern Standard Time (EST))		
dst start month	The month that DST starts. Valid values are 1 to 12 (January to December).	dst start month: 3
dst end month	The month that DST ends. Valid values are 1 to 12 (January to December).	dst end month: 4
dst [start |end] week	Not applicable to absolute time.	N/A
dst start day	The day of the month that DST starts. Valid values are 1 to 31.	dst start day: 15

Custom Configuration File Parameter	Description	Example
dst end day	The day of the month that DST ends. Valid values are 1 to 31.	dst end day: 31
dst start hour	The hour that DST starts. Valid values are an integer from 0 (midnight) to 23.	dst start hour: 5
dst end hour	The hour that DST ends. Valid values are an integer from 0 (midnight) to 23.	dst end hour: 23
Relative Time		
dst start month	The month that DST starts. Valid values are 1 to 12 (January to December).	dst start month: 4
dst end month	The month that DST ends. Valid values are 1 to 12 (January to December).	dst end month: 5
dst start week	The week in the specified month in which DST starts. Valid value is a positive or negative integer from 1 to 5. 1 = first full week of month -1 = last occurance "dst start day" in the month 2 = second full week of month -2 =second last occurance "dst start day" in the month . . . 5 = fifth full week of month -5 = fifth last occurance "dst start day" in the month	dst start week: 2
dst end week	The week in the specified month in which DST ends. Valid value is a positive or negative integer from 1 to 5. 1 = first full week of month -1 = last occurance "dst start day" in the month 2 = second full week of month -2 =second last occurance "dst start day" in the month . . . 5 = fifth full week of month -5 = fifth last occurance "dst start day" in the month	dst end week: -1
dst start day	The day of the specified week in the specified month that DST starts on. Valid values are an integer from 1 to 7. 1 = Sunday 2 = Monday . . . 7 = Saturday	dst start day: 2

Custom Configuration File Parameter	Description	Example
dst end day	The day of the specified week in the specified month that DST ends on. Valid values are an integer from 1 to 7. 1 = Sunday 2 = Monday . . . 7 = Saturday	dst end day: 7
dst start hour	The hour that DST starts. Valid values are an integer from 0 (midnight) to 23.	dst start hour: 10
dst end hour	The hour that DST ends. Valid values are an integer from 0 (midnight) to 23.	dst end hour: 23

Example 1

The following is an example of a custom time zone configuration in the configuration files using relative time (for EST):

```
time zone name: Custom
dst [start|end] relative date: 1 #relative
time zone minutes: 300
dst minutes: 60
```

Example 2

The following is an example of a custom time zone configuration in the configuration files using absolute time:

```
time zone name: Custom
dst [start|end] relative date: 0 #absolute


#start of DST
dst start month: 3 #March
dst start week: 2 #second full week
dst start day: 1 #Sunday

#End of DST
dst end month: 11 #November
dst end week: 1 #first full week
dst end day: 1 #Sunday
```

DHCP Time Offset (Option 2) Support

DHCP Option 42 enables the phone to be configured with the Network Time Protocol (NTP) server addresses. However, NTP provides the Coordinated Universal Time (UTC) time so the phone requires the offset from UTC in order to deliver the correct local time.

A User or Administrator can set the offset of UTC using DHCP Option 2.

An Administrator can enable Option 2 in the configuration files by setting the parameter "**time zone name**". If this parameter contains the **DP-Dhcp** value, the phone derives the time and date from UTC and the time offset offered by the DHCP server.

Using the IP Phone UI, a User or Administrator can enable the phone to use DHCP Option 2 by setting the following values from the Country Code list on the phone:

Country Name	Country Code
Dhcp	DP

Note: The country name, country code, and time zone name are case sensitive.

On the IP Phone UI for 5-line phones, a User or Administrator can select ***Preferences->Time and Date->Time Zone->Others*** and choose "**DP-Dhcp**" from the displayed time zone list.

On the IP Phone UI for 3-line phones, you select ***Preferences->Time and Date->Time Zone->Others*** and enter "**DP**" for the country code, or press "*****" and select "**Dhcp**"from the displayed time zone list.

On the 6739i, a User or Administrator can select ***Set Time->Timezone->Others*** and choose "**DP-Dhcp**" from the displayed time zone list.

If you enable DHCP Option 2 via the IP Phone UI, the change takes place dynamically.

Notes:

1. When DHCP Option 2 is enabled on the phone, the phone still uses the values configured for ***Daylight Savings*** to control daylight savings time.

2. The default behavior for the phone is to use the NTP server from Option 42 (or current configuration setting) and the current time zone settings.

3. If the time zone name parameter is set to a value other than Dhcp, then DHCP Option 2 is disabled.

References

For more information about setting DP-DHCP for the timezone, see Appendix A, "Time Zone Name" on page A-50.

For more information about setting the country code, see Appendix A, "Country Codes (from Standard ISO 3166)" on page A-179.

Custom Time Zone Support

A User or Administrator can also set a custom time zone on the phone.to be configured with the Network Time Protocol (NTP) server addresses. However, NTP provides the Coordinated Universal Time (UTC) time so the phone requires the offset from UTC in order to deliver the correct local time.

On the IP Phone UI for 5-line phones, a User or Administrator can select ***Preferences->Time and Date->Time Zone->Others*** and choose "**Custom**" from the displayed time zone list.

On the IP Phone UI for 3-line phones, you select ***Preferences->Time and Date->Time Zone->Others*** and enter "**Custom**" for the country code.

On the 6739i, a User or Administrator can select ***Set Time->Timezone->Others*** and choose "**Custom**" from the displayed time zone list.

References

For more information about setting a custom timezone, see Appendix A, "Time Zone Name" on page A-50.

Configuring Time and Date Using the Configuration Files

Use the following information to set a time and date format, time zone, and daylight savings time using the configuration files..

Configuration Files
For specific date and time parameters you can set in the configuration files, see Appendix A, the section, "Time and Date Settings" on page A-48. For specific parameters you can set for custom time zone settings, see Appendix A, the section, "Custom Time Zone and DST Settings" on page A-57.

Configuring Time and Date Using the IP Phone UI

Use the following procedure to set a time and date, time and date format, time zone, and daylight savings time using the IP Phone UI.

IP Phone UI

Step	Action
1	Press [Options key] on the phone to enter the Options List.
Set Time Format:	
2	Select **Time and Date.**
3	Select **Time Format.** Valid values are **12hr** and **24hr**. **Note:** The default Time Format is **12hr**.
4	Press **Change** to toggle between 24hr and 12hr format.

IP Phone UI

Step	Action
5	Press **Done** to save the Time Format you selected.
Set Date Format:	
6	Select **Date Format.**
7	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 • DD MMM YYYY • WWW DD MMM • DD MMM • DD.MM.YYYY **Note:** The default Date Format is **WWW MMM DD** (Day of Week, Month, Day).
8	Press **Done** to save the Date Format.
Set Time Zone:	
9	Select **Time Zone**.
10	***For 3-Line LCD Displays:*** Press * to display a list of countries and select a country. ***For 5-Line LCD Displays:*** Select from the following list of countries: • America • Asia • Atlantic. • Australia • Europe • Pacific • Others (DP-Dhcp, Custom) **Note:** For more information about setting the time zone to “DP-Dhcp” or “Custom”, see “DHCP Time Offset (Option 2) Support” on page 5-23, “Custom Time Zone Support” on page 5-24, or Appendix A, “Time Zone Name” on page A-50.
11	Select a Time Zone from the list of values. For valid values, see Appendix A, the section, “Time and Date Settings” on page A-48. **Note:** The default Time Zone is **US-Eastern**.
12	Press **Done** to save the Time Zone you selected.
Set Daylight Savings Time:	
13	Select **Daylight Savings**.

IP Phone UI	
Step	**Action**
14	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**.
15	Press **Done** to save the Daylight Savings value you selected.

***For the 6739i:*.**

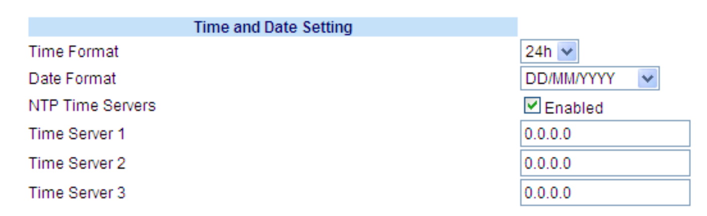
IP Phone UI	
Step	**Action**
1	Press the **Options** key on the phone to enter the Options List.
Set Time Format:	
2	Press **Set Time.**
3	Press **Time Format** and select a value for the time format on your phone. Valid values are **12hr** and **24hr**. **Note:** The default Time Format is **12hr**.
Set Date Format:	
4	Press **Date Format.**
5	Select a date format from the list of values. 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 • DD MMM YYYY • WWW DD MMM • DD MMM • DD.MM.YYYY **Note:** The default Date Format is **WWW MMM DD** (Day of Week, Month, Day).
Set Time Zone:	
6	Press **Time Zone**.

IP Phone UI

Step	Action
7	Press a country. Valid values are: • America • Asia • Atlantic. • Australia • Europe • Pacific • Others (DP-Dhcp, Custom) **Note:** For more information about setting the time zone to “DP-Dhcp” or “Custom”, see “DHCP Time Offset (Option 2) Support” on page 5-23, “Custom Time Zone Support” on page 5-24, or Appendix A, “Time Zone Name” on page A-50.
8	Press a Time Zone from the list of values. For valid values, see Appendix A, the section, “Time and Date Settings” on page A-48. The default Time Zone is **US-Eastern**.
9	Press the to return to the previous screen.
Set Daylight Savings Time:	
10	Press **Daylight Savings**.
11	Press a Daylight Savings time option. Valid values are: • OFF • 30 min summertime • 1 hr summertime • automatic **Note:** The default for Daylight Savings is **Automatic**.
12	Press the to return to the previous screen.
13	Press the button or the button at any time to return to the idle screen.

Configuring Time and Date Using the Aastra Web UI

Use the following procedure to set a time and date, time and date format, time zone, and daylight savings time using the Aastra Web UI.

	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 • DD MMM YYYY • WWW DD MMM • DD MMM • DD.MM.YYYY **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.

Time Servers

A time server is a computer server that reads the actual time from a reference clock and distributes this information to the clients in a network. The time server may be a local network time server or an internet time server. The Network Time Protocol (NTP) is the most widely used protocol that distributes and synchronizes time in the network with the time on the time server.

On the IP phones, you can enable or disable a Time Server to be used to synchronize time on the phones with the Timer Server you specify. An Administrator can use the IP Phone UI, Aastra Web UI, or configuration files to enable/disable the Time Server and specify a Time Server 1, Timer Server 2, and/or Time Server 3. A User can enable/disable the Time Server using the IP Phone UI or Aastra Web UI only. The Time Server is enabled by default.

Setting Time Server Using the Configuration Files

Use the following procedure to enable/disable the Time Server and optionally set the IP Address of Time Servers 1, 2, and/or 3.

Configuration Files
For specific parameters you can set in the configuration files, see Appendix A, the section, "Time Server Settings" on page A-55.

Setting Time Server Using the IP Phone UI

Use the following procedure to set the Time Server and optionally set the IP Address of Time Servers 1, 2, and/or 3.

IP Phone UI

Step	Action
Enable/the Time Server by specifying Time Server 1, 2, and/or 3:	
1	Press on the phone to enter the Options List.
2	Select **Preferences->Time and Date->Time Server**.
3	Select **Timer Server 1**, **Time Server 2**, and/or **Time Server 3**.
4	Enter the IP address of the Time Server, in dotted decimal format. Use the available softkeys to help you enter the information.
5	Click **Done** to save your changes.
For the 6739i:	
1	Press on the phone to enter the Options List.
2	Press **Set Time.**
3	Press **Timer Server 1**, **Time Server 2**, and/or **Time Server 3**. A text box displays.
4	Press the text box and enter the IP address of the Time Server, in dotted decimal format using the keyboard that displays.
5	Press **Enter** to save your changes.

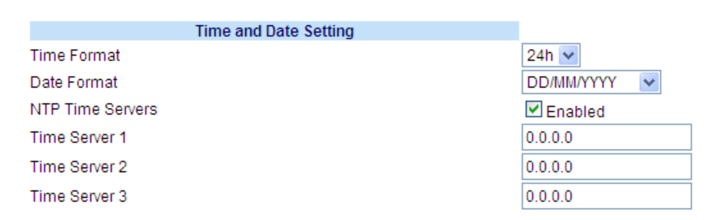
IP Phone UI

Step	Action
6	Press the to return to the previous screen.
7	Press the button or the button at any time to return to the idle screen.

Setting Time Server Using the Aastra Web UI

Use the following procedure to set the Time Server and optionally set the IP Address of Time Servers 1, 2, and/or 3.

Aastra Web UI

1	Click on **Basic Settings->Preferences->Time and Date Setting**. 
To enable/disable Time Server:	
2	The “**NTP Time Server**” field is enabled by default. If you need to disable the Time Server, uncheck the box. The Time Server 1, 2, and 3 fields are grayed out when disabled.
To set Time Server 1, 2, and/or 3:	
3	**Note:** The “**Time Server**” field must be enabled to enter values in the “**Time Server 1, 2, and 3**” fields. In the “**Time Server 1**”, “**Time Server 2**”, and/or “**Time Server 3**” field(s), enter the IP address of the Time Server 1, 2, and/or 3 in your network, in dotted decimal format. Default for this field is **0.0.0.0**. For example, 132.234.5.4
4	Click Save Settings to save your changes.

Backlight Mode (6755i, 6757i, and 6757i CT only)

The 6755i, 6757i, and 6757i CT have a backlight feature that allows you to turn the backlight on the LCD:

- **Off -** Backlight is always OFF.
- **Auto** (Default)- Automatically turns ON the backlight when the phone is in use, and then automatically turns OFF the backlight when the phone is idle after a specified length of time.

"The Auto" setting sets the phone to turn off the backlighting after a period of inactivity. The period of time that the phone waits before turning the backlight off is also configurable.

You can set this backlight feature using the configuration files and the IP Phone UI.

Configuring the Backlight Mode Using the Configuration Files

Use the following information to set the backlight mode and backlight timer on the IP Phones.

Configuration Files

For specific parameters you can set in the configuration files, see Appendix A, the section, "Backlight Mode Settings (6755i, 6757i, 6757i CT)" on page A-67.

Note: Using the configuration files, you can set the backlight to Off (always off) or Auto (On and then off after a period of inactivity).

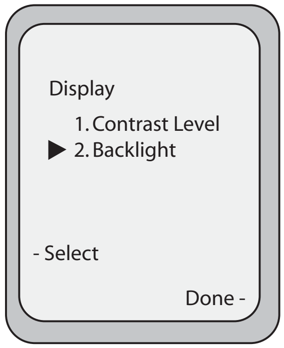
Configuring the Backlight Mode Using the IP Phone UI

Use the following procedure to set the backlight mode and backlight timer on the IP Phone using the IP Phone UI.

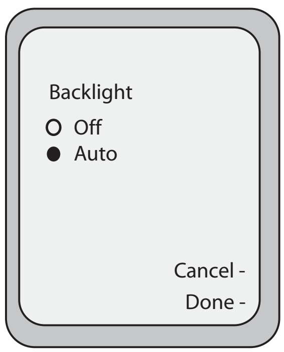
IP Phone UI

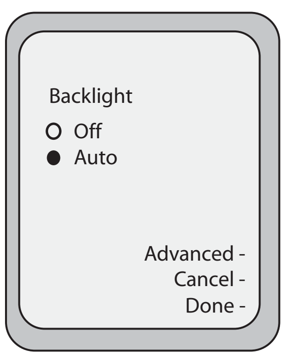
Step	Action
1	Press on the phone to enter the Options List.
2	Select **Preferences.**
3	Select **Display**.

IP Phone UI

<table>
<tr><th>Step</th><th>Action</th></tr>
<tr><td>4</td><td>Select Backlight.


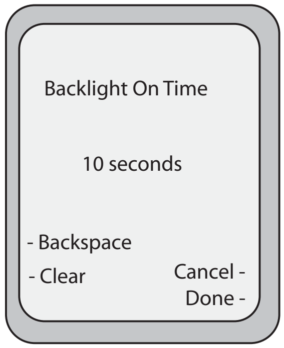
</td></tr>
<tr><td>5</td><td>Use the UP and DOWN arrow keys to select the Backlight status for your phone.
Default is “Auto”. Valid options are:
• Off
• Auto (Default)

Note: Setting the Backlight to “Auto” displays an ADVANCED button for you to set the Auto timer.


</td></tr>
<tr><td>6</td><td>If you select “Off”, press Done to save your setting.</td></tr>
<tr><td>7</td><td>If you select “Auto”, press the Advanced softkey to set the automatic timer.


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

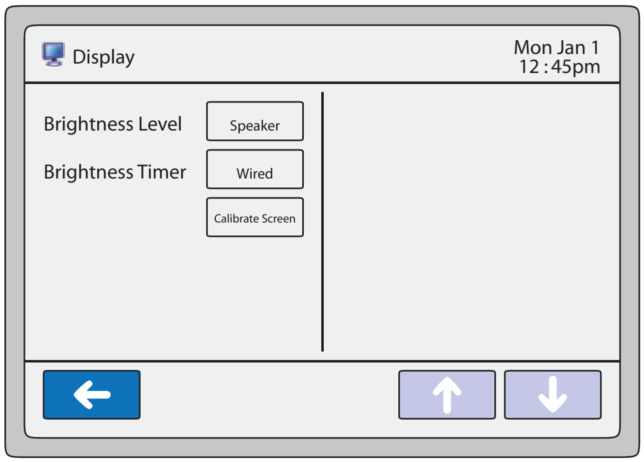
IP Phone UI

Step	Action
8	Using the keypad, enter the amount of seconds you want the phone to stay backlit when the phone is idle. Valid values are **1** to **7200** seconds. Default is **600 seconds** (equals 10 minutes). When this period of time is reached, the phone turns OFF the backlight. Use the "**Backspace**" and/or "**Clear**" softkeys to delete entries if required. 
9	Press **Done** to save your setting. The setting applies immediately to the phone.

Display (6739i only)

The **Display** option on the 6739i allows you to set the following on your phone:

- Brightness Level (6739i only)
- Brightness Timer (6739i only)
- Calibrate Screen (6739i only)



Display Screen

Brightness Level (6739i only)

The "**Brightness Level**" option allows you to set the amount of light that illuminates the touchscreen’s LCD display. Use this option to set your preference of brightness.

Setting Brightness (6739i only).

IP Phone UI

Step	Action
1	Press Options on the phone to enter the Options List
2	Press the **<Display>** button.
3	Press the **<Brightness Level>** button. The following options display. • Level 1 • Level 2 • Level (default)3 • Level 4 • Level 5
4	Press the level of brightness you want on your phone. Level 1 is the least bright and Level 5 is the most bright. Default is Level 3.
5	Press the to return to the previous menu or press the to return to the idle screen.

Brightness Timer (6739i only)

The "**Brightness Timer**" option allows you to set the amount of time you want the touchscreen's LCD display to stay illuminated before turning the brightness off during a period of inactivity. For example, if you set the Brightness Timer to 60, when the phone reaches 60 seconds of inactivity, the LCD brightness goes OFF.

Setting Brightness Timer (6739i only).

IP Phone UI

Step	Action
1	Press Options on the phone to enter the Options List
2	Press the **<Display>** button.
3	Press the **<Brightness Timer>** button. A text box displays.
4	Press the text box. A pop-up keyboard displays.
5	Press the "123" key and enter a value, in seconds, for the Brightness Timer. You can set the Brightness Timer from 0 to 7200 seconds. Default is 600 (10 minutes).
6	Press the to return to the previous menu or press the to return to the idle screen.

Calibrate Screen (6739i only)

The “**Calibrate Screen**” option allows you to calibrate the color touchscreen. This process makes fine adjustments to the color screen on the phone for optimal display purposes.

Caibrating the Screen (6739i only)

IP Phone UI

Step	Action
1	Press Options on the phone to enter the Options List
2	Press the **<Display>** button.
3	Press the **<Calibrate Screen>** button. A “*Recalibrate touchscreen?*” prompt displays.
4	Press “**Yes**” to continue or “**No**” to cancel the recalibration.
5	If you press “**Yes**”, the following messages display: *“Waiting for touchscreen activity to subside....”*
6	Press the screen as indicated by each prompt using a **SOFT** fine tip stylus pen. Do not use a sharp object such as an ink pen or pencil which may damage the touch screen.: *“Touch the UPPER-LEFT corner of the screen.”* *“Touch the UPPER-RIGHT corner of the screen.”* *“Touch the LOWER-RIGHT corner of the screen.”* *“Touch the LOWER-LEFT corner of the screen.”* If you touched the screen in the wrong location, the following message displays: *“Received unreliable data. You will be asked to touch the same point again.”* Retouch the screen in the correct location to continue. The following message displays: *“Validating calibration data....please wait.”* Then the following message and prompt displays: *“Sanity Check OK. Save changes?”*
7	Press the <**Yes**> button to complete the recalibration, and the phone automatically restarts.

Audio DHSG Headset (6739i, 6753i, 6755i, 6757i, 6757i CT)

DHSG is a standard for telecommunication headsets. The Aastra IP Phones 6739i, 6753i, 6755i 6757i, and 6757i CT support the use of a DHSG headset.

Aastra's DHSG headset support has been tested and verified with the following products:

- GN Netcom GN 9350, GN 9350e, GN 9120, GN 9125.
  All require the GN 14201-10 cable.
- Plantronics CS60, CS70N, CS351N, and Voyager 510S.
  All require the APS-1 cable.
- Sennheiser BW900. Requires the TCI01 adapter box.

Use of a non-verified DHSG headset solution is at the customer¡¦s own discretion and the customer should be aware that some DHSG headsets require an optional cable in order to be electrically DHSG compliant. Aastra is not responsible for any damage to the IP phone or headset that may result from the use of non-verified headsets, or from incorrectly connecting headsets or cables.

Warning: Only the Aastra DHSG cable should be connected directly to your phone. **No** 3rd party DHSG cables should be connected directly to the Aastra IP phone.

Any damage caused by connecting a 3rd party DHSG headset cable directly to your phone will void your warranty with Aastra Telecom.

Reference

For more information about installing a DHSG headset on your phone, see the *IP Phone-specific Installation Guide*.

Configuring DHSG on the Phone

You can enable or disable the use of a DHSG headset using the parameter "**dhsg**" in the configuration files, or at the location ***Options->Preferences->Set Audio->DHSG*** in the IP Phone UI. Default for DHSG is disabled (OFF).

Configuring DHSG Using the Configuration Files

Use the following information to configure the use of a DHSG headset on the IP Phones.

Configuration Files
For specific parameters you can set in the configuration files, see Appendix A, the section, "DHSG Settings (6753i, 6755i, 6757i, 6757i CT)" on page A-69.

Configuring DHSG using the IP Phone UI **(6739i, 6753i, 6755i, 6757i, 6757i CT)**

Use the following procedure to configure DHSG using the IP Phone UI.

IP Phone UI

Step	Action
1	Press on the phone to enter the Options List.
2	Select **Preferences.**
3	Select **Set Audio**.
4	Select **DHSG** and toggle the DHSG support ON or OFF
5	If you select “**Off**”, press **Done** to save your setting.
6	Press **Done** to save your setting. The setting applies immediately to the phone.
For the 6739i:	
1	Press on the phone to enter the Options List.
2	Press the <**Audio**> button.
3	Press the **<Headset Device>** button. The following values display: • Wired (default) • DHSG • Bluetooth
4	Press the value you want to set for the headset device.
5	Press the to return to the previous menu or press the to return to the idle screen.

Audio Hi-Q on G.722 Calls

The Aastra IP phones support the Hi-Q (high quality) audio technology which delivers enhanced performance and voice clarity for Aastra’s 6700i series of SIP phones. Incorporating wideband audio technology, Aastra Hi-Q significantly improves the audio quality of calls. This technology provides a more lifelike conversation when the G.722 wideband Codec is used. The phones with 5-line LCD screens display a large icon when Hi-Q is being used on the call. On 3-line LCD screens, the text of “**Hi-Q**” displays on the same line as the call timer.

Live Dial Pad*

The "Live Dialpad" option on the IP phone turns the Live Dial Pad mode ON or OFF. With live dial pad ON, the IP phone automatically dials out and turns ON Handsfree mode as soon as a dial pad key or softkey is pressed. With live dial pad OFF, if you dial a number while the phone is on-hook, lifting the receiver or pressing the initiates a call to that number.

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

A User can turn the "Live Dialpad" ON and OFF using the IP Phone UI only. A System Administrator can turn it ON and OFF using the IP Phone UI or the configuration files.

Enabling/Disabling Live Dialpad Using the Configuration Files

Use the following procedure to enable/disable Live Dialpad on the IP Phones.

Configuration Files

For specific parameters you can set in the configuration files, see Appendix A, the section, "Live Dialpad Settings" on page A-69.

Enabling/Disabling Live Dialpad Using the IP Phone UI

Use the following procedure to enable/disable Live Dialpad on the IP Phones.

IP Phone UI

Step	Action
1	Press on the phone to enter the Options List.
2	Select **Preferences**.
3	Select **Live Dialpad**.
4	**On 3-Line LCD phones:** Press the **UP** and **DOWN** arrow keys to toggle the live dialpad setting ON or OFF. **On 8 and 11-Line LCD phones**: Press **Change** to toggle the live dialpad setting ON or OFF.
5	Press **Done** to save the setting.
For the 6739i:	
1	Press on the phone to enter the Options List.
2	Press the **<Audio>** button.
3	Press the **<Headset Device>** button. The following values display: • Wired (default) • DHSG • Bluetooth
4	Press the value you want to set for the headset device.
5	Press the to return to the previous menu or press the to return to the idle screen.

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:

Available Language	**Associated Language File** (included in the firmware file when downloaded from configuration server)
English	Default (resides on the phone)
German*	lang_de.txt
Danish*	lang_dk.txt
Spanish	lang_es.txt
Mexican Spanish	lang_es_mx.txt
Finnish*	lang_fi.txt
French	lang_fr.txt
Canadian French	lang_fr_ca.txt
Italian*	lang_it.txt
Portuguese*	lang_pt.txt
Portuguese Brazilian*	lang_pt_br.txt
Russian*	lang_ru.txt
Swedish*	lang_sv.txt

*This language is not applicable to the 6757i 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-179) and <ISO 3166> is the country code specified in Standard ISO 3166 (see Country Codes (from Standard ISO 3166) on page A-179).
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.

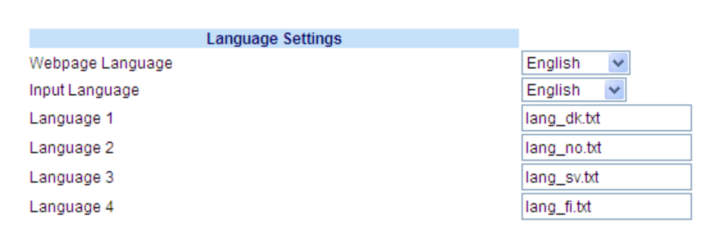
References

For more information about specifying the language to use, see the section, "Specifying the Screen Language to Use" on page 5-43.

For more information about language codes and country codes, see Appendix A, the section, "Language Pack Settings" on page A-177.

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 Screen 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

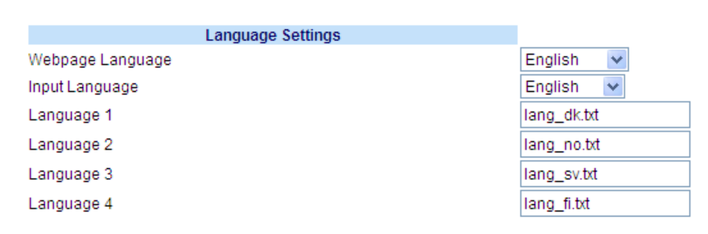
For specific parameters you can set in the configuration files, see Appendix A, the section, "Language Settings" on page A-175 and "Language Pack Settings" on page A-177.

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 [Options key] on the phone to enter the Options List.
2	Select **Language**.
3	Select **Screen Language**.
4	• Select **English** (English - default), **Danish, Finnish, Français** (French), **Español** (Spanish), **Deutsch** (German), **Italiano** (Italian), **Norwegian**, **Português** (Portuguese), **Русский** (Russian), and **Sweedish** **Notes**: **1**. Valid values for the 6757i CT cordless handset 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 5-40.
5	Press **Done** to save the changes. The change is dynamic. The IP phone UI displays all menu items in the language you chose.
For the 6739i:	
1	Press [Options key] on the phone to enter the Options List.

IP Phone UI	
Step	**Action**
2	Press the <**Language>** button.
3	Press the **<Screen Language>** button. The following options display: • **English** (English - default) • **Danish** • **Finnish** • **Français** (French) • **Español** (Spanish) • **Deutsch** (German) • **Italiano** (Italian) • **Norwegian** • **Português** (Portuguese) • **Русский** (Russian) • **Swedish** **Note:** 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 5-40.
4	Press the screen language you want to set on the phone. The change takes place immediately.
5	Press the to return to the previous menu or press the to return to the idle screen.

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

	Aastra Web UI
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 any of the following in the “Language 1”, “Language 2”, “Language 3”, and “Language 4” fields to display the Aastra Web UI in Danish, Finnish, French, Spanish, German, Italian, Norwegian, Portuguese, Russian, or Swedish **lang_de.txt** **lang_dk.txt** **lang_es.txt** **lang_es_mx.txt** **lang_fi.txt** **lang_fr.txt** **lang_fr_ca.txt** **lang_it.txt** **lang_no.txt** **lang_pt.txt** **lang_pt_br.txt** **lang_ru.txt** **lang_sv.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 5-40. For more information about language codes and country codes, see Appendix A, the section, “Language Pack Settings” on page A-177.
3	Click [Save Settings] 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.**
6	In the “**Webpage Language**” field, select a language to apply to the Aastra Web UI. Valid values are: • **English** (English - default) • **Danish** • **Finnish** • **Français** (French) • **Español** (Spanish) • **Deutsch** (German) • **Italiano** (Italian) • **Norwegian** • **Português** (Portuguese) • **Русский** (Russian) • **Swedish** **Notes**: **1**. Valid values for the 6757i CT cordless handset 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 5-40.
7	Click [Save Settings] to save your changes. The Aastra Web UI displays all screens in the language you chose.

Specifying the Input Language to Use

The phones support text and character inputs in various languages (English, German, French, Spanish, and Italian).

Inputting textual or character information into the IP Phone UI, Aastra Web UI, and XML scripts can now be done in various languages using the keypad on the phone. The System Administrator and User can enable this feature using the Aastra Web UI or the IP Phone UI. An Administrator can also use the configuration files to enable this feature. Users can then use text and characters in a specific language when performing inputs on the phone.

The following tables identify the language characters that a User can enter on the 5i Series phones that support the Input Language feature.

Keypad Text/Character Input Tables

***English** (default)*

Key	Uppercase Characters	Lowercase Characters
0	0	0
1	1;=_,-'&()	1.:;=_,-'&()
2	ABC2	abc2
3	DEF3	def3
4	GHI4	ghi4
5	JKL5	jkl5
6	MNO6	mno6
7	PQRS7	pqrs7
8	TUV8	tuv8
9	WXYZ9	wxyz9
*	* <SPACE>	* <SPACE>
#	#/\@	#/\@

French

Key	Uppercase Characters	Lowercase Characters
0	0	0
1	1.:;=_,-'&()	1.:;=_,-'&()
2	ABC2ÀÂÇÁÅÆ	abc2àâçáåæ
3	DEF3ÉÈÊË	def3éèêë
4	GHI4ÎÏ	ghi4îï
5	JKL5	jkl5
6	MNO6ÑÓÒÔÖ	mno6ñóòôö
7	PQRS7	pqrs7
8	TUV8ÚÙÛÜ	tuv8úùûü
9	WXYZ9	wxyz9
*	* <SPACE>	* <SPACE>
#	#/\@	#/\@

Spanish

Key	Uppercase Characters	Lowercase Characters
0	0	0
1	1.:;=_,-'&()	1.:;=_,-'&()
2	ABC2ÁÀÇ	abc2áàç
3	DEF3ÉÈ	def3éè
4	GHI4ÏÍ	ghi4ïí
5	JKL5	jkl5
6	MNO6ÑÓÒ	mno6ñóò
7	PQRS7	pqrs7
8	TUV8ÚÜ	tuv8úü
9	WXYZ9	wxyz9
*	* <SPACE>	* <SPACE>
#	#/\@	#/\@

German

Key	Uppercase Characters	Lowercase Characters
0	0	0
1	1.:;=_,-'&()	1.:;=_,-'&()
2	ABC2ÄÀ	abc2äà
3	DEF3É	def3é
4	GHI4	ghi4
5	JKL5	jkl5
6	MNO6Ö	mno6ö
7	PQRS7ß	pqrs7ß
8	TUV8Ü	tuv8ü
9	WXYZ9	wxyz9
*	* <SPACE>	* <SPACE>
#	#/\@	#/\@

Italian

Key	Uppercase Characters	Lowercase Characters
0	0	0
1	1.:;=_,-'&()	1.:;=_,-'&()
2	ABC2ÀCÇ	abc2àcç
3	DEF3ÉÈË	def3éèë
4	GHI4	ghi4
5	JKL5	jkl5
6	MNO6ÓÒ	mno6óò
7	PQRS7	pqrs7
8	TUV8Ù	tuv8ù
9	WXYZ9	wxyz9
*	* <SPACE>	* <SPACE>
#	#/\@	#/\@

Portguese

Key	Uppercase Characters	Lowercase Characters
0	0	0
1	1.:;=_,-'&()	1.:;=_,-'&()
2	ABC2ÁÀÂÃÇ	abc2áàâãç
3	DEF3ÉÊ	def3éê
4	GHI4Í	ghi4í
5	JKL5	jkl5
6	MNO6ÓÔÕ	mno6óôõ
7	PQRS7	pqrs7
8	TUV8ÚÜ	tuv8úü
9	WXYZ9	wxyz9
*	* <SPACE>	* <SPACE>
#	# / \ @	# / \ @

Russian

Key	Uppercase Characters	Lowercase Characters
0	0	0
1	1.:;=_,-'&()	1.:;=_,-'&()
2	АБВГ2ABC	абвг2abc
3	ДЕЁЖЗ3DEF	Деёжз3def
4	ИЙКЛ4GHI	ийкл4ghi
5	МНОП5JKL	мноп5jkl
6	РСТУ6MNO	рсту6mno
7	ФХЦЧ7PQRS	фхЧч7pqrs
8	ШЩЪЫ8TUV	шщъы8tuv
9	ЬЗЮЯ9WXYZ	ьзюя9wxyz
*	* <SPACE>	* <SPACE>
#	#/\@	#/ \@

Configuring Language Input Using the Configuration Files

An Administrator can specify the input language to use by entering a specific parameter in the configuration files. An Administrator must enter the following parameter to enable this feature:

- **input language**

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

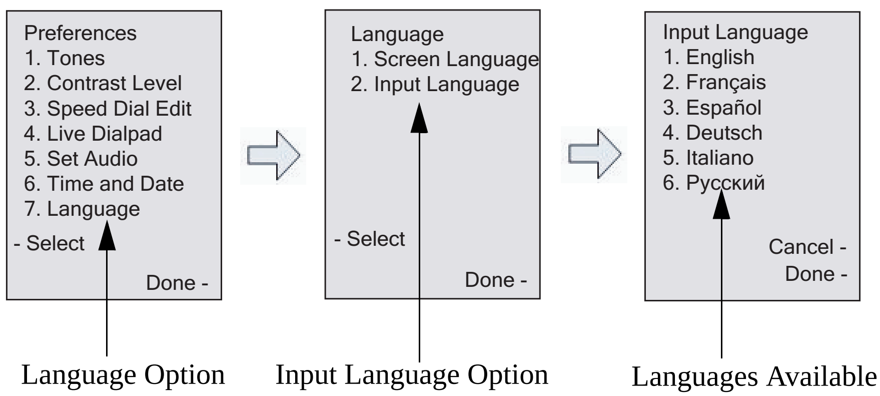
Configuration Files

For specific parameters you can set in the configuration files, see Appendix A, the section, "Language Settings" on page A-175.

Configuring Language Input Using the IP Phone UI

Once "Language Input" is enabled, an Administrator or User can change the input language on the phone using the IP Phone UI. The "**Input Language**" option appears under the Language option in the IP Phone UI.

Example



Use the following procedure to change the input language using the IP Phone UI.

Aastra IP Phone UI

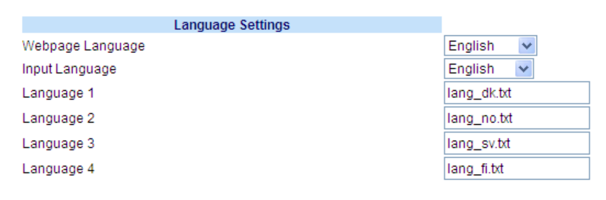
Step	Action
1	Press the **Options** key.
2	Select **Language** from the Options List.
3	Select **Input Language** from the Language List.

Aastra IP Phone UI	
Step	**Action**
4	Select the language you want to use on the IP phone for inputting text and characters. Valid values are: • **English** (default) • **Français** (French) • **Español** (Spanish) • **Deutsch** (German) (not applicable to 6757i CT cordless handset) • **Italiano** (Italian) (not applicable to 6757i CT cordless handset) • **Русский** (Russian)) (not applicable to 6757i CT cordless handset) • **Nordic**
5	Press **Done** when you have selected a language.
For the 6739i:	
1	Press  on the phone to enter the Options List.
2	Press the <**Language>** button.
3	Press the **<Input Language>** button. The following options display: • **English** (English - default) • **Français** (French) • **Español** (Spanish) • **Deutsch** (German) • **Italiano** (Italian) • **Português** (Portuguese) • **Русский** (Russian) • **Nordic**
4	Press the input language you want to set on the phone. The change takes place immediately.
5	Press the  to return to the previous menu or press the  to return to the idle screen.

Configuring Language Input Using the Aastra Web UI

Once “Language Input” is enabled, an Administrator or User can also change the input language on the phone using the Aastra Web UI. The “**Input Language**” option appears at the path ***Basic Settings->Preferences->Language Settings.***

Use the following procedure to set the input language using the Aastra Web UI.

	Aastra Web UI
1	Click on **Basic Settings-> Preferences->Language Settings.** 
2	Select a language from the "**Input Language**" field. Setting this field allows you to specify the language to use when entering text in the Aastra Web UI, IP Phone UI, or in XML applications on the phone. Valid values are: • **English** (default) • **Français** (French) • **Español** (Spanish) • **Deutsch** (German) (not applicable to 6757i CT cordless handset) • **Italiano** (Italian) (not applicable to 6757i CT cordless handset) • **Русский** (Russian)) (not applicable to 6757i CT cordless handset) • **Nordic** **Note:** Available input languages are dependent on the configuration enabled by your System Administrator.
3	Click Save Settings to save your changes.

Configuring Language Input for an XML Application

A System Administrator can enable input languages in XML applications using the **<AastraIPPhoneInputScreen>** object and the “**inputLanguage**” attribute.

Reference

For more information about using XML objects for defining input language, contact Aastra Telecom Customer Support regarding the “Aastra XML Development Guide.”

UTF- 8 Codec for Multi-National Language Support

The IP Phones and expansion modules include support for ISO 8859-2 (Latin2) of multi-national languages when displaying and inputing in the IP Phone UI and the Aastra Web UI.

Note: This feature is not applicable to the handsets on the 6757i CT and the 9480i CT.

UTF-8 is also compatible with XML encoding on the IP Phones.

The following table illustrates the Latin 2 character set now used on the IP Phones.

	–0	–1	–2	–3	–4	–5	–6	–7	–8	–9	–A	–B	–C	–D	–E	–F
A–	NBSP 00A0 160	Ą 0104 161	˘ 02D8 162	Ł 0141 163	¤ 00A4 164	Ľ 013D 165	Ś 015A 166	§ 00A7 167	¨ 00A8 168	Š 0160 169	Ş 015E 170	Ť 0164 171	Ź 0179 172	SHY 00AD 173	Ž 017D 174	Ż 017B 175
B–	° 00B0 176	ą 0105 177	˛ 02DB 178	ł 0142 179	´ 00B4 180	ľ 013E 181	ś 015B 182	ˇ 02C7 183	¸ 00B8 184	š 0161 185	ş 015F 186	ť 0165 187	ź 017A 188	˝ 02DD 189	ž 017E 190	ż 017C 191
C–	Ŕ 0154 192	Á 00C1 193	Â 00C2 194	Ă 0102 195	Ä 00C4 196	Ĺ 0139 197	Ć 0106 198	Ç 00C7 199	Č 010C 200	É 00C9 201	Ę 0118 202	Ë 00CB 203	Ě 011A 204	Í 00CD 205	Î 00CE 206	Ď 010E 207
D–	Đ 0110 208	Ń 0143 209	Ň 0147 210	Ó 00D3 211	Ô 00D4 212	Ő 0150 213	Ö 00D6 214	× 00D7 215	Ř 0158 216	Ů 016E 217	Ú 00DA 218	Ű 0170 219	Ü 00DC 220	Ý 00DD 221	Ţ 0162 222	ß 00DF 223
E–	ŕ 0155 224	á 00E1 225	â 00E2 226	ă 0103 227	ä 00E4 228	ĺ 013A 229	ć 0107 230	ç 00E7 231	č 010D 232	é 00E9 233	ę 0119 234	ë 00EB 235	ě 011B 236	í 00ED 237	î 00EE 238	ď 010F 239
F–	đ 0111 240	ń 0144 241	ň 0148 242	ó 00F3 243	ô 00F4 244	ő 0151 245	ö 00F6 246	÷ 00F7 247	ř 0159 248	ů 016F 249	ú 00FA 250	ű 0171 251	ü 00FC 252	ý 00FD 253	ţ 0163 254	˙ 02D9 255

Minimum Ringer Volume

To prevent the user from turning off the ringer, an Administrator can configure a parameter called "**ringer volume minimum**" to set the minimum ringer volume level. When the minimum ringer level is reached while the user keeps pressing the button, the level of sound does not change.

Note: This minimum ringer volume does not affect the "silent" ring tone. When the silent ring tone is selected, no ringing will be played by the phone

Configuration Files

For the specific parameter you can set in the configuration files, see Appendix A, the section, "Minimum Ringer Volume" on page A-195.

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 only. 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/or programmable keys on all the phones. 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	9480i 9480i CT 6739i 6755i 6757i 6757i CT
topsoftkeyN locked	topsoftkeyN type topsoftkeyN label topsoftkeyN value topsoftkeyN line	6757i 6757i CT
prgkeyN locked	prgkeyN type prgkeyN value prgkeyN line	9143i 6730i 6731i 6753i 6755i
featurekeyN locked	featurekeyN type featurekeyN label	9480i CT 6757i CT
expmodX keyN locked	expmodX keyN type expmodX keyN value expmodX keyN line	6700i-series phones only (not applicable to the 6730i, and 6731i)

Note: The 3-Line LCD phones prevent users from setting a speed dial key via the Phone UI on a key that has been locked.

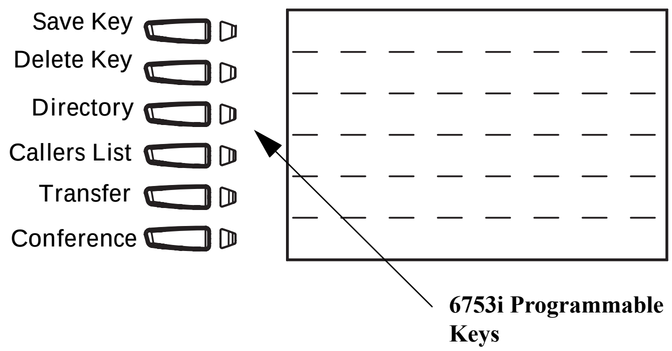
Locking the IP Phone Keys using the Configuration Files

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-231.

Locking/Unlocking the SAVE and DELETE keys (6753i)

There are 4 programmable keys on the 6753i phone located to the left of the paper label. Two additional keys (SAVE and DELETE) can be made programmable by the Administrator, providing a total of 6 programmable keys if required.



If a System Administrator unlocks the **SAVE** and **DELETE** keys, these keys can be configured with the same functions that are available for the other programmable keys. Only the System Administrator can unlock these keys.

The **Save** key allows you to save entries to the Directory and perform a Save-To from the Callers List. It also allows you to save speed dial information to a programmable key. You can also use the **Save** key while using specific XML applications.

The **Delete** key allows you to remove entries from the Directory List and Callers List. (Must enter the Directory or Callers list and select an entry, then press twice to delete entry).

By default, the **Save** and **Delete** keys are locked so that a user can use them for saving and deleting only. An Administrator can unlock these keys using the configuration files, allowing the keys to be programmed with other functions if required. An Administrator can use the following parameters in the configuration file to lock and unlock the **Save** and **Delete** keys:

- `prgkey1 locked`
- `prgkey2 locked`

The value of "**0**" unlocks the keys, and the value of "**1**" locks the keys. The default is "1" (lock).

The following is an example of unlocking the **Save** and **Delete** keys using the configuration files:

Example:

- **`prgkey1 locked: 0`**
- **`prgkey2 locked: 0`**

Once the **Save** and **Delete** keys are unlocked, a User can change the function of the keys using the Aastra Web UI. An Administrator can change the function of the keys using the Aastra Web UI or the configuration files.

IMPORTANT: If you change the functions of the **Save** and **Delete** keys, and then an Administrator locks Keys 1 and 2, the functions are automatically set back to the default settings of "**Save**" and "**Delete**".

Note: The **Save** and **Delete** functions are limited to Key 1 and Key 2 on the 6753i IP phone only.

Locking and Unlocking the Save and Delete Keys using the Configuration Files.

Configuration Files
For the specific parameter you can set in the configuration files, see Appendix A, the section, "Locking the SAVE and DLETE Keys (6753i)" on page A-234.

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 Dial Plan field accepts up to 512 characters. If a User enters a dial plan longer than 512 characters, or a parsing error occurs, the phone uses the default dial plan of **“x+#|xx+*”.** 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
;	Allows a secondary dial tone to be audible before dialing a number.
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]
;	Used when a secondary dial tone is required on the phone. (For example, “9;xxxxxx”, when a user has to dial “9” to get and outside line and needs a secondary dial one presented

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.

Secondary Dial Tone

The IP phones now support a feature that allows the user to dial a predefined dial string, obtain a dial tone, and continue dialing. A User or Administrator can configure this using the existing Dial Plan feature on the phone.

You can enter a new character string in the dial plan that allows you to configure the secondary dial tone. The character string is of the form "**.;.**" , where the period indicates an arbitrary number of digits and the semicolon indicates that the phone is to present a dial tone after the previous dialed digit. For example, in the string:

"9;xxxxx"

the user dials “**9**” to get the outside line, listens for the dial tone, and continues to dial the applicable number. The “**;**” tells the phone to present a second dial tone after the previously dialed digit. “The “xxxxx” in the example tells the phone that a phone number is dialed after the secondary dial tone is audible.

You can enter the Secondary Dial Tone string in the Dial Plan using the configuration files or the Aastra Web UI.

You use the following parameter in the configuration files to configure a secondary dial tone:

- **sip dial plan**

Example

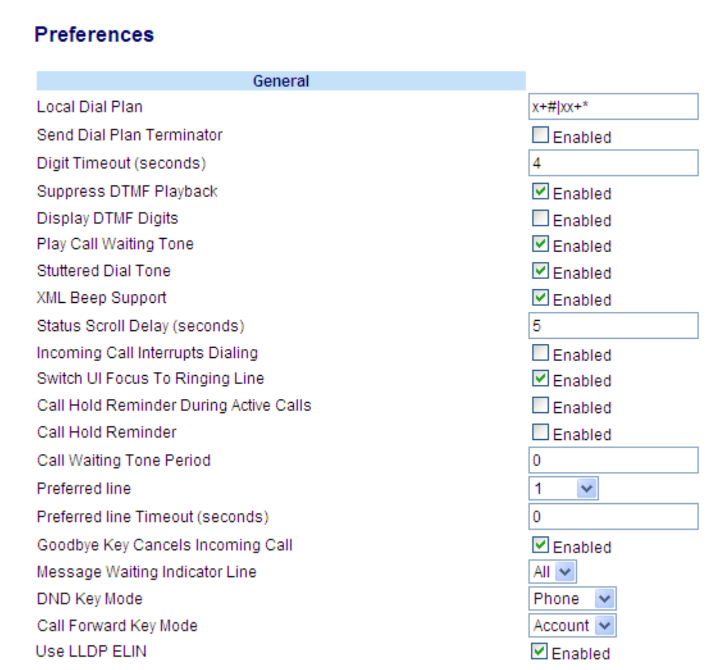
sip dial plan: “9;5551313”

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-70.

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



</td></tr>
<tr><td>2</td><td>In the "Local Dial Plan" field, enter a valid local dial plan (up to 512 alphanumeric characters) for the IP phone. Default is X+#|XX+*. Enter prepended digits or a ";"to present a secondary dial tone if required.

Note: If a User enters a dial plan longer than 512 characters, or a parsing error occurs, the phone uses the default dial plan of "x+#|xx+*".</td></tr>
<tr><td>3</td><td>Enable the "Send Dial Plan Terminator" field by checking the check box. (Disable this field by unchecking the box). Default is disabled.</td></tr>
<tr><td>4</td><td>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.</td></tr>
<tr><td>5</td><td>Click Save Settings to save your changes.</td></tr>
</table>

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.

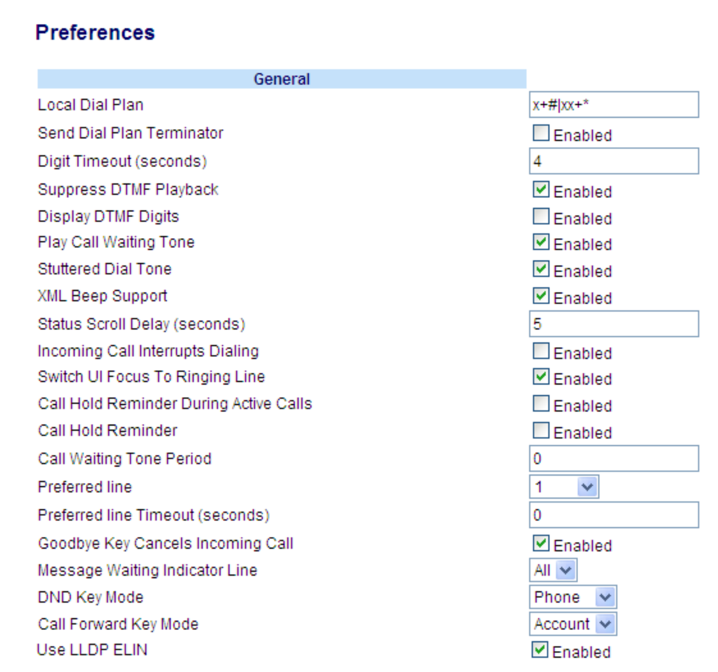
DTMF playback suppression is enabled by default. Suppressing DTMF playback can be configured using the Aastra Web UI and the configuration files.

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-185.

<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>Enable the "Suppress DTMF Playback" field by checking the check box.
(Disable this field by unchecking the box). Default is enabled.</td></tr>
<tr><td>3</td><td>Click Save Settings to save your settings.

You must restart your IP phone for the changes to take affect.</td></tr>
<tr><td>4</td><td>Click on Operation->Reset.</td></tr>
<tr><td>5</td><td>In the "Restart Phone" field click Restart to restart the IP phone and apply the changes.</td></tr>
</table>

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.

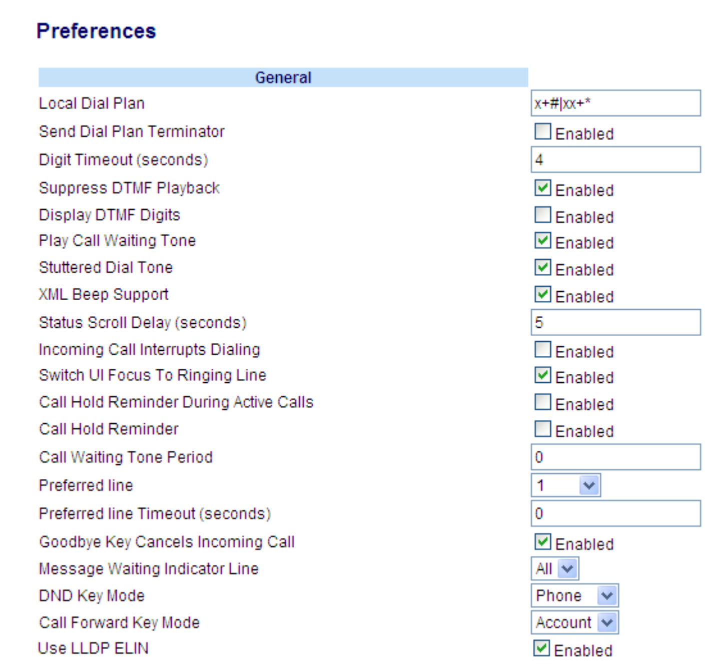
If you enable the Display DTMF Digits parameter, the digits you are dialing from the keypad or from a softkey or programmable key display to the IP phone's LCD display. This parameter is disabled by default (no digits display when dialing).

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-186.

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

</td></tr>
<tr><td>2</td><td>Enable the "Display DTMF Digits" field by checking the check box.
(Disable this field by unchecking the box). Default is disabled.</td></tr>
<tr><td>3</td><td>Click Save Settings to save your settings.

You must restart your IP phone for the changes to take affect.</td></tr>
<tr><td>4</td><td>Click on Operation->Reset.</td></tr>
<tr><td>5</td><td>In the "Restart Phone" field click Restart to restart the IP phone and apply the changes.</td></tr>
</table>

Call Waiting

The call waiting feature notifies a user on an active call 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 6757i 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 on a global or per-line basis using the configuration files or the Aastra Web UI.

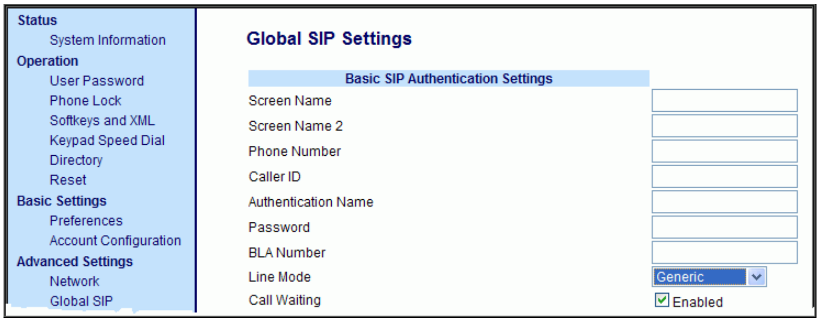
Configuring Call Waiting

Use the following procedures to configure the Call Waiting feature on the IP phone.

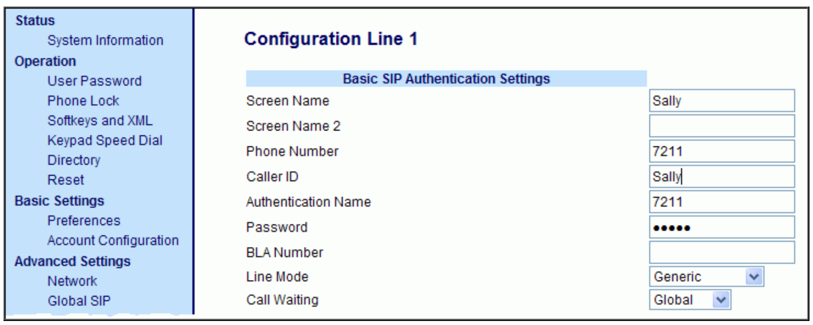
Configuration Files
For specific parameters you can set in the configuration files, see Appendix A, the section, “Call Waiting Settings” on page A-78 or “SIP Per-Line Call Waiting Setting” on page A-91.

Aastra Web UI

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



Or, for per-line configuration, click on **Advanced Settings->Line *N* (1-9)0->Basic SIP Authentication Settings.**



Configure Global Call Waiting

2 The "**Call Waiting**" field is enabled by default. To disable call waiting on a global basis, uncheck this box.

Configure Per-Line Call Waiting

3 The "**Call Waiting**" field is set to “**Global**” by default. To enable call waiting for a specific line, select “**enabled**” from the list in this field. To disable call waiting for a specific line, select “**disabled**” from the list in this field.

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

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 using the “**Play Call Waiting Tone**” parameter. 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.

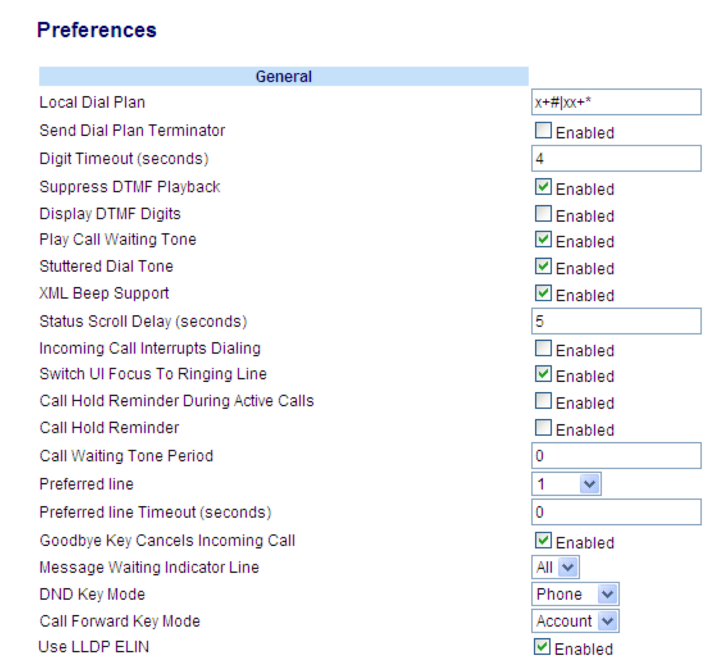
A User or Administrator can configure this feature using the Aastra Web UI. An Administrator can also configure this feature using the configuration files.

Configuring Call Waiting Tone

Use the following procedures to configure the Call Waiting Tone feature 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-78.

<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 "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 at the location Advanced Settings->Global SIP (or Line X)->Basic SIP Authentication Settings.</td></tr>
<tr><td>3</td><td>Click Save Settings to save your changes.</td></tr>
</table>

Call Waiting Tone Period

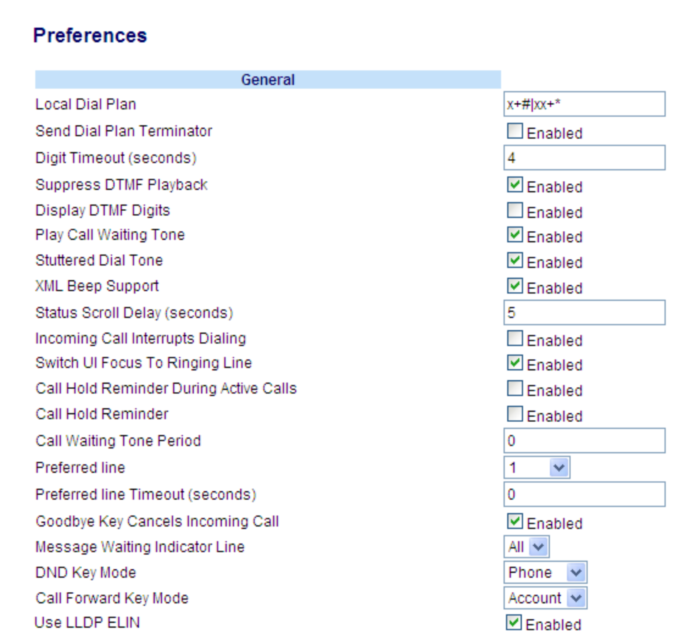
A User or Administrator can specify a specific time period (in seconds) for the call waiting tone to play at regular intervals on an active call using the parameter "**call waiting tone period**". A value of "0" is the default and plays the call waiting tone only once on the active call. When the incoming caller hangs up, the call waiting tone stops on the existing active call.

You can enable or disable this feature in the configuration files or in the Aastra Web UI.

Configuring "Call Waiting Tone Period"

You use the following procedures to enable or disable "Call Waiting Tone Period".

Configuration Files
For specific parameters you can set in the configuration files, see Appendix A, the section, "Call Waiting Settings" on page A-78.

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

</td></tr>
<tr><td>2</td><td>In the "Call Waiting Tone Period" field,enter a time period, in seconds, that the call waiting tone will be audible on an active call when another call comes in. Default is 0 seconds.

When enabled, the call waiting tone plays at regular intervals for the amount of time set for this parameter. For example, if set to "30" the call waiting tone plays every 30 seconds. When set to "0", the call waiting tone is audible only once on the active call</td></tr>
<tr><td>3</td><td>Click Save Settings to save your changes.</td></tr>
</table>

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.

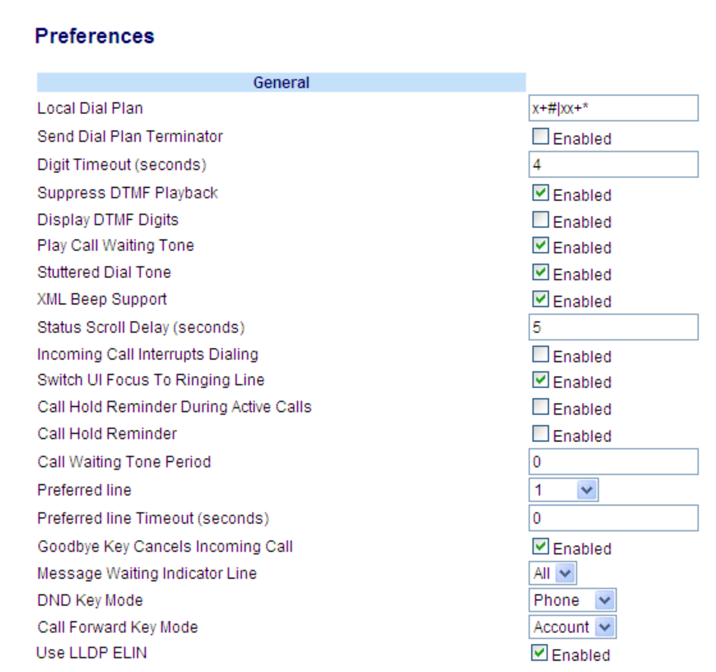
You can configure this feature using the configuration files and the Aastra Web UI.

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-163.

Aastra Web UI

1	Click on **Basic Settings->Preferences->General.** 
2	The "**Stuttered Dial Tone**" field is enabled by default. To disable this field, uncheck the box.
3	Click Save Settings to save your changes.

XML Beep Support

The IP phones have a feature that allows you to enable or disable a beep on the phone when 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. If you disable this feature, then no beep is heard when the XML application arrives to the phone.

If your System Administrator has set a value for this feature in a custom XML application or in the configuration files, the value you set in the Aastra Web UI overrides the Administrator's setting. Setting and saving the value in the Aastra Web UI applies to the phone immediately.

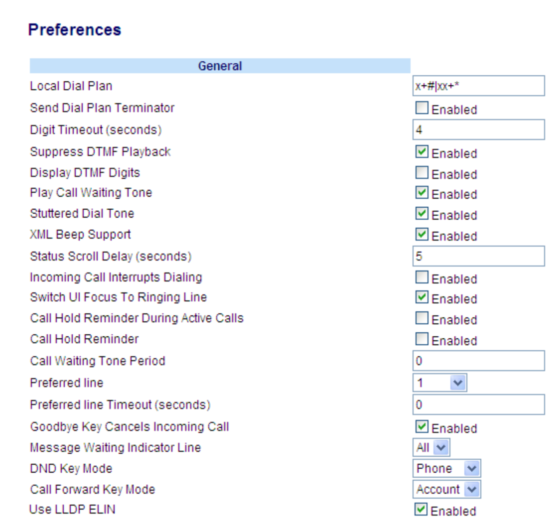
Reference

For more information about enabling/disabling the XML Beep Support in an XML script, see "XML Customized Services" on page 5-284.

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-145.

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.

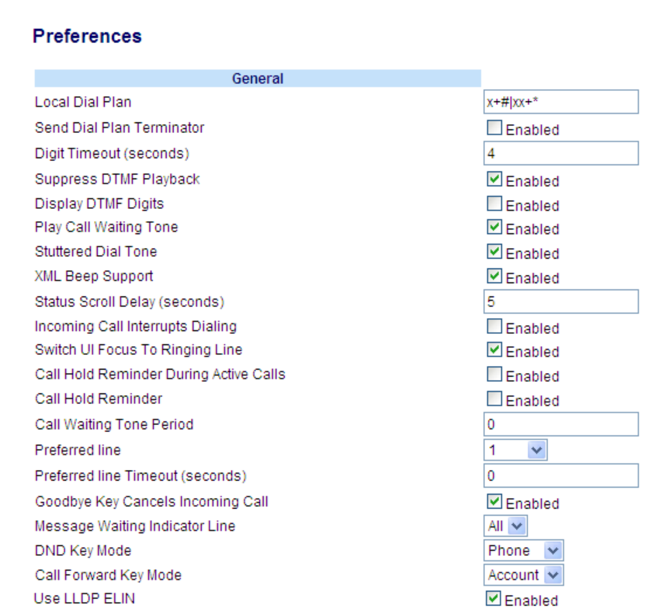
Reference

For more information about configuring the status scroll delay for XML status messages, see "XML Customized Services" on page 5-284.

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-145.

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.

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"** (Web UI) parameter or **"incoming call cancels dialing"** (in config file) 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 this parameter (0 = disable), which is the default, the phone does not interrupt the outgoing call during dialing and the number you were dialing continues to display in the LCD. The phones sends the incoming call to a free line on the phone (or sends busy signal if all remaining lines are busy) and the LED for that line blinks. 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 the 3-Line LCD display phones, 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.

3. This feature works only if the User selects a line for which to dial out. It is recommended that the Administrator always keeps Live Dialpad ON in order for the User to have to select a line before dialing out.

Transfer/Conference Call Behavior

If you are dialing the phone to transfer or conference a call, and your phone receives an incoming call, your dialing is never interrupted (regardless of whether the "Incoming Call Interrupts Dialing" is enabled or disabled). For Transfer and Conference, the incoming calls always go to an available line (other than the one you are using for dialing) and the incoming call's line LED blinks. The LCD still displays your dialing screen.

Intercom Behavior

If “**Incoming Call Interrupts Dialing”** (Web UI) or **“incoming call cancels dialing**” (config file) is enabled and you are dialing an outgoing Intercom call, the enabled interrupt setting takes precedence over an enabled “**Allow Barge In**” setting. The incoming call interrupts your dialing on an outgoing intercom call. On an incoming intercom call, the enabled “**Allow Barge In**” and “**Auto-Answer**” occurs while you are dialing to transfer or conference the call. However, the incoming call goes to an available idle line, and the LED blinks while you are dialing the second half of the conference or transfer.

If “**Incoming Call Interrupts Dialing”** (Web UI) or **“incoming call cancels dialing”** (config file) is disabled, an incoming intercom goes to an available idle line and the LED blinks for that line. The phone answers the call under all conditions.

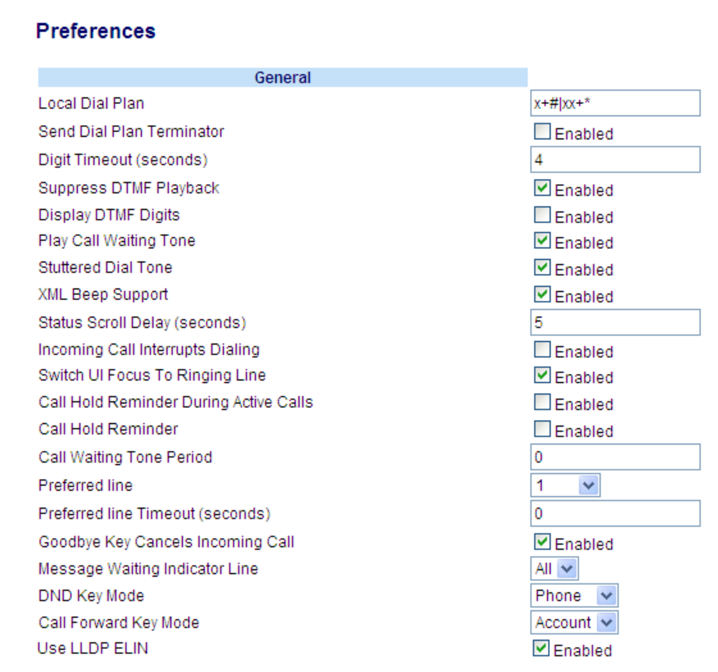
Configuring Incoming Call Interrupts Dialing

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-156.

<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>

Switch Focus to Ringing Line

An Administrator or User can control the behavior of the phone when it receives an incoming call when it is already in a connected call. By default, the phone switches focus to the ringing line to enable the user to see who is calling them.

You can turn off this functionality so that the phone now stays focused on the connected call. You can do this using the "**switch focus to ringing line**" parameter in the configuration files or the Aastra Web UI.

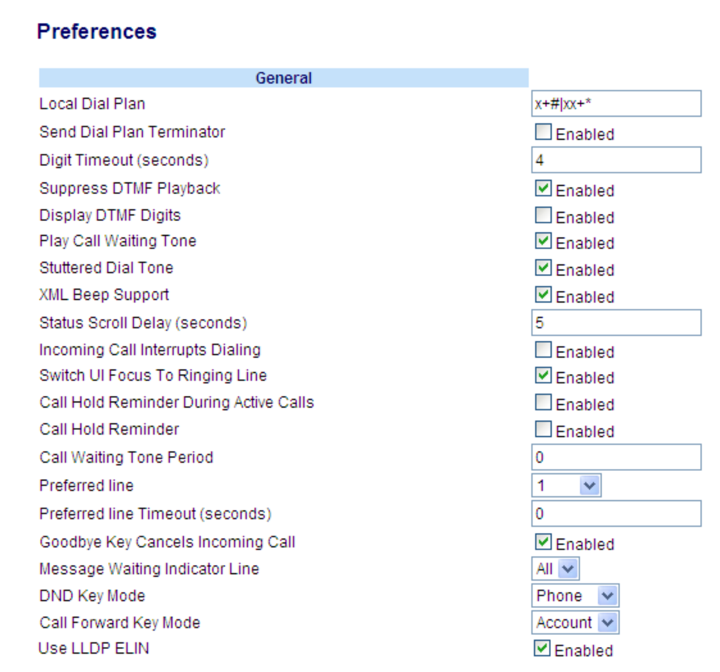
Note: If you configure the BLF/Xfer key(s) and/or Speed Dial/Xfer key(s) on the phone, you can enable or disable the switching of the user interface focus to ringing line while the phone is in the connected state.

Configuring "Switch Focus to Ringing Line"

You use the following procedures to enable or disable "Switch Focus to Ringing Line".

Configuration Files

For specific parameters you can set in the configuration files for enabling disabling "Switch Focus to Ringing Line", see Appendix A, the section, "Switch Focus to Ringing Line" on page A-158.

	Aastra Web UI
1	Click on **Basic Settings->Preferences->General.** . 
2	The "**Switch Focus to Ringing Line**" field is enabled by default. To disable this field, uncheck the box.
3	Click  to save your changes.

Call Hold Reminder During Active Calls

The IP phones allow a User or Administrator to enable or disable the ability for the phone to initiate a continuous reminder tone on the active call when another call is on hold. For example, when this feature is enabled, and the call on Line 1 is on hold, and then the User answers a call on Line 2 and stays on that line, a reminder tone is played in the active audio path on Line 2 to remind the User that there is still a call on hold on Line 1.

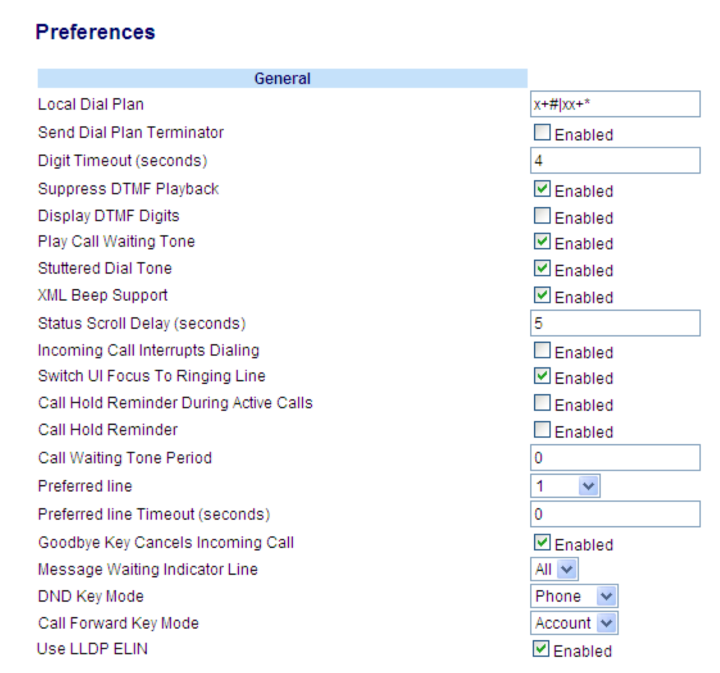
When this feature is disabled, a ring splash is heard when the active call hangs up and there is still a call on hold.

Your can enable or disable this feature using the "call hold reminder during active calls" parameter in the configuration files or in the Aastra Web UI.

Configuring "Call Hold Reminder During Active Calls"

You use the following procedure to enable or disable "Call Hold Reminder During Active Calls".

Configuration Files
For the specific parameter you can set in the configuration files for enabling/disabling "Call Hold Reminder During Active Calls", see Appendix A, the section, "Call Hold Reminder During Active Calls" on page A-158.

<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>In the "Call Hold Reminder During Active Calls" field, enable this feature by placing a check mark in the box.

When this feature is enabled, a reminder tone is heard on the active call when another call is on hold. When disabled, a ring splash is heard when the active call hangs up and there is still a call on hold.</td></tr>
<tr><td>3</td><td>Click Save Settings to save your changes.</td></tr>
</table>

Call Hold Reminder (on single hold)

In previous releases, the call hold reminder ring splash was triggered when you hung up a call and there was at least one other call on hold. The reminder ring splash timer started only when the active call hung up and there was still another call on hold.

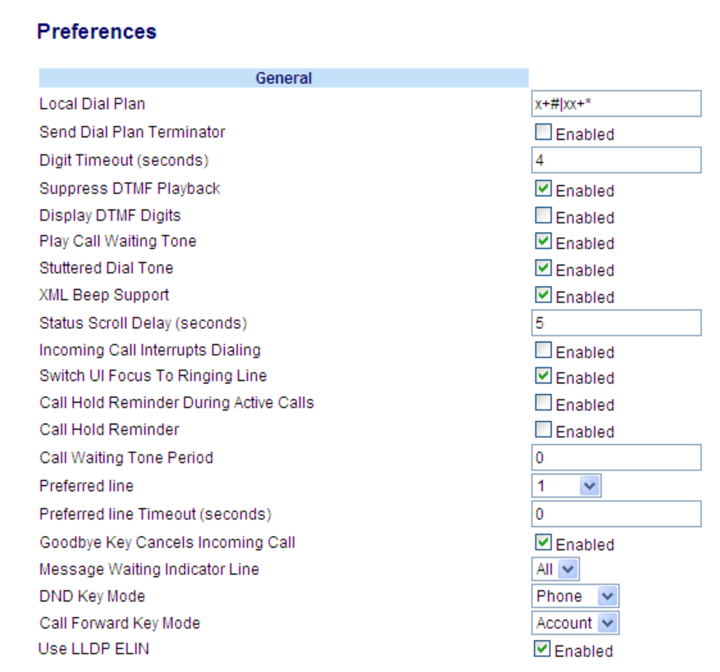
On the IP phones, a User or Administrator can enable or disable a feature that would start the reminder ring splash timer as soon as you put a call on hold (even when no other calls are active on the phone). When enabled, the phone initiates a reminder ring splash periodically for the single call on hold. When disabled, no reminder ring splash is audible.

You can enable or disable this feature using the "call hold reminder" parameter in the configuration files or in the Aastra Web UI.

Configuring "Call Hold Reminder"

You use the following procedure to enable or disable "Call Hold Reminder".

Configuration Files
For the specific parameter you can set in the configuration files for enabling/disabling "Call Hold Reminder", see Appendix A, the section, "Call Hold Reminder" on page A-159.

<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 "Call Hold Reminder" field is disabled by default. To enable this feature, place a check mark in the box.

When this feature is enabled, the reminder ring splash timer starts as soon as you put a call on hold (even when no other calls are active on the phone).The phone initiates a reminder ring splash periodically for the single call on hold. When disabled, no reminder ring splash is audible.</td></tr>
<tr><td>3</td><td>Click 

to save your changes.</td></tr>
</table>

Call Hold Reminder Timer & Frequency

There are two parameters an Administrator can set on the IP Phones along with the "call hold reminder" and "call hold reminder during active calls" parameters:

- call hold reminder timer
- call hold reminder frequency

These parameters specify the time delay and time frequency of the ring splash that sounds when you are on an active call and have placed another call on hold. You can configure these parameters using the configuration files only.

Notes:
1. You must enable the "call hold reminder" and/or "call hold reminder during active calls" parameter(s) for the above parameters to work.
2. A value of "0" for the "call hold reminder timer" parameter disables the call hold reminder feature.
3. A value of "0" for the "call hold reminder frequency" parameter prevents additional rings.

Configuring "Call Hold Reminder Timer"

You use the following procedure to configure the "Call Hold Reminder Timer".

Configuration Files

For the specific parameter you can set in the configuration files for setting the "Call Hold Reminder Timer", see Appendix A, the section, "Call Hold Reminder Timer" on page A-159.

Configuring "Call Hold Reminder Frequency"

You use the following procedure to configure the "Call Hold Reminder Frequency".

Configuration Files

For the specific parameter you can set in the configuration files for setting the "Call Hold Reminder Frequency", see Appendix A, the section, "Call Hold Reminder Frequency" on page A-160.

Preferred Line and Preferred Line Timeout

An Administrator or User can define a **preferred line** as well as a **preferred timeout**. If a preferred line is selected, after a call ends (incoming or outgoing), the display switches back to the preferred line. Next time you go off-hook, you pickup on the preferred line. You can specify the number of seconds it takes for the phone to switch back to the preferred line using the "**preferred timeout**" parameter.

An Administrator can configure the "**preferred line**" and the "**preferred timeout**" parameters using the configuration files or the Aastra Web UI. A User can configure these parameters using the Aastra Web UI only.

The following table provides the behavior of the preferred line focus feature with other features on the phone.

Phone Feature	Preferred Line Behavior
call return	The phone switches back to the focused line immediately after the call ends.
speed dial	The line is already specified when the speed dial is created. The phone switches back immediately after the call ends.
conference	For incoming calls, the phone switches back immediately after the call ends.
transfer	For incoming or outgoing calls, the current behavior is that the same line used to transfer the call does not change. For incoming calls, the phone switches back immediately after the call transfers.
blf	The phone switches back immediately after the call ends.
park	The phone switches back immediately after the call ends.
voicemail	The phone switches back immediately after the call ends.
redial	The phone switches back immediately after the call ends.
dialing	For incomplete dialing on a non-preferred line, the focus does not change if some digits are entered. If no digits are entered or digits were cleared, the focus changes to preferred line after the time out has passed without activities.
caller id	If the "Switch UI Focus To Ringing Line" parameter is disabled, the User is able to see the Caller ID when the phone switches the focus to the ringing line.
factory default	Factory default and recovery mode clears the "preferred line" and "preferred line timeout" parameters, and the phone operates in a non-preferred line mode.

Notes:
1. If you specify a value of "**0**" for the **preferred line** parameter, it disables the preferred line focus feature.

2. If you specify a value of "**0**" for the **preferred line timeout** parameter, the phone returns the line to the preferred line immediately.

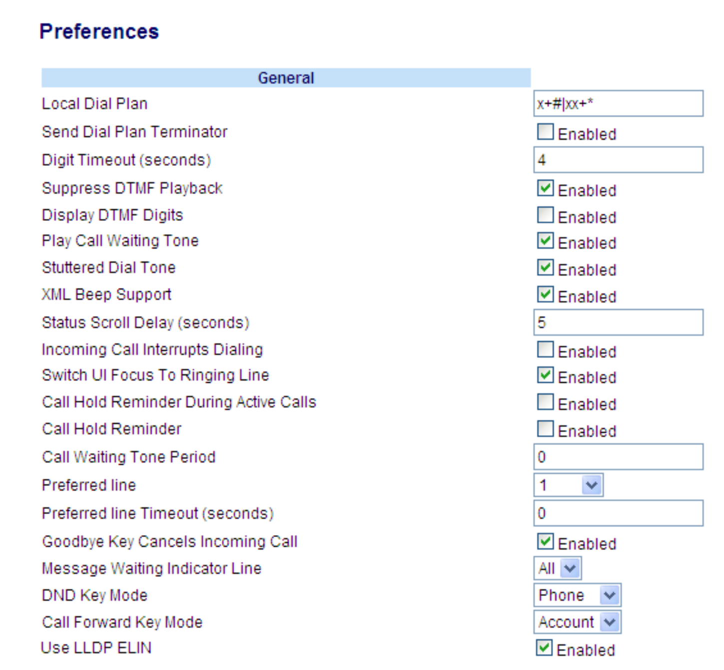
Configuring the Preferred Line and Preferred Line Timeout

You use the following procedures to configure the Preferred Line and the Preferred Line Timeout on the IP Phones.

Configuration Files

For specific parameters you can set in the configuration files for configuring the Preferred Line and Preferred Line Timeout, see Appendix A, the section, "Preferred Line and Preferred Line Timeout" on page A-161.

Use the following parameters to configure preferred line focus using the Aastra Web UI.

	Aastra Web UI
1	Click on **Basic Settings->Preferences->General.** 
2	In the "**Preferred Line**" field, select a preferred line to switch focus to after incoming or outgoing calls end on the phone. Valid values are: • **None** (disables the preferred line focus feature) • **1 to 9** Default is **1**. For example, if you set the preferred line to "**1**", when a call (incoming or outgoing) ends on the phone (on any line), the phone switches focus back to Line 1.
3	In the "**Preferred Line Timeout**" field, enter the amount of time, in seconds, that the phone switches back to the preferred line after a call (incoming or outgoing) ends on the phone, or after a duration of inactivity on an active line. Default is **0**. Valid values are: • **0 to 999**
4	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 8 and 11-line LCD phones:

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 3-line LCD phones:

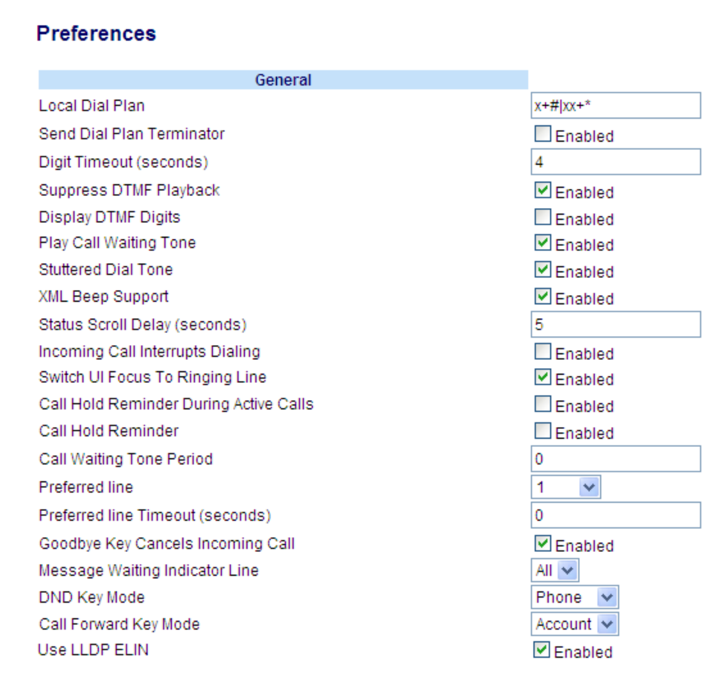
If you enable this parameter, and the phone receives another call when an active call is already present, the "**ignore**" option *only* displays in the LCD window. The phone ignores the incoming call. If you press the **DOWN** arrow key, the phone answers the incoming call.

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-156.

	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  to save your changes.

Configurable Status Code on Ignoring Incoming Calls

When a user presses the "**Ignore**" key on the phones during an incoming call, the phone rejects the incoming call with a status code of "486 Busy Here". The IP phones allow an administrator to configure this status code. You can configure the status code using the configuration files only.

Note: Valid status codes are based on RFC3261.

Use the following parameter to configure a status code when ignoring incoming calls:

- **sip ignore status code**

Configuring Status Codes on Ignoring Incoming Calls

You can use the following procedure to set the status code sent in the response when a user presses the "Ignore" key.

Configuration Files

For the specific parameter you can set in the configuration files, see Appendix A, the section, "Status Code on Ignoring Incoming Calls" on page A-157.

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).

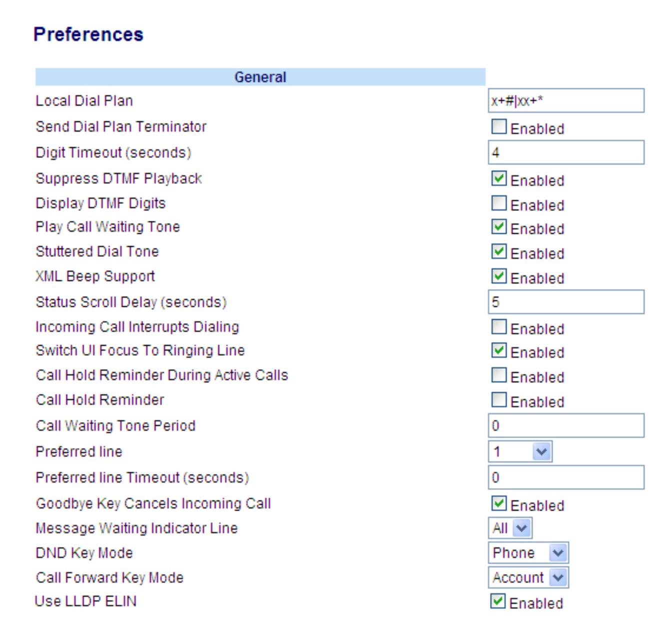
A User can configure the MWI using the Aastra Web UI only. An Administrator can configure the MWI on single or all lines using the configuration files or the Aastra Web UI.

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-163.

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.

Customizable Message Waiting Indicator (MWI) Request URI

In Release 3.1, an Administrator can now enter a parameter in the configuration files to customize the request-URI for MWI feature subscription. This new parameter is called "**sip linex mwi request uri**".

This feature can be set on a per-line basis using the configuration files only.

Note: "**Sip Explicit MWI Subscription**" must be enabled to use this feature. For more information about the Sip Explicit MWI Subscription" parameter, see "Advanced SIP Settings (optional)" on page 4-83.

Configuring Message Waiting Indicator (MWI) Request URI

Use the following procedure to configure an MWI request URI on the IP phone.

Configuration Files
For the specific parameter you can set in the configuration files, see Appendix A, the section, "Message Waiting Indicator Settings" on page A-163.

DND Key Mode

The IP phones have a feature you can enable called "Do not Disturb (DND). An Administrator or User can set “do not disturb” based on the accounts on the phone (all accounts or a specific account). You can set specific modes for the way you want the phone to handle DND. The three modes you can set on the phone for DND are:

- **Account**
- **Phone**
- **Custom**

An Administrator or User can set the DND mode using the Aastra Web UI at the path *Basic Settings->Preferences->General->**DND Key Mode***. An Administrator can also set the DND Key Mode using the configuration files.

Note: You must configure a DND key on the phone to use this feature.

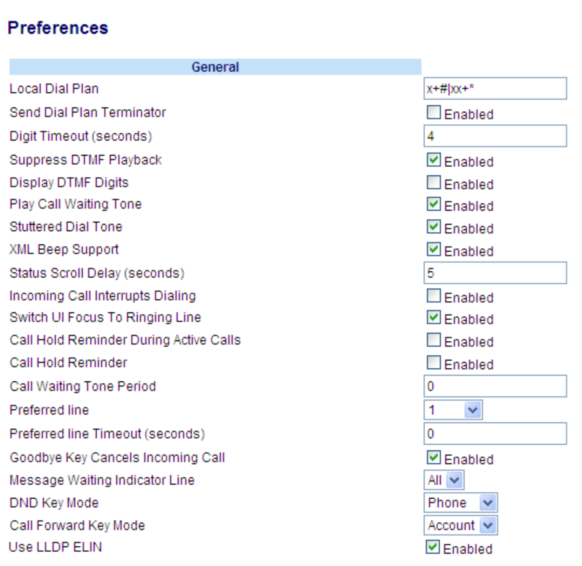
Reference

For more information about how DND works and how you can use it on the phones, see the section, “Do Not Disturb (DND)” on page 5-182.

Configuring the DND Key Mode

Use the following procedures to set the DND Key Mode on the phone.

Configuration Files
For specific parameters you can set in the configuration files, see Appendix A, the section, “DND Key Mode Settings” on page A-165.

	Aastra Web UI
1	Click on **Basic Settings->Preferences->General.** 
2	In the "**DND Key Mode**" field, select a "do not disturb" (DND) mode to use on the phone. Valid values are: Account, Phone, Custom. Default is **Phone**. • **account** Sets DND for a specific account. DND key toggles the account in focus on the IP Phone UI, to ON or OFF. • **phone** Sets DND ON for all accounts on the phone. DND key toggles all accounts on the phone to ON or OFF. • **custom** Sets the phone to display custom screens after pressing the DND key, that list the account(s) on the phone. The user can select a specific account for DND, turn DND ON for all accounts, or turn DND OFF for all accounts. **Notes**: **1.** If there is only one account configured on the phone, then the mode setting is ignored and the phone behaves as if the mode was set to "Phone". **2.** Using the Aastra Web UI, if you change the DND Key Mode to "phone", all accounts synchronize to the current setting of Account 1.
3	Configure a DND key on the phone using the procedures in the section, "Do Not Disturb (DND)" on page 5-182.
4	Click **Save Settings** to save your changes. The change takes effect immediately without a reboot.

Reference

For more information, see the section, "Do Not Disturb (DND)" on page 5-182.

Call Forward Mode

Call Forward (CFWD) 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.

An Administrator or User can configure CFWD on the phone-side by setting a mode for the phone to use (**Account**, **Phone**, or **Custom**). Once the mode is set, you can use the IP Phone UI to use the CFWD feature at *Options->Call Forward* or by pressing a configured Call Forward softkey/programmable key/extension module key.

The following describes the behavior for each CFWD mode.

- **Account mode -** The account mode allows you to configure CFWD on a per account basis. Pressing a configured CFWD key applies to the account in focus.
- **Phone mode -** The Phone mode allows you to set the same CFWD configuration for all accounts (**All**, **Busy**, and/or **No Answer**). When you configure the initial account, the phone applies the configuration to all other accounts. (In the Aastra Web UI, only the account you configured is enabled. All other accounts are grayed out but set to the same configuration.) Using the Aastra Web UI, if you make changes to that initial account, the changes apply to all accounts on the phone.
- **Custom mode -** The Custom mode allows you to configure CFWD for a specific account or all accounts. You can configure a specific mode (**All**, **Busy**, and/or **No Answer**) for each account independently or all accounts. On the 3-line LCD phones, you can set all accounts to **ALL On** or **ALL Off**. On the 8 and 11-Line LCD phones, you can set all accounts to **All On**, **All Off**, or copy the configuration for the account in focus to all other accounts using a **CopytoAll** softkey.

Note: If there is only one account configured on the phone, then the mode setting is ignored and the phone behaves as if the mode was set to “Phone”.

The states you can set for Call Forward are **All**, **Busy**, **No Answer**. You can enable different call forwarding rules/modes independently (for example, you can set different phone numbers for Busy, All, and NoAns modes and then turn them on/off individually. The behavior of these states is dependent on the mode (account, phone, or custom) you configure on the phone.

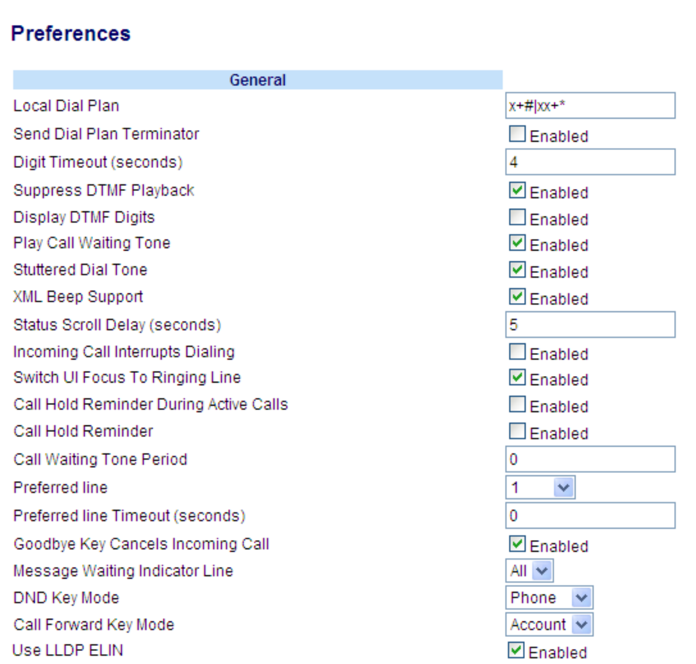
Reference

For more information about how Call Forwarding works and how you can use it on the IP Phones, see “Call Forwarding” on page 5-218.

Configuring Call Forward Key Mode

Use the following procedures to set the Call Forward key mode on the IP phones.

Configuration Files
For specific parameters you can set in the configuration files, see Appendix A, the section, "Call Forward Key Mode Settings" on page A-141.

Aastra Web UI	
1	Click on **Basic Settings->Preferences->General**.  **Note:** If there is no CFWD key configured on the phone or it is removed, you can still configure Call Forwarding via the IP Phone UI at the path *Options->Call Forward.*

Aastra Web UI	
2	In the "**Call Forward Key Mode**" field, select a call forward mode to use on the phone. Valid values are: Account, Phone, Custom. Default is **Account**. • **account** The account mode allows you to configure CFWD on a per account basis. Pressing a configured CFWD key applies to the account in focus. • **phone** The Phone mode allows you to set the same CFWD configuration for all accounts (**All**, **Busy**, and/or **No Answer**). When you configure the initial account, the phone applies the configuration to all other accounts. (In the Aastra Web UI, only the account you configured is enabled. All other accounts are grayed out but set to the same configuration.) Using the Aastra Web UI, if you make changes to that initial account, the changes apply to all accounts on the phone. • **custom** The Custom mode allows you to configure CFWD for a specific account or all accounts. You can configure a specific state (**All**, **Busy**, and/or **No Answer**) for each account independently or all accounts. On the 3-Line LCD phones, you can set all accounts to **ALL On** or **ALL Off**. On the 8 and 11-Line LCD phones, you can set all accounts to **All On**, **All Off**, or copy the configuration for the account in focus to all other accounts using a **CopytoAll** softkey. **Notes**: **1.** If there is no CFWD key configured on the phone or it is removed, you can still set the CFWD modes via the IP Phone UI at the path *Options->Call Forward.* **2.** If there is only one account configured on the phone, then the mode setting is ignored and the phone behaves as if the mode was set to "Phone". **3.** When configuring a CFWD state (**All**, **Busy**, **No Answer**) for an account, you must configure a CFWD number for that state in order for the state to be enabled.
3	Click Save Settings to save your changes. The change takes effect immediately without a reboot.

Reference

For more information, see the section, "Call Forwarding" on page 5-218.

Link Layer Discovery Protocol for Media Endpoint Devices (LLDP-MED) and Emergency Location Identification Number (ELIN)

The IP Phones support Link Layer Discovery Protocol for Media Endpoint Devices (LLDP-MED). LLDP-MED is designed to allow for things such as:

- Auto-discovery of LAN policies (such as VLAN, Layer 2 Priority and Diffserv settings) leading to "plug and play" networking.
- Extended and automated power management of Power over Ethernet endpoints.
- Inventory management, allowing network administrators to track their network devices, and determine their characteristics (manufacturer, software and hardware versions, serial / asset number).

On the IP Phones, LLDP-MED performs the following:

- Supports the MAC/PHY configuration (e.g. speed rate/duplex mode).
- Supports VLAN info from the network policy; this takes precedence over manual settings.
- Allows you to enable/disable LLDP-MED if required.
- Allows you to configure time interval between successive LLDP Data Unit (LLDPDU) frames.
- Allows LLDP packets to be received from the LAN port.
- Allows the phone to use the location information, Explicit Congestion Notification (ECN) Emergency Location Identification Number (ELIN), sent by the switch, as a caller ID for making emergency calls.

➡ **Note:** If the phone receives location information in ECN ELIN format (10 to 25 numeric string), the phone replaces the caller ID SIP header with the ECN ELIN value and the SIP URI does not change. The phone determines if this is an emergency number by checking the emergency dial plan configured on the phone.

The following table identifies the configuration parameters for LLDP and ELIN and which method you can use to configure each parameter. This table also indicates whether the parameters can be configured by an Administrator, a User, or both.

Parameter	Method of Configuration	Who Can Configure
lldp	Configuration Files	Administrator
lldp interval	Configuration Files	Administrator
use lldp elin	Configuration Files	Administrator
LLDP Support	IP Phone UI	Administrator
LLDP	Aastra Web UI	Administrator
LLDP Packet Interval	Aastra Web UI	Administrator
LLDP ELIN	Aastra Web UI	Administrator and User

Configuring LLDP-MED and ELIN

Use the following procedures to configure LLDP-MED and ELIN on the IP phones.

Configuration Files

For specific parameters you can set in the configuration files, see Appendix A, the section, "LLDP-MED and ELIN Settings" on page A-142.

Use the following procedure to enable/disable LLDP-MED using the IP Phone UI.

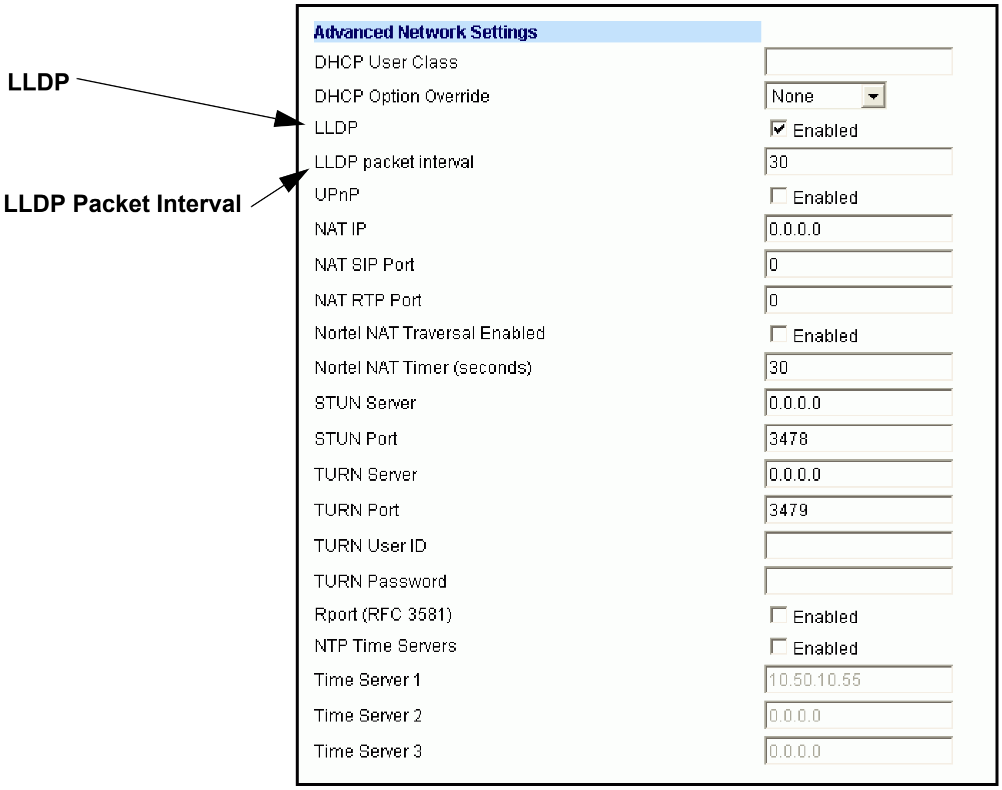
Note: You cannot configure the "LLDP Interval" or the "Use LLDP ELIN" parameters via the IP Phone UI.

Aastra IP Phone UI

Step	Action
1	Press **Options**, and then select **Administrator Menu**.
2	Select **Network Settings**.
3	Select **Ethernet & VLAN**.
4	Select **LLDP Support**.
5	Press **CHANGE** to toggle the LLDP setting to **Enabled** or **Disabled**. This field enables or disables Link Layer Discovery Protocol for Media Endpoint Devices (LLDP-MED) on the IP Phone.
6	Press **DONE** to save the change.

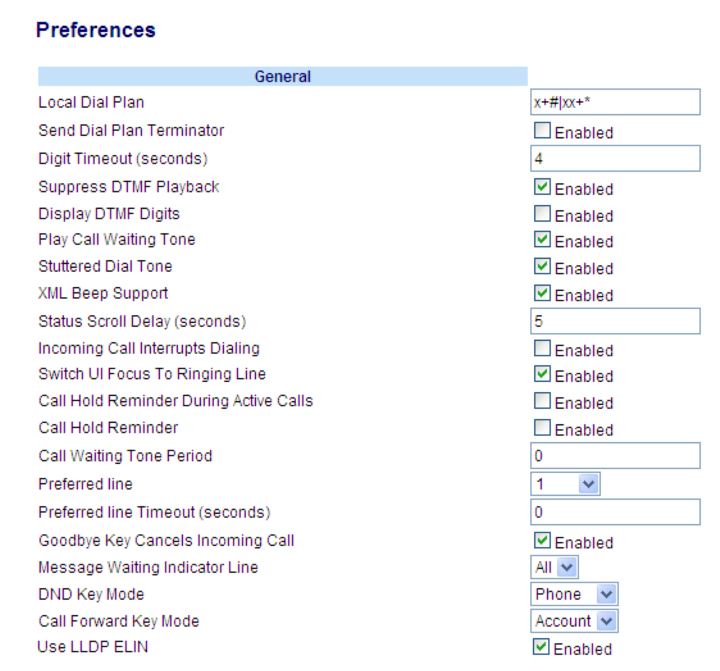
Use the following procedure to configure LLDP-MED using the Aastra Web UI:

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



</td></tr>
<tr><td>2</td><td>The “LLDP” field is enabled by default. To disable LLDP, click the check mark in the box to clear the check mark.</td></tr>
<tr><td>3</td><td>In the “LLDP Packet Interval” field, enter the time, in seconds, between the transmission of LLDP Data Unit (LLDPDU) packets.

The value of zero (0) disables this parameter. Valid values are 0 to 2147483647. Default is 30.</td></tr>
<tr><td>4</td><td>Click Save Settings to save your changes.</td></tr>
</table>

	Aastra Web UI
5	Select **Basic Settings->Preferences->General**. 
6	The "**Use LLDP ELIN**" field is enabled by default. To disable LLDP ELIN, click the check mark in the box to clear the check mark. This field enables or disables the use of an Emergency Location Identification Number (ELIN) received from LLDP as a caller ID for emergency numbers.
7	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 5-103.

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 intercom 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 5-103.

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.

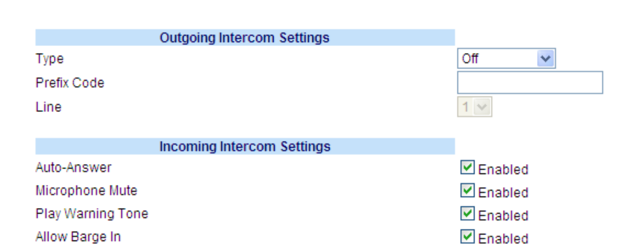
Use the following procedures to configure Intercom calls on the IP phone.

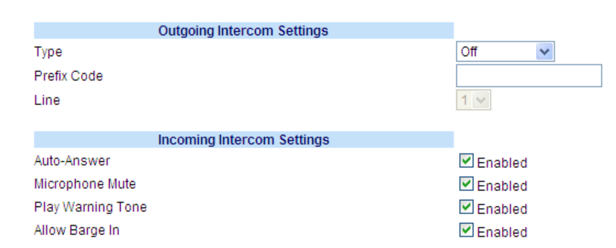
Configuration Files

For specific parameters you can set in the configuration files for outgoing Intercom, see Appendix A, the section, "Outgoing Intercom Settings" on page A-187.

For specific parameters you can set in the configuration files for incoming Intercom, see Appendix A, the section, "Incoming Intercom Settings" on page A-188.

Aastra Web UI

Outgoing intercom settings	
1	Click on **Basic Settings->Preferences->Outgoing Intercom Settings**. 
2	Select an Intercom type for outgoing Intercom calls from the **Type** list box. Valid values are **Phone-Side**, **Server-Side**, **Off**. Default is **Off**.

	Aastra Web UI
3	If Server-Side is selected, enter a prefix to add to the phone number in the "**Prefix Code"** field. **Note:** For Sylantro servers, enter *96.
4	If Phone-Side or Server-Side is selected, select a line from the **Line** list box for which you want the IP phone to use as its configuration on the Intercom call. **Note:** The IP phone uses the configuration from the line you select from this list box. The call itself is made using the first available line at the time of the call.
5	Click [Save Settings] to save your changes.
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.
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.

Group Paging RTP Settings

An Administrator or User can configure a specific key (softkey, programmable key, or expansion module key) on the phone that allows you to send/receive a Real Time Transport Protocol (RTP) stream to/from pre-configured multicast address(es) without involving SIP signaling. This is called Group Paging on the IP phones. You can specify up to 5 listening multicast addresses.

An Administrator can use the following parameters in the configuration files to set Group Paging RTP Settings:

- **paging group listening**
- **softkeyN type, topsoftkeyN type, prgkeyN type, or expmodX keyN type**
- **sofkeyN label**
- **softkeyN value, topsoftkeyN value, prgkeyN value, or expmodX keyN value**

An Administrator or User can use the following parameters in the Aastra Web UI to set Group Paging RTP Settings:

- **Paging Listen Addresses** (Path**:** *Basic Settings->Preferences->Group Paging RTP Settings*)
- **<Paging> Key** (*Operation->Softkeys and XML, Programmable Keys, or Expansion Module Keys*)

Note: The Group Paging RTP Settings are dependant upon the setting for the "**Allow Barge In**" parameter.

How it works

After pressing a configured "Paging" key on the phone, the Phone sends RTP to a preconfigured multicast address(es) (IP port). Any phone in the local network then listens for RTP on the preconfigured multicast address(es) (IP port). For both sending and receiving of the multicast RTP there is no sip signaling involved. The Phone displays the multicast RTP sent/received address(es) to the user.

Note: Multicast RTP is one way only - from sender to the receiver (i.e. from sender to the multicast address(es) (receiver)).

The phone uses a preconfigured G711 uLaw CODEC for multicast RTP.

For Paging Systems, the phone only plays RTP traffic; users have the ability to drop a rogue page. The recipient can drop the incoming page if required. The recipient can also set Do Not Disturb (DND) to ignore any incoming pages.

Note: For outgoing RTP multicasts, all other existing calls on the phone are put on hold.

For incoming RTP multicasts, the ringing display is dependant on the "**Allow Barge-In**" parameter. If this parameter is **disabled**, and there is not other call on the phone, then the paging is automatically played via the preferred audio device (see the model-specific *IP Phone User Guide* for setting Audio Mode on the phone).

If there is an existing call on the phone, the call initially displays in the ringing state. The user has the option to accept/ignore the call. If the "**Allow Barge-In**" parameter is **enabled**, the RTP multicast call barges in, and any existing calls are put on hold.

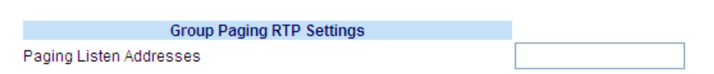
If an RTP multicast session already exists on the phone, and the phone receives another incoming RTP multicast session, the priority is given to the first multicast session and the second multicast session is ignored. The behavior for the incoming calls in this case is also based on the setting for the "**Allow Barge-in**" parameter. The incoming call is handled as if there were an existing call already on the phone.

Configuring Group Paging RTP Settings

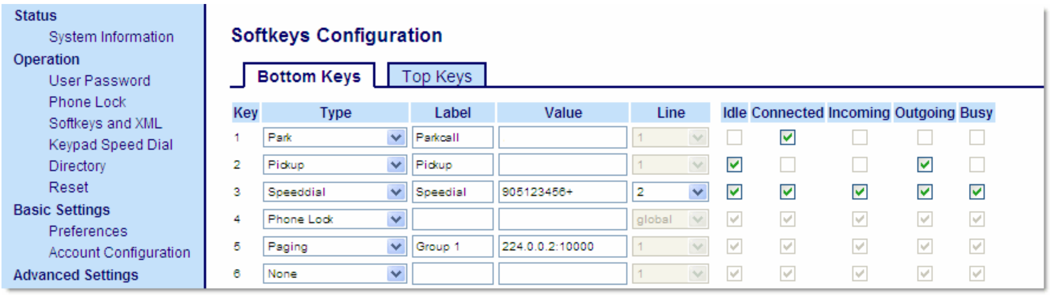
Use the following procedure to configure Group Paging RTP Settings using the configuration files.

Configuration Files
For specific parameters you can set in the configuration files, see Appendix A, the section, "Enable Microphone During Early Media" on page A-190.

Use the following procedure to configure RTP streaming for Paging applications using the Aastra Web UI.

Aastra Web UI	
1	Click on **Basic Settings->Preferences->Group Paging RTP Settings**. 
2	In the "**Paging Listen Addresses**" text box, enter the multicast IP address(es) and port number on which the phone listens for incoming multicast RTP packets. **Note:** Enter the IP address in dotted decimal format (for example, **224.0.0.2:10000,239.0.1.20:15000**) If this field is blank, Paging listening capability is disabled on the phone.

Aastra Web UI

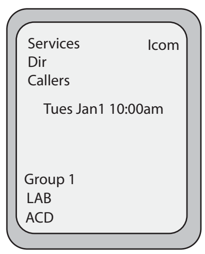
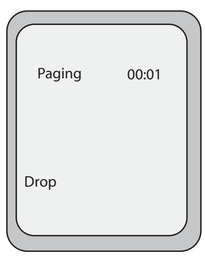
3	Click on **Softkeys and XML.** or Click on **Programmable Keys**. or Click on **Expansion Module Keys**. 
4	Choose a key that you want to assign as the Paging Key and select **Paging** from the "**Type**" field.
5	In the "**Label**" field, enter a key label to assign to the Paging key (for example, "**Group 1**").
6	In the "**Value**" field, enter a multicast IP address(es) and a port number for the Paging key. When you press this key, the phone initiates an outgoing multicast RTP session to the specified address(es) using the specified port. (For example, **224.0.0.2:10000).** **Note:** When you select **Paging** for the "**Type**" field, the "**Line**" field and state fields are disabled.
7	Click  to save your changes.

Using the Configured Paging Key on the IP Phone

The following procedure describes the use of the Paging key. The procedures assumes you have already configured the Paging key using the configuration files or the Aastra Web UI.

Notes:
1. Recipient of a Paging call can set a global "Do Not Disturb" (DND) to ignore any incoming pages.
2. For incoming Paging, the phones use the Intercom configuration settings. The incoming Page is dependant on the "Allow Barge-In" parameter setting and the "*Idling/On Call*" state.

Aastra IP Phone UI

Step	Action
	On the IP Phone, press the key you configured for Paging (labeled as "**Group 1**" in the following illustration).  The phone opens a multicast RTP session and an outgoing OR incoming phone screen displays as follows:  Press the **Drop** key to end the multicast RTP session and return to the idle screen. **Note:** If you enable global DND on the phone, the incoming multicast RTP session is dropped.

Speed DialKey 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 8 and 11-Line LCD phones, 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 "Enabling/Disabling Redial, Xfer, and Conf Keys" on page 5-109.

2. On the 3-Line LCD phones, the Xfer and Conf keys are assigned by default to available keys, respectively. These keys are programmable keys and can be reassigned if applicable. To disable these keys, see "Enabling/Disabling Redial, Xfer, and Conf Keys" on page 5-109.

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 Settings" on page A-199.

Mapping Redial and Conf Keys as Speed Dials

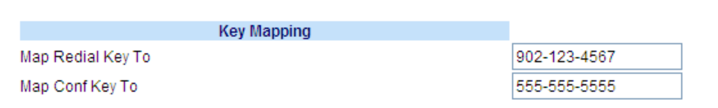
You can map the **Redial** and **Conference** keys on the IP phone to use as speed dial keys. When the **Redial** or **Conference** key is pressed, the number configured for the key automatically speed dials. If no number is configured, the **Redial** and **Conference** keys return to their original 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 speed dialing on the 6757i CT Base Station, the **Redial** and **Conference** keys on the 6757i CT handset retain their original functionality. The **Redial** and **Conference** keys on the handset are not configured for speed dial.

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

Configuration Files
For specific parameters you can set in the configuration files, see Appendix A, the section, "Mapping Key Settings" on page A-199.

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 speed dial when the Redial key is pressed.
3	Enter a number in the "**Map Conf Key To**" field, that the IP phone will use to speed dial when the Conf Key is pressed.
4	Click  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**)**
- **UK**
- **Australia**
- **Europe (generic tones)**
- **France**
- **Germany**
- **Italy**
- **Mexico**
- **Brazil**
- **United Kingdom (UK)**
- **Russia**
- **Malaysia**

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

IMPORTANT: 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-154 or "Ring Tone Per-Line Settings" on page A-155.

IP Phone UI

Step	Action
For global configuration only:	
1	Press 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, Brazil, Russian, Malaysia, 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.
For the 6739i:	
1	Press on the phone to enter the Options List.
2	Press the <**Audio>** button.
3	Press the **<Ring Tone>** button. The following values display: • Tone 1 (Default) • Tone 2 • Tone 3 • Tone 4 • Tone 5 • Silent
4	Press the value you want to set for the Ring Tone.
5	Press the **<Tone Set>** button.

IP Phone UI

Step	Action
6	Select the country for which you want to apply the Tone Set. Valid values are **Australia**, **Europe, France**, **Germany**, **Italy**, **Mexico, Brazil, Russian, Malaysia, UK**, and **US**. Default is **US**.
7	Press the to return to the previous menu or press the to return to the idle screen.

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, Brazil, Russia, Malaysia, 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 5-112 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 5-112 for valid values.
6	Click Save Settings 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, Broadworks, and Sylantro servers using the configuration files and the Aastra Web UI. Configuration of priority alerting is on a global-basis only.

How Priority Alerting Works

When the IP phone detects an incoming call, the phone firmware inspects the INVITE request in the IP packet for an "Alert-Info" header.

If it contains an "Alert-Info" header, the firmware strips out the URL and keyword parameter and maps it to the appropriate Bellcore tone.

If there is no keyword parameter in the "Alert-Info" header, or the INVITE message contains no "Alert-Info" header, then the IP phone firmware uses the Bellcore standard ring tone.

Asterisk/Broadworks Servers

The ring tone keywords that can display in the "Alert-Info" header for an Asterisk and Broadworks server are:

Asterisk/Broadworks 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.

Example

The following are examples of the the Asterisk/Broadworks Server ring tone keywords:

```
Alert-Info: <http://127.0.0.1/Bellcore-dr2>
or
Alert-Info: <Bellcore-dr2>
```

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

A System Administrator can configure the ring tone cadences if required, using the configuration files. 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.

Mexican Tone Set Cadences

The following are Mexican tone set cadences.

Tone	Frequency (Hz)	Cadence (on/off)
Dial	425	Continuous
Secondary Dial	425	300/100/300/1300
Ringing	425	1000/4000
Busy	425	500/500
Congestion	425	250/250
Call Waiting	425	100/100/100/10000
Ring Cadence		1000/4000

Brazilian Tone Set Cadences

The following are Brazilian tone set cadences.

Tone	Frequency (Hz)	Cadence (on/off)
Dial	425	Continuous
Secondary Dial	425	300/100/300/1300
Ringing	425	1000/4000
Busy	425	250/250
Congestion	425	500/500
Call Waiting	425	100/100/100/10000
Ring Cadence		1000/4000

Russian Tone Set Cadences

The following are Russian tone set cadences.

Tone	Frequency (Hz, dBm0)	Cadence (ms) tone pause tone	Note
Dial	425, -10		Continuous
Special Dial	425, -10	500/50	Repetitive
Busy	425, -10	500/500	Repetitive
Ringing	425, -10	1000/4000	Repetitive
Congestion	425, -10	200/200	Repetitive
Call Waiting	425, -10	200/600/200	Non-repetitive

Malaysian Tone Set Cadences

The following are Malaysian tone set cadences.

Tone	Frequency (Hz)	Cadence (on/off)
Dial	425	Continuous
Secondary Dial	425	160/160
Ringing	425 * 50	400/200/400/2000
Busy	425	500/500
Congestion	425	250/250
Call Waiting	425	100/200/100/8600
Ring Cadence	-	400/200/400/2000

Limitations for Malaysian Cadences

- The phone does not apply a different volume level to the first part of the call waiting tone.
- The level of the 50Hz modulation signal for ring back is 90%.

Configuring Priority Alerting and Ring Tone Cadences using the Configuration Files

Use the following procedures to configure priority alerting and ring tone cadences on the IP phones.

Configuration Files

For specific parameters you can set in the configuration files, see Appendix A, the sections,

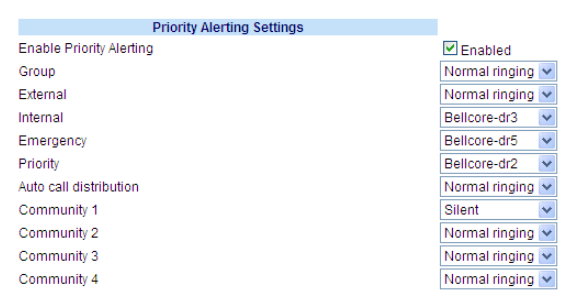
- "Priority Alert Settings" on page A-166.
- "Bellcore Cadence Settings" on page A-172.

Note: You can configure Bellcore cadences using the configuration files only.

Configuring Priority Alerting using the Aastra Web UI.

Use the following procedure to configure Priority Alerting using the Aastra Web UI.

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.

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 -117.

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 -117.

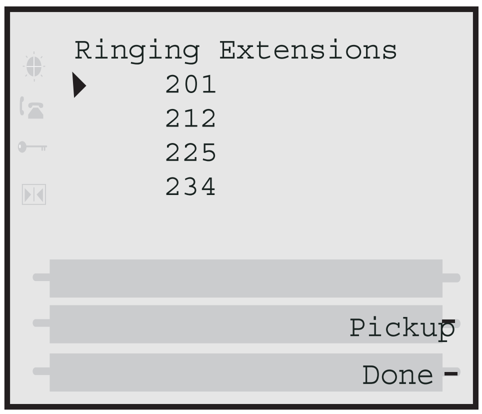
Reference

For more information about enable/disabling call waiting on the IP Phone, see the section, "Call Waiting" on page 5-66.

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 -150
- 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.:



Reference

For more information about using the **AastraIPPhoneTextMenu** object, contact Aastra Customer Support regarding the "Aastra XML Development Guide."

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 speed dial 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.

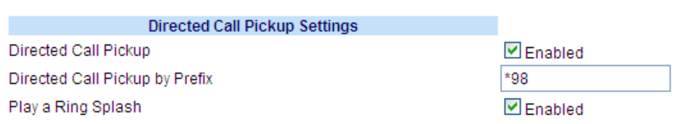
Enabling/Disabling Directed Call Pickup

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-196.

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.
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.

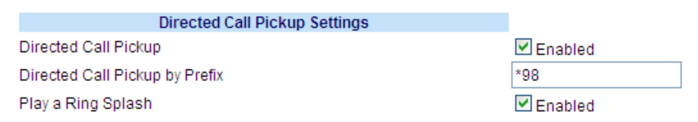
> **Note:** You must enable Directed Call Pickup before performing these procedures. See "Enabling/Disabling Directed Call Pickup" on page 5-124.

Configuration Files

To set BLF or BLF\List in the configuration files for Directed Call Pickup, see Appendix A, the sections:

- "Softkey/Programmable Key/Feature Key/Expansion Module Key Parameters" on page A-202.
- "BLF List URI Settings" on page A-235.

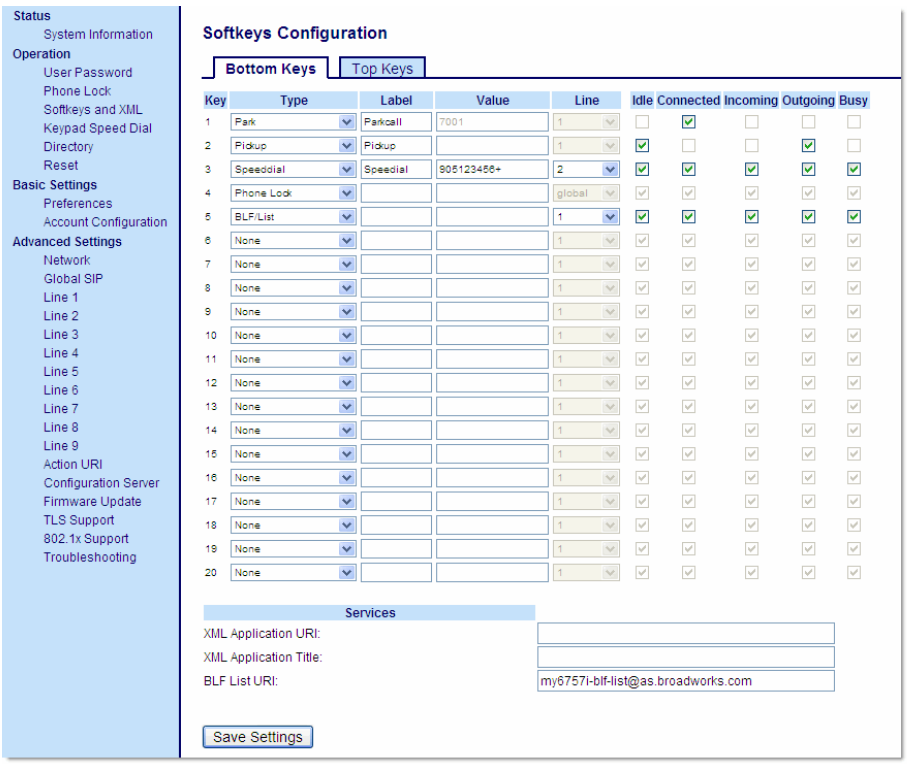
Use the following procedure to configure BLF or BLF/List for Directed Call Pickup in the Aastra Web UI.

	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

<table>
<tr><td>5</td><td>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 6757i Key Screen is shown as an example.



</td></tr>
<tr><td>6</td><td>Select a softkey or programmable key to configure.</td></tr>
<tr><td>7</td><td>In the "Type" field, select "BLF" (Asterisk), "BLF\List" (BroadSoft BroadWorks).</td></tr>
<tr><td>8</td><td>For 8 and 11-Line LCD 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.</td></tr>
<tr><td>9</td><td>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.</td></tr>
<tr><td>10</td><td>Click 

to save your changes.</td></tr>
<tr><td>11</td><td>In the "Line" field, select a line number that is actively registered to the appropriate SIP proxy you are using.</td></tr>
</table>

	Aastra Web UI
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, my6757i-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 6757i 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.

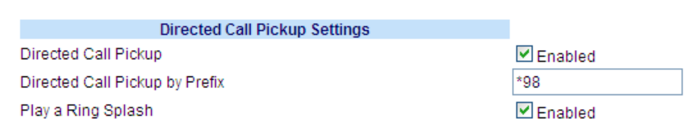
Notes:

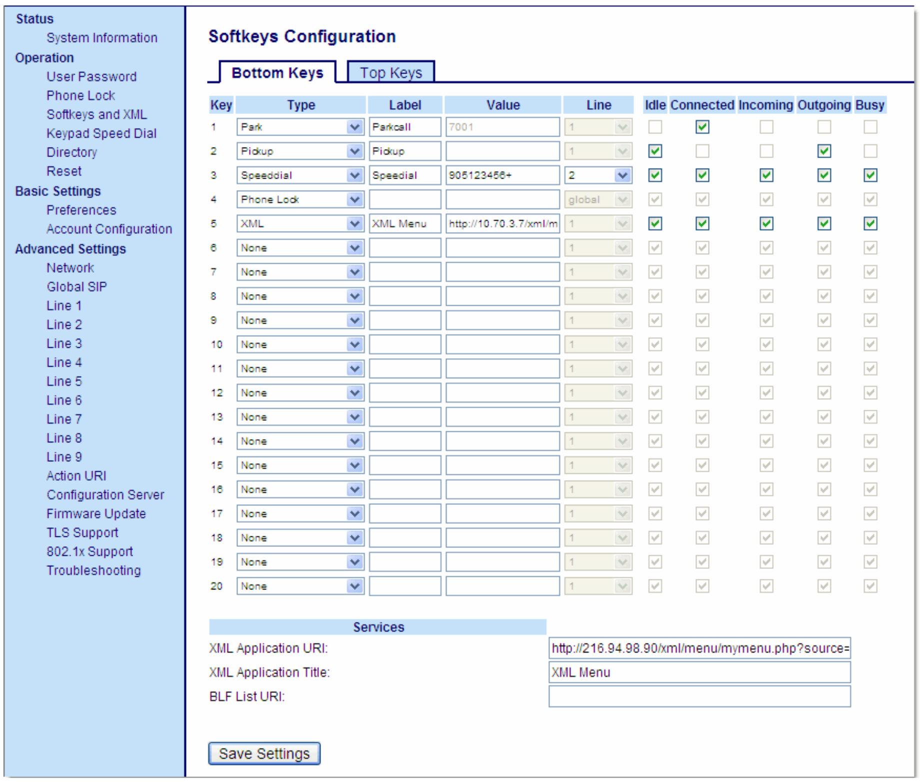
1. Before implementing this procedure, you must create an XML application that the phone uses when the XML softkey or programmable key is pressed. This XML application must be entered as a URI in the "Value" field of the XML key. For information about creating an XML script, see the *Aastra XML Developer's Guide*.

2. You must enable Directed Call Pickup before performing these procedures. See "Enabling/Disabling Directed Call Pickup" on page 5-124.

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-202.

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 6757i 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 8 and 11-Line LCD softkeys**: In the "**Label**" field, enter the name of the person who’s extension you are monitoring.
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 the *Aastra XML Developer’s Guide*.
10	Select the line state (idle, connected, incoming, outgoing, busy) that you want to apply to the XML softkey or programmable key.
11	Click  to save your changes.

Softkeys/Programmable Keys/Feature Keys/Expansion Module Keys

You can configure the softkeys, programmable keys, feature keys, and expansion module keys that are applicable to a specific phone model, 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 (8 and 11-Line LCD phones)

The 9480i, 9480i CT, and 6755i IP phone have 6 softkeys you can configure to perform specific functions, The 6757i and 6757i CT IP phones have 12 softkeys you can configure. With up to 3 expansion modules attached to a 67xi model phone, you can get an additional 108 softkeys (M670i) or an additional 180 softkeys (M675i) to configure. 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
9480i	6	Not Applicable	-	9	-
9480i CT	6	Not Applicable	-	9	15
6739i	55	36 to 108* (Model M670i) 60 to 180** (Model M675i)	-	9	-
6755i	6	36 to 108* (Model M670i) 60 to 180** (Model M675i)	6	9	-
6757i	12	36 to 108* (Model M670i) 60 to 180** (Model M675i)	-	9	-
6757i CT	12	36 to 108* on Base Station (Model M670i) 60 to 180** on Base Station (Model M675i)	-	9	15

*The M670i expansion module consists of 36 softkeys. You can have up to 3 expansion modules on an IP phone totaling 108 softkeys. Valid for 6739i, 6753i, 6755i, 6757i, and 6757i CT phones.

**The M675i expansion module consists of 60 softkeys. You can have up to 3 expansion modules on an IP phone totaling 180 softkeys. Valid for 6739i, 6755i, 6757i, and 6757i CT phones.

State-Based Softkeys (8 and 11-Line LCD phones 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
Speed Dial	idle, connected, incoming, outgoing, busy
DND	idle, connected, incoming, outgoing, busy
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
Last Call Return	idle, connected, incoming, outgoing, busy
Call Forward	idle, connected, incoming, outgoing, busy
BLF/Xfer	idle, connected, incoming, outgoing, busy
Speed Dial/Xfer	idle, connected, incoming, outgoing, busy
Directory	idle, connected, incoming, outgoing, busy
Callers List	idle, connected, incoming, outgoing, busy
Incom (Intercom)	idle, connected, incoming, outgoing, busy
Services	idle, connected, incoming, outgoing, busy
Phone Lock	All states disabled.
Paging	All states disabled.
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: speed dial
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 8 and 11-Line LCD phones” on page A-203.

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: speed dial
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	Speed Dial displays for softkey6. Connected displays as applicable.

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

Programmable Keys (9143i, 6730i, 6731i, 6753i, 6755i)

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
9143i	-	Not Applicable	7	9	-
6730i	-	Not Applicable	8	6	-
6731i	-	Not Applicable	8	6	-
6753i	-	36 to 108* (Model M670i)	6**	9	-
6755i	6	36 to 108* (Model M670i) 60 to 180** (Model M675i)	6	9	-

*The M670i expansion module consists of 36 softkeys. You can have up to 3 expansion modules on an IP phone totaling 108 softkeys.

**On the 6753i, two of the 6 programmable keys are the DELETE and SAVE keys and can be programmed only if Administrator allows.

Softkey/Programmable Key/Expansion Module Key Functions

You can configure the softkeys and programmable keys on the phones and any attached expansion module keys to perform specific functions using the configuration files or the Aastra Web UI.

Note: On the 6739i, you can also configure softkeys using the IP Phone UI. *For more information, see the Aastra Model 6739i IP Phone User Guide.*

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.

Note: These functions apply to all model phones unless specifically indicated that the function does not apply.

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.
Speed Dial	speed dial	Speed Dial	Indicates the key is configured for speed dial use. You can configure a softkey to speed dial a specific number. Optionally, you can also configure a speed dial 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 speed dial prefixes, see "Speed Dial Prefixes" on page 5-148. You can also create speed dial keys and edit the keys using the IP Phone keypad. For more information about speed dial keys and editing speed dial 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 5-150. For more information about BLF Subscription Period, see "BLF Subscription Period" on page 5-155.
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 5-150.
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 5-165. For more information about ACD subscription period, see "ACD Subscription Period" on page 5-173.

Softkey/ Programmable Key Function	Configuration File Parameter	Aastra Web UI Parameter	Description
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 5-177.
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 5-182.
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 5-284.
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 6757i 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	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	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 Softkey" on page 5-207.
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 Softkey" on page 5-207.
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) (Sylantro Servers)" on page 5-214.

Softkey/ Programmable Key Function	Configuration File Parameter	Aastra Web UI Parameter	Description
Call Forward	callforward	Call Forward	Indicates the key is set to be used to access the Call Forward menus on the phone. For more information about call forwarding, see the section "Call Forwarding" on page 5-218.
BLF/Xfer	blf/xfer	Call Forward	Indicates the key is set to be used as a BLF key AND as a Transfer key. For more information about the BLF/Xfer feature, see the section "BLF/Xfer and Speed Dial/Xfer Keys" on page 5-157.
Speed Dial/Xfer	speed dial/xfer	Speed Dial/Xfer	Indicates the key is set to be used as a speed dial key AND as a Transfer key. For more information about the Speed Dial/Xfer feature, see the section "BLF/Xfer and Speed Dial/Xfer Keys" on page 5-157.
Speed Dial/Conf	speed dial/conf	Speed Dial/Conf	Indicates the key is set to be used as a speed dial key AND as a conference key. For more information about the Speed Dial/Conf feature, see the section "Speed Dial/Conference Key" on page 5-162.
Callers List	callers	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 5-262.
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 5-270.
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 5-101.
Conference (Applicable to the 6753i only)	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 5-327.

Softkey/ Programmable Key Function	Configuration File Parameter	Aastra Web UI Parameter	Description
Transfer (Applicable to the 6753i only)	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.*
Services	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 cordless handsets on CT models)	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 5-55.
Paging	paging	Paging	Indicates the softkey is set for Group Paging on the phone. Pressing this key automatically sends a Real Time Transport Protocol (RTP) stream to pre-configured multicast address(es) without involving SIP signaling. For more information about the Paging key, see "Group Paging RTP Settings" on page 5-105.
Empty (Not applicable to programmable keys or expansion module keys)	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

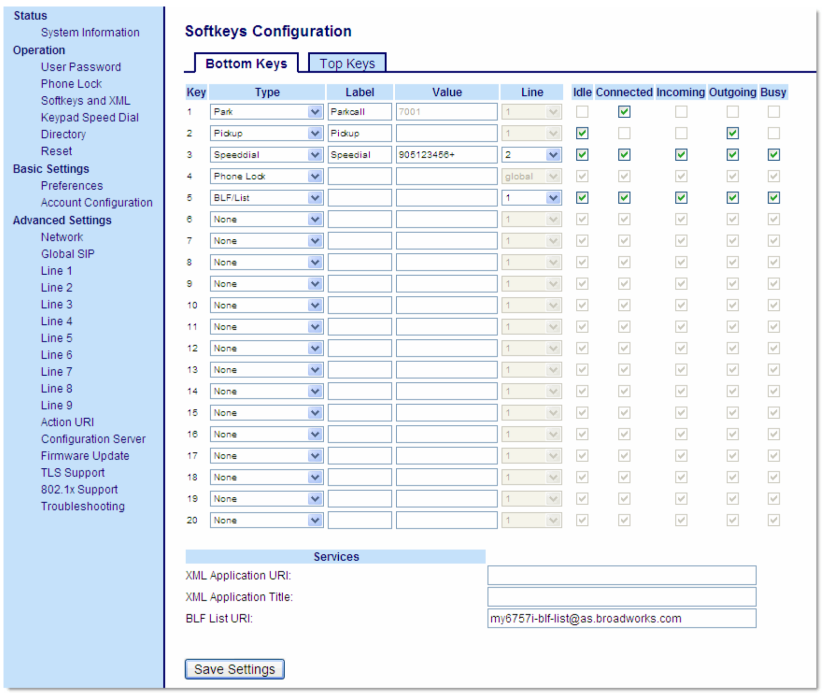
Use the following procedures to configure the softkeys and programmable keys on the IP phone.

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-202.

<table>
<tr><th colspan="2">Aastra Web UI</th></tr>
<tr><td>1</td><td>

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 6757i Key Screen is shown as an example.



</td></tr>
<tr><td>2</td><td>Select a key to configure.</td></tr>
<tr><td colspan="2">For Softkeys and Expansion Module Keys:</td></tr>
<tr><td>3</td><td>

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-202.

</td></tr>
<tr><td>4</td><td>If applicable, enter a label in the "Label" field.</td></tr>
</table>

	Aastra Web UI
5	If applicable, in the "**Value**" field, enter a value to associate with the softkey. For example, for a speed dial value, you can enter a number you want to use for the speed dial key, or 12345+ as a speed dial 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-202.
10	In the "**Value**" field, enter a value to associate with the programmable key. For example, for a speed dial value, you can enter a number you want to use for the speed dial key, or 12345+ as a speed dial 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.

6757i Cordless (CT) Feature Keys

In addition to the softkeys on the 6757i 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 6757i 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 v key from the idle state to go off-hook. Then, press the y 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 6757i 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 6757i CT handset.

The following **Handset Key Functions Table** lists the available functions for the keys on the 6757i 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 6757i CT User's Guide.* For information about other Intercom features, see "Incoming/Outgoing Intercom with Auto-Answer and Barge In" on page 5-101.
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 -270.
Callers	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 -262.
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 6757i 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 Softkey" on page 5-207.
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 Softkey" on page 5-207.

Feature Key Function	Configuration File Parameter	Aastra Web UI Parameter	Description
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 6757i CT User's Guide.* For information about enabling centralizing conferencing, see "Centralized Conferencing (for Sylantro and Broadsoft Servers)" on page 5-327.
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 6757i CT accepts a maximum of 50 entries with the public attribute. For more information about the public/private keys, see your *Aastra IP Phone 6757i 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 6757i 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 6757i CT User's Guide.*

Reference

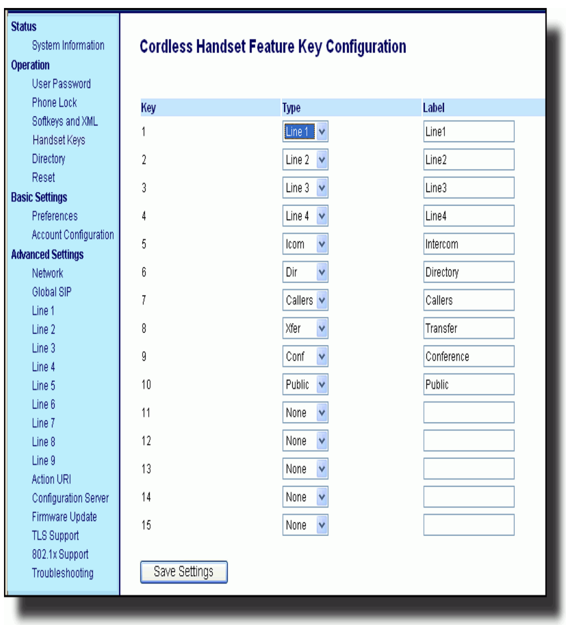
For more information about features key functions for your 6757i CT, see your *Aastra IP Phone 6757i CT User's Guide*.

Configuring Handset Feature Keys

You can program up to 15 feature keys on the 6757i CT IP phone using the configuration files or the Aastra Web UI. Use the following procedure to program the feature keys on your 6757i 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 (9480i CT and 6757i CT)" on page A-221.

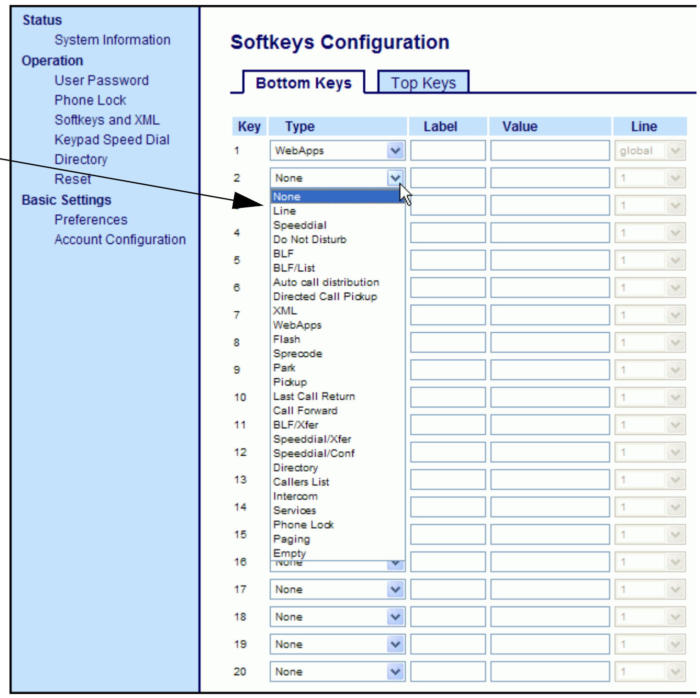
	Aastra Web UI
1	Click on **Operation->Handset Keys.** 
2	Select the handset key you want to program.
3	Select the function for that handset key from the "**Key Function**" field.
4	Click  to save the function you selected to the handset key. The key programming information is sent to the 6757i Base Station and to all the cordless handsets associated with that Base Station. Any key programmed to "None" does not appear in the handset's list.

Customizing the Key Type List in the Aastra Web UI

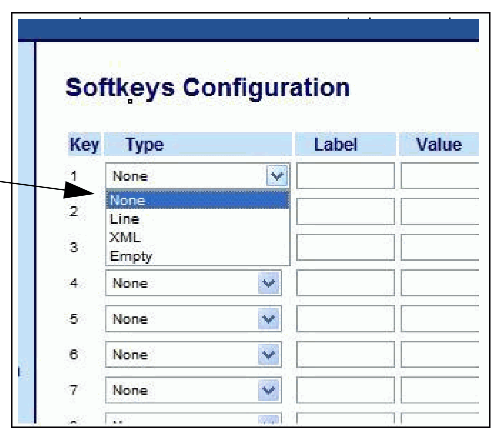
An Administrator can configure which key types display in the Aastra Web UI list for a Softkey, Programmable Key, Expansion Module Key, and/or Feature Key (CT Models). Currently, in the Aastra Web UI for a phone, you can select a type of key from a list of approximately 26 key types to assign to a softkey, programmable key, expansion module key, and/or feature key.

Using the configuration files, you can specify key types to display in the key type list that apply to a User's environment.

List of Available Key Types



Example of a
List of Configured Keys Types
(None, Line, XML, Empty)



In addition to being able to specify which key types display in the list, the Administrator can also determine in which order the key types display.

You can use the following configuration file parameters to control which key types to display and specify in which order to display them in:

- **softkey selection list**
- **feature key selection list**

If no value is specified for the “softkey selection list” and/or “feature key selection list” parameters, the key “Type” list displays all of the key types by default.

If an Administrator configures specific key types for a phone in the configuration file, and the phone for which he downloads the configuration to already has key types configured on it, those key types display in the key list for those keys, in addition to the key types specified by the Administrator. For example, a phone has a Park key and a Pickup key already configured on the phone, and the Administrator downloads a configuration file to the phone that has specific key types of None, Line, Speed Dial, and XML. After the configuration file is downloaded, the Park key list will show None, Line, Speed Dial, XML, and Park; the Pickup key list will show None, Line, Speed Dial, XML, and Pickup; all other keys that were configured as None before the download will show only None, Line, Speed Dial, and XML.

Notes:
1. Any key types configured that do not apply to the phone are ignored.
2. The SAVE and DELETE keys appear by default as Keys 1 and 2 on the 6753i and 9143i, and as Keys 5 and 6 on the 6730i and 6731i, unless your Administrator configured these keys as other functions.
3. An Administrator must use the English value when configuring the key types in the configuration files.
5. After configuring specific key types for a phone, the key types in the Aastra Web UI display the same for both the User and Administrator Web interfaces for that phone.

Customizing the Key Type List Using the Configuration Files

Use the following procedure to configure the Key Type List that displays in the Aastra Web UI.

Configuration Files

For specific parameters you can set in the configuration files, see Appendix A, the section, “Customizing the Key Type List” on page A-229.

Speed Dial Prefixes

The normal function of the **speed dial** 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 speed dial configuration in the configuration files:

```
softkey1 type: speed dial
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 speed dial option allows you to configure a preset string of numbers followed by a "+". This feature allows the phone to speed dial 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 speed dial configuration in the configuration files:

```
softkey2 type: speed dial
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 speed dial prefix using the configuration files or the Aastra Web UI.

Enabling/Disabling Ability to Add or Edit a Speed Dial Key

The IP Phones allow you to set a parameter, "**speed dial edit**" using the configuration files that allows you to enable or disable the ability to add a speed dial key or edit a speed dial key from the IP Phone UI. Disabling this parameter prevents a user from adding or editing a speed dial key.

The default for this parameter is enabled, allowing you to create and edit speed dial keys on the phone using the Press-and-hold feature, softkeys, programmable keys, expansion module keys and key pad, speed dial menu in the IP Phone UI, and the SAVE TO key.

If this parameter is set to disabled, it blocks the user from using any of the features on the phone to create or edit a speed dial key.

Enabling/Disabling the Ability to Add or Edit a Speed Dial Key Using the Configuration Files

Use the following procedure to enable/disable the ability to add and edit a speed dial key.

Configuration Files
For specific parameters you can set in the configuration files, see Appendix A, the section, "Enabling/Disabling Ability to Add/Edit Speeddial Keys" on page A-235.

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 8 and 11-Line LCD phones, 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 3-Line LCD phones, 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 M670i and M675i Expansion Modules.

You can configure a BLF key on the IP Phones using the configuration files or the Aastra Web UI.

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 3-Line LCD phones, 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.

Note: 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.

You can configure a BLF/List key on the IP Phones using the configuration files or the Aastra Web UI. You can also specify a BLF list URI that the phone uses to access the required BLF list. You can specify a BLF List URI using the "**list uri**" parameter in the configuration files or the BLF List URI field in the Aastra Web UI at the path *Operation->Softkeys/Programmable Keys/Expansion Module Keys->Services->BLF List URI*. For more information about the "list uri" parameter, see Appendix A, the section, "BLF List URI Settings" on page A-235.

Note: On the 6739i, you can configure a BLF/List softkey using the IP Phone UI also. *For more information, see the Aastra Model 6739i IP Phone User Guide.*

Example

A receptionist has a 6757i 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 6757i screen next to a softkey button. The softkey icons on the screen change depending on the state of the extensions.

On 3-Line LCD phones 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

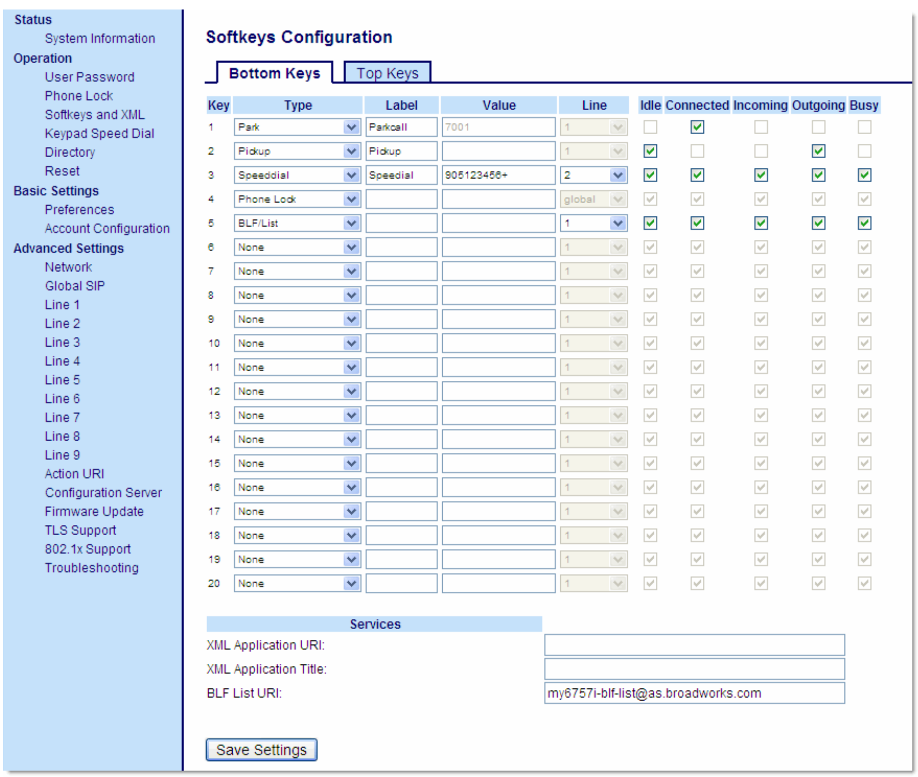
For sample BLF configurations, see Appendix D, “Sample BLF Softkey Settings.”

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 sections, • "Softkey/Programmable Key/Feature Key/Expansion Module Key Parameters" on page A-202. • "BLF List URI Settings" on page A-235.

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 6757i 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 8 and 11-Line LCD 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, my6757i-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 6757i 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).

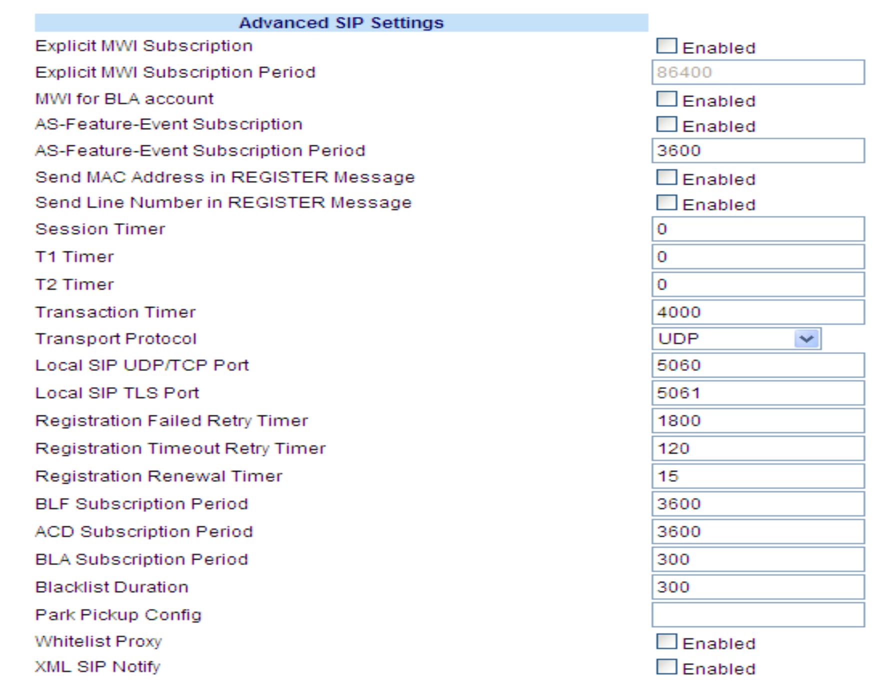
Setting this parameter to a value lower than 3600 allows the configured BLF feature to become active more quickly after a software/firmware upgrade or after a reboot of the IP phone. If you enter a value lower than 120 for this parameter, the default value (3600) will be used by the IP phone.

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-103.

	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  to save your changes.

BLF/Xfer and Speed Dial/Xfer Keys

The IP Phones have a transfer (Xfer) enhancement feature you can use with the BLF and Speed Dial keys - **BLF/Xfer** and **Speed Dial/Xfer**.

The BLF key allows one or more extensions to be monitored, and once there is any state change with those extensions, the key shows the status of the monitored lines. The Xfer key allows a call to be transferred to other recipients blindly or consultatively.

The Speed Dial key allows a number to be dialed quickly by pressing one key configured for speed dialing. After answering a call, the recipient can transfer the call to an extension by:

1. Pressing Xfer key
2. Entering the number of the extension or pressing speed dial or BLF key.
3. Pressing Xfer key again

The BLF and Speed Dial transfer enhancement feature provides a simpler way of transferring calls using the keys called **BLF/Xfer** and **Speed Dial/Xfer**. The **BLF/Xfer** key combines the BLF and Xfer key's functionality together allowing the user to transfer calls or use BLF with one key. Similarly, the **Speed Dial/Xfer** key combines the Speed Dial key and Xfer key's functionality together allowing the user to press one key to speed dial and transfer calls.

Note: It is recommended that you enable the "**switch focus to ringing**" parameter when using the BLF and Speed Dial transfer key feature. For more information about this parameter, see "Switch Focus to Ringing Line" on page 5-77.

BLF/Xfer Key Requirements and Functionality

- **BLF/Xfer and BLF**

  A BLF/Xfer key can be configured for subscribing to an extension and monitor the status of the extension, similar to the BLF key functionality. Changes of the state of the monitored extension are indicated by a LED / Icon.

- **BLF/Xfer and Blind Transfer Calls**

  When the focused line is in the "**Connected**" state, pressing the BLF/Xfer key transfers the call to the extension unconditionally, disregarding the status of the monitored extension.

  If transferring a call to an extension fails, a message "*Transfer Failed*" displays on the phone, and you can reconnect the call (get the call back) by pressing the line key again.

- **BLF/Xfer and Call forward**

  When the focused line is in the "**Ringing**" state, pressing the BLF/Xfer key forwards the call to the extension unconditionally, disregarding the status of the monitored extension.

- **BLF/Xfer and Speed Dial**

  When the focused line and the monitored extension are idle, pressing the BLF/Xfer key causes the phone to go offhook and dial the number of the extension.

Speed Dial/Xfer Key Requirements and Functionality

The Speed Dial/Xfer key has the following capabilities:

- **Speed Dial/Xfer and Speed Dial**

  When the phone is in the "**Idle**" state, pressing the Speed Dial/Xfer key causes the phone to go offhook and dial the predefined extension.

- **Speed Dial/Xfer and Blind Transfer**

  When the phone is connected to a call, pressing the Speed Dial/Xfer key blind transfers the call to the predefined target.

  If transferring a call fails, a message "*Transfer Failed*" displays, and you can reconnect the call (get the call back) by pressing the line key again.

- **Speed Dial/Xfer and Call Forward**

  When the phone is in the "**Ringing**" state, pressing the Speed Dial/Xfer key forwards the call to the predefined extension.

Note: On the 6739i, you can configure a BLF./Xfer and Speed Dial/Xfer softkeys using the IP Phone UI also. *For more information, see the Aastra Model 6739i IP Phone User Guide.*

Configuring the BLF/Xfer Key and the Speed Dial/Xfer Key Using the Configuration Files

You use the following parameters in the configuration files to configure the BLF/Xfer key and/or Speed Dial/Xfer key on the IP Phone.

Softkey Parameters	**Programmable Key Parameters**	**Expansion Module Parameters**
softkeyN type softkeyN label softkeyN value softkeyN line softkeyN states	prgkeyN type prgkeyN value prgkeyN line	expmodN keyN type expmodN keyN label expmodN keyN value expmodN keyN line
Top Softkey Parameters		
topsoftkeyN type topsoftkeyN label topsoftkeyN value topsoftkeyN line		

Examples:

```
softkey1 type: speed dialxfer
softkey1 label: BX7801
softkey1 value: 7801
softkey1 line: 1
softkey1 states: idle connected incoming outgoing busy

prgkey1 type: blfxfer
prgkey1 value: 35
prgkey1 line: 1
```

Refer to the following in Appendix A to configure a BLF/Xfer and Speed Dial/Xfer key on the IP phone using the configuration files.

Configuration Files

To set a BLF/Xfer and Speed Dial/Xfer key using the configuration files, see Appendix A, "Softkey/Programmable Key/Feature Key/Expansion Module Key Parameters" on page A-202.

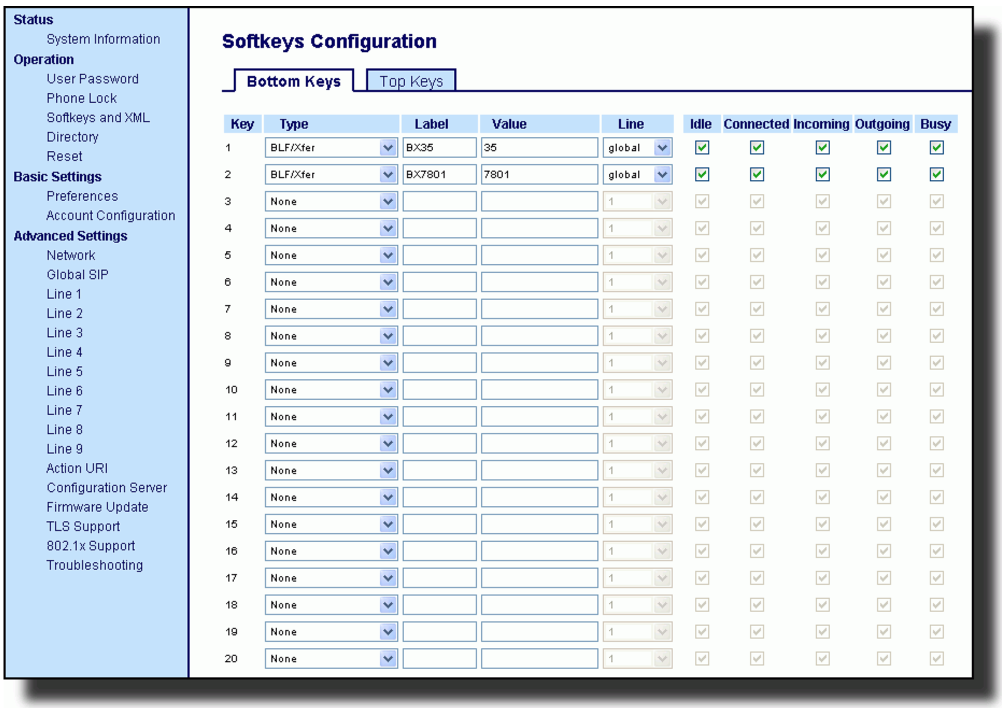
Configuring the BLF/Xfer Key and the Speed Dial/Xfer Key Using the Aastra Web UI

You configure the BLF/Xfer key and/or the Speed Dial/Xfer Key on the IP phone similar to configuring a BLF key or speed dial key using the Aastra Web UI. Use the following procedure to configure BLF/Xfer and/or Speed Dial/Xfer.

Aastra Web UI

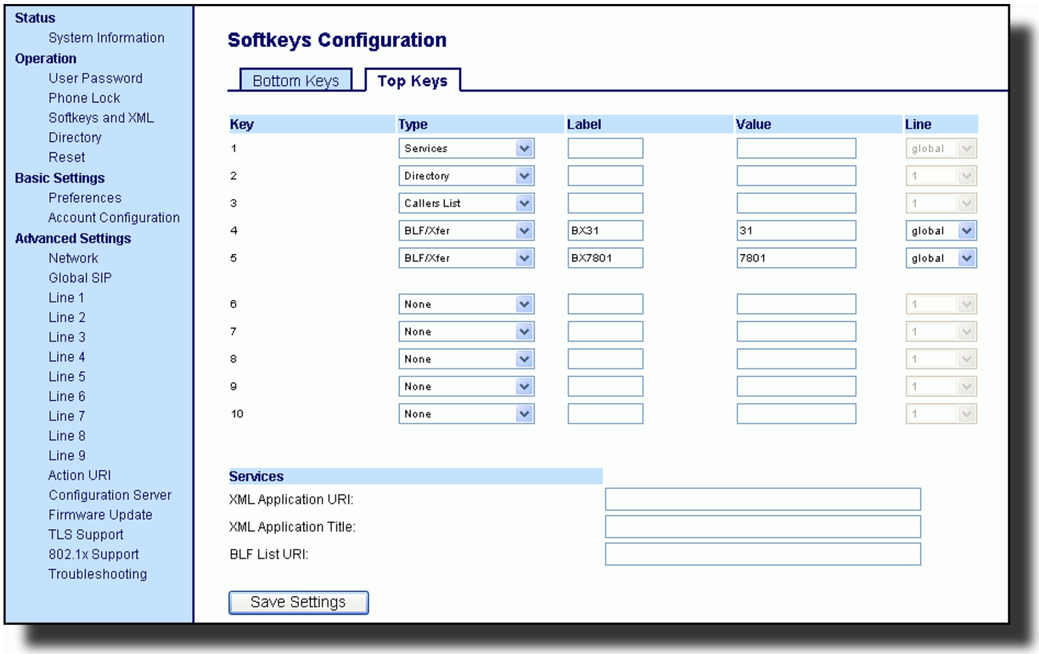
1 Click on **Operation->Softkeys and XML.**
or
Click on **Operation->Programmable Keys.**
or
Click on **Operation->Expansion Keys.**

Bottom Keys Tab (6757i example).



Aastra Web UI

Top Keys Tab (6757i example)



2	Choose a key that you want to assign the BLF/Xfer key or a Speed Dial/Xfer key to, and select **BLF/ Xfer** or **Speed Dial/Xfer** from the "**Type**" field.
3	In the "**Label**" field, enter a key label to assign to the BLF/Xfer key (for example, "**BX35**").
4	In the "**Value**" field, enter the monitored extension (for example, "**35**").
5	In the "**Line**" field, select the line for which you want to use the key functionality.
6	On the Bottom Key tab, in the States field, select the state(s) (idle, connected, incoming, outgoing, busy) for which you want to use on the key. **Note:** States are not applicable to programmable keys.
7	Click  Save Settings to save your changes.

Speed Dial/Conference Key

The IP Phones allow you to configure a softkey/programmable key/expansion module key to be used as a speed dial conference key (**Speed Dial/Conf** key) while remaining in the current call. This key allows a user on a call, to conference another party at a pre-defined number while remaining in the conference call.

For example, while on an active call, a user can use the Speed Dial/Conf key to dial a recording service and have the resulting conference recorded.

Note: If currently in a conference, the Speed Dial/Conf key is disabled on the active call.

How it Works

If you configure a softkey/programmable key/expansion module as a **Speed Dial/Conf** key, and you press this key while on an active call, the focused line changes to the dialing line. A **Cancel** softkey displays on the phone (only those phones that have LCDs larger then 3 lines), allowing you to abort the conference speed dial if required. The message "*Ringing...*" displays below the number when the far end is ringing. The message "*Conf. Unavailable*" briefly displays when a conference is already in progress or when the CT handset is in use. The active call is not put on hold when the speed dial number is dialed.

Limitations for Speed Dial/Conference Key

The following are limitations for the Speed Dial/Conference key:

- The CT handsets are not supported.
- The feature is not compatible with centralized conferencing.

The softkey/programmable key is called "**Speed Dial/Conf**" in the Web UI drop down list. In the configuration file, use "**speed dialconf**" as the softkey type.

Note: On the 6739i, you can configure a Speed Dial/Conf softkey using the IP Phone UI also. *For more information, see the Aastra Model 6739i IP Phone User Guide.*

Configuring the Speed Dial/Conf Key Using the Configuration Files

To configure the Speed Dial/Conf key using the configuration files, you enter "**speeddialconf**" for the key type. The following parameters are examples you can use to configure the Speed Dial/ Conf key:

softkey1 type: **speeddialconf**
softkey1 label: **Sales**
softkey1 value: **5645**
softkey1 line: **3**

topsoftkey1 type: **speeddialconf**
topsoftkey1 label: **Sales**
topsoftkey1 value: **5645**
topsoftkey1 line: **3**

prgkey1 type: **speeddialconf**
prgkey1 value: **5645**
prgkey1 line: **1**

expmod1 key1 type: **speeddialconf**
expmod1 key1 label: **Sales**
expmod1 key1 value: **5645**
expmod1 key1 line: **3**

Refer to the following in Appendix A to configure a Speed Dial/Conf key on the IP phone using the configuration files.

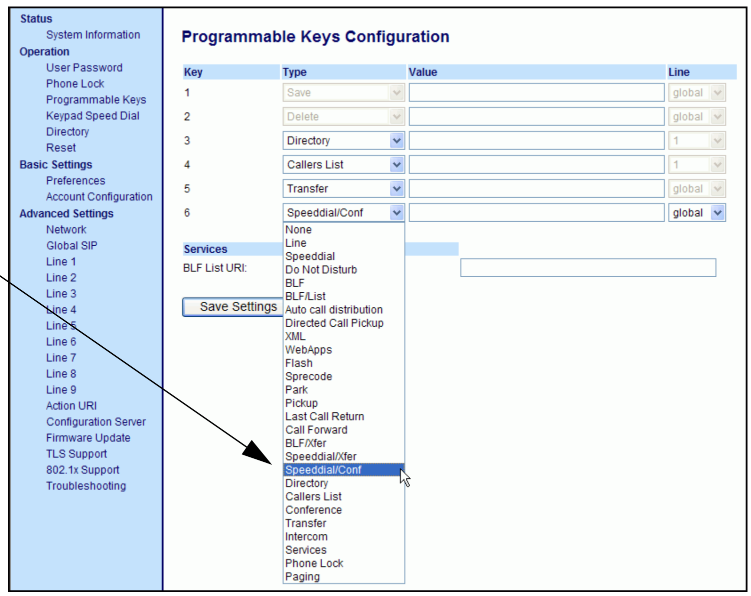
Configuration Files
To set a Speed Dial/Conf key using the configuration files, see Appendix A, "Softkey/Programmable Key/ Feature Key/Expansion Module Key Parameters" on page A-202.

Configuring the Speed Dial/Conf Key Using the Aastra Web UI

Use the following procedure to configure the Speed Dial/Conf Key using the Aastra Web UI.

<table>
<tr><th colspan="2">Aastra Web UI</th></tr>
<tr><td>1</td><td>Click on Operation->Sottkeys and XML.
or
Click on Operation->Programmable Keys.
or
Click on Operation->Expansion Module Keys.

Speeddial/Conf Option



</td></tr>
<tr><td>2</td><td>In the “Type” field, select Speed Dial/Conf from the list of options.</td></tr>
<tr><td>3</td><td>In the “Label” field, enter a key label to assign to the Speed Dial/Conf key (for example, “Sales”).</td></tr>
<tr><td>4</td><td>In the “Value” field, enter the number that the phone dials when the Speed Dial/Conf key is pressed (for example, “4556”).</td></tr>
<tr><td>5</td><td>In the “Line” field, select the line for which you want to use the key functionality.</td></tr>
<tr><td>6</td><td>For phones with softkeys:

In the States field, select the state(s) (idle, connected, incoming, outgoing, busy) for which you want to use on the key.

Note: States are not applicable to programmable keys (9143i, 6730i, 6731i, 6753i, 6755i).</td></tr>
<tr><td>7</td><td>Click [Save Settings] to save your changes.</td></tr>
</table>

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 5-166.

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 Timer 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.

Note: On the 6739i, you can configure an ACD softkey using the IP Phone UI also. *For more information, see the Aastra Model 6739i IP Phone User Guide.*

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-202.

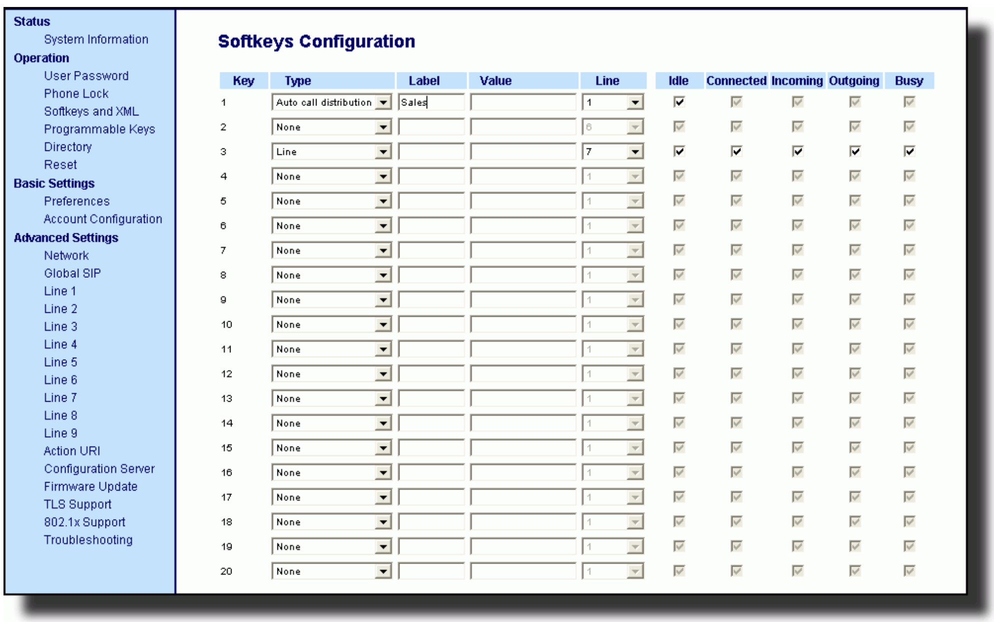
Configuring the ACD Auto-Available Timer.

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-198.

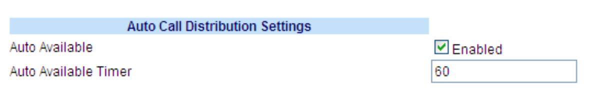
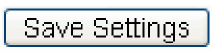
Configuring an ACD Key Using the Aastra Web UI

Use the following procedure to configure an ACD softkey, programmable key, or expansion module key using the Aastra Web UI. This procedure uses the 6755i 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 6757i 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 programmable keys on the 6730i, 6731i, 6753i, 6755i and 9143i.) For example: **Sales**
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**.
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  to save your changes.

Using the ACD Feature on your IP Phone

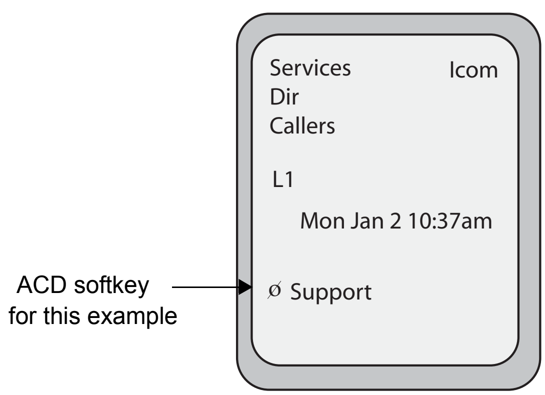
The ACD feature allows you to login to a phone queue in order to receive distributed calls on your IP phone. To login to a phone queue, your system administrator must preconfigure an ACD softkey or programmable key on your Aastra IP phone.

For models 8 and 11-Line LCD phones, 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 Ø 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.

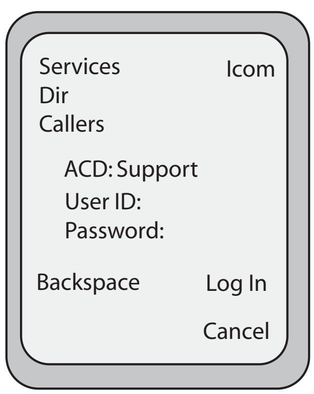
The table below describes the meaning of the LED, and each icon, as they may appear on your IP phone:

Phone Model	Status: Logged In and Available	Status: Unavailable	Logged Out
9143i	Solid Red LED	Blinking red LED	No LED
9480i, 9480i CT	Solid Red LED √ icon	Blinking Red LED Blinking √ icon	No LED Ø icon
6730i, 6731i, 6753i	Solid Red LED	Blinking red LED	No LED
6755i, 6757i, 6757i CT	Solid Red LED √ icon	Blinking Red LED Blinking √ icon	No LED Ø icon

Logging In to a Phone Queue (8 and 11-Line LCD phones)

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 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.” 
2	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.

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 (3-Line LCD phones)

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).

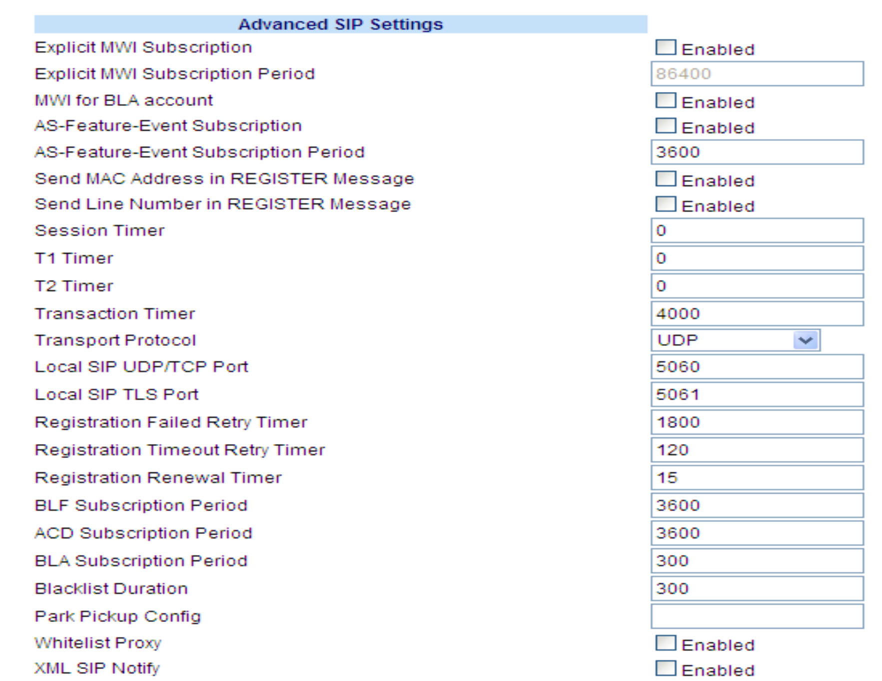
Setting this parameter to a value lower than 3600 allows the configured ACD feature to become active more quickly after a software/firmware upgrade or after a reboot of the IP phone. If you enter a value lower than 120 for this parameter, the default value (3600) will be used by the IP phone.

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-103.

	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.

BLA Subscription Period

The IP Phones include a SIP BLA subscription period parameter that allows an Administrator to set the amount of time, in seconds, of the BLA subscription period.

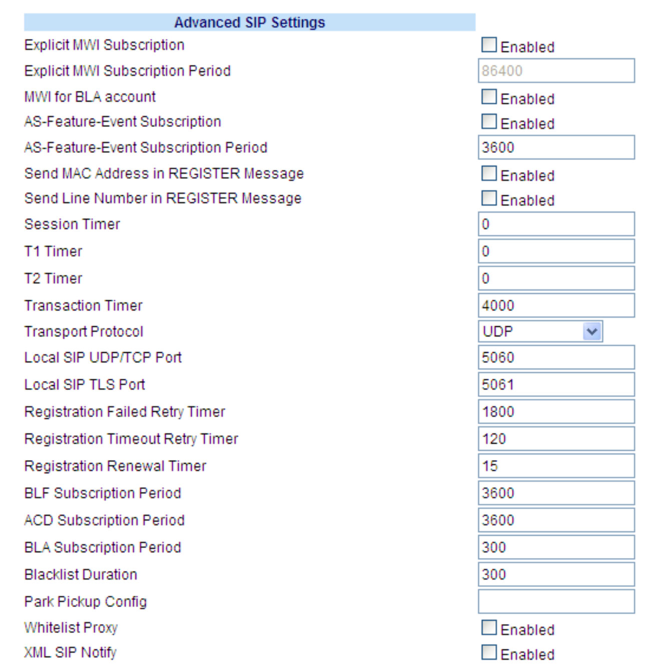
If this parameter is set to zero (0), the phone uses the value specified for the BLA expiration in the subscribe message received from the server. If no value is specified in the Subscribe message received from the server, the phone uses the default value of 300 seconds.

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

Configuring the BLA Subscription Period

Use the following procedures to configure the BLA Subscription Period.

Configuration Files
To configure the BLA subscription period on the IP phones using the configuration files, see Appendix A, the section, "Advanced SIP Settings" on page A-103.

<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>In the “BLA Subscription Period” field, enter a value, in seconds, that the phone waits to receive a BLA subscribe message from the server. If you specify zero (0), the phone uses the value specified for the BLA expiration in the subscribe message received from the server. If no value is specified, the phone uses the default value of 300 seconds. Valid values are 0 to 3700. Default is 300 seconds.</td></tr>
<tr><td>3</td><td>Click 
 to save your changes.</td></tr>
</table>

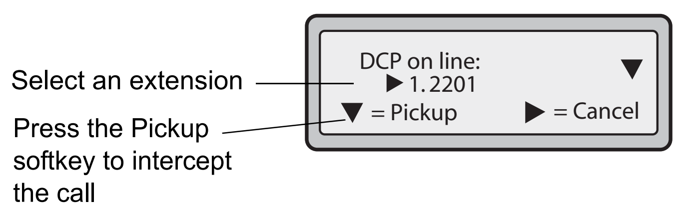
Directed Call Pickup/Group Call Pickup (for Sylantro Servers)

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.



Note: On the 6739i, you can configure DCP/GCP softkey using the IP Phone UI also. *For more information, see the Aastra Model 6739i IP Phone User Guide.*

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

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

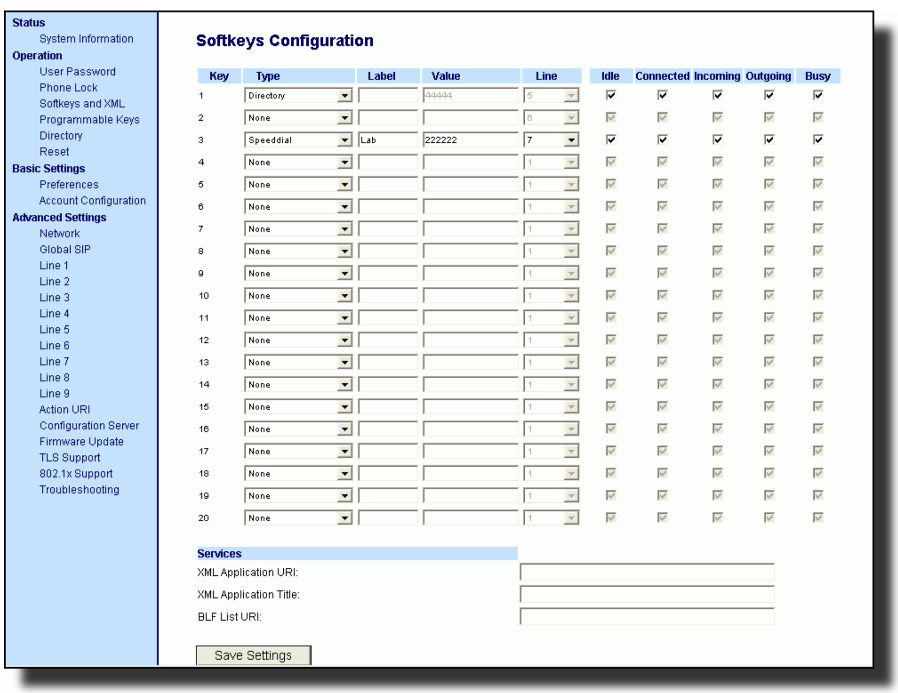
Configuration Files

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-202.

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 6757i IP Phone as an example.

Aastra Web UI

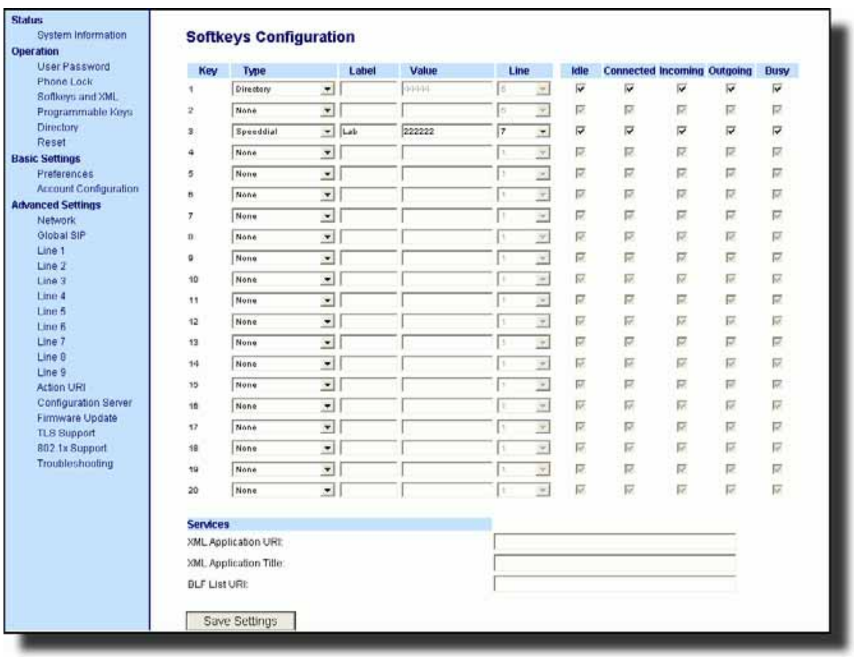
1	Click on **Operation->Softkeys and XML** **or** Click on **Operation->Programmable Keys** **or** Click on **Operation->Expansion Module <N>.** Depending on your phone-model, the key configuration screen displays. The 6755i Key Screen is shown as an example. 
2	Select a key for which to configure Directed Call Pickup.
3	In the "**Type**" field, select **Directed Call Pickup**.

Aastra Web UI	
4	In the **"Label"** field, specify a name for this Directed Call Pickup softkey. For example: **DCP2200**
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.

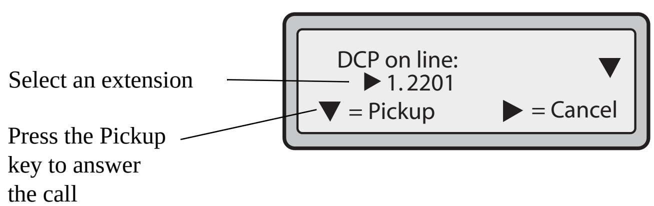
	Aastra Web UI
1	Click on **Operation->Softkeys and XML** **or** Click on **Operation->Programmable Keys** **or** Click on **Operation->Expansion Module <N>.** Depending on your phone-model, the key configuration screen displays. The 6755i Key Screen is shown as an example. 
2	Select a key for which to configure Group Call Pickup.
3	In the "**Type**" field, select **Directed Call Pickup**.
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.
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). An Administrator or User can set “do not disturb” based on the accounts on the phone (all accounts or a specific account). You can set specific modes for the way you want the phone to handle DND. The three modes you can set on the phone for DND are:

- **Account**
- **Phone**
- **Custom**

DND Account-Based Configuration

An Administrator or User can configure DND on the phone-side by setting a mode for the phone to use (**account**, **phone**, or **custom**). Once the mode is set, you can use the IP Phone UI to use the DND feature.

Notes:
1. If there is only one account configured on the phone, then the mode setting is ignored and the phone behaves as if the mode was set to “Phone”.

2. You must configure a DND key on the phone to be able to use this feature via the phone’s keypad.

The following describes the DND key behavior for each DND mode.

- **Account -** DND key toggles the account in focus on the IP Phone UI, to ON or OFF if DND enabled for that account.
- **Phone -** DND key toggles all accounts on the phone to ON or OFF.
- **Custom -** DND key displays custom screens on the IP Phone UI. User can select whether to enable/disable DND per account, enable DND on all accounts, or disable DND on all accounts.

The following table describes the DND key and Message Waiting Indicator (MWI) LEDs when you enable DND on the IP Phone.

Softkey LED Behavior for All Modes	MWI LED Behavior for All Modes
DND key LED RED if current account in focus has DND ON. **DND key LED OFF** when current account in focus has DND disabled.	**MWI LED ON** if current account in focus has DND ON. **MWI LED OFF** if current account in focus has DND OFF.

You can configure the DND softkey and the DND mode (**Account**, **Phone**, **Custom**) using the configuration files or the Aastra Web UI. Once you configure DND, you can access the DND screen on the IP Phone UI.

Important Notes

- In the Aastra Web UI, the "Account Configuration" page replaces the previous "Call Forward Settings" page.
- In the IP Phone UI, the new DND key feature now has new menu screens.
- **If you make changes to the configuration for DND via the IP Phone UI, you must refresh the Aastra Web UI screen to see the changes.**

Note: On the 6739i, you can configure a DND softkey using the IP Phone UI also. *For more information, see the Aastra Model 6739i IP Phone User Guide.*

Configuring DND Using the Configuration Files

You use the following parameters to configure DND on the IP Phone:

- **dnd key mode**
- **softkeyN type, topsoftkeyN type, prgkeyN type, or expmodX keyN type**
- **softkeyN states** (optional)

Note: If there is no DND key configured or if it is removed, DND is disabled on the IP Phone.

<u>Example</u>

The following is an example of configuring the mode for DND in the configuration files:

```
dnd key mode: 2
softkey1 type: dnd
softkey1 states: idle connected incoming outgoing busy
```

In the above example, softkey 1 is configured for DND for line 1 only, with a "**custom**" configuration. Pressing softkey 1 displays DND screens for which you can customize on the phone. For specific screens that display in the IP Phone UI, see the section, "Using DND Modes via the IP Phone UI" on page 5-187.

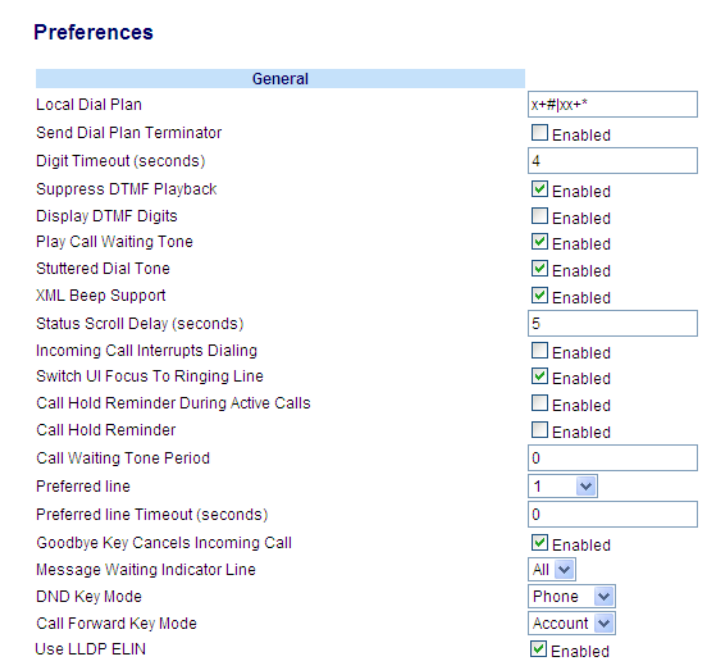
Configuration Files

To set DND in the configuration files, see Appendix A, the sections:

- "DND Key Mode Settings" on page A-165.
- "Softkey/Programmable Key/Feature Key/Expansion Module Key Parameters" on page A-202.

Configuring DND Using the Aastra Web UI

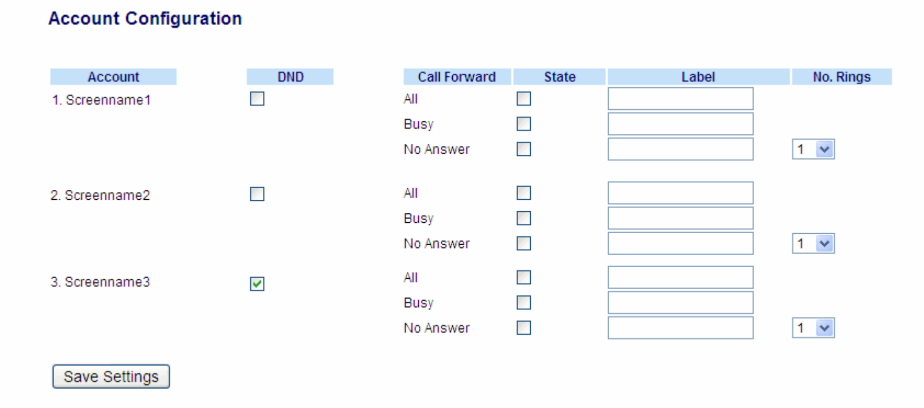
Use the following procedure to configure DND mode using the Aastra Web UI:

<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>In the “DND Key Mode” field, select a “do not disturb” (DND) mode to use on the phone.
Valid values are: Account, Phone, Custom. Default is Phone.

• account Sets DND for a specific account. DND key toggles the account in focus on the IP Phone UI, to ON or OFF.
• phone Sets DND ON for all accounts on the phone. DND key toggles all accounts on the phone to ON or OFF.
• custom Sets the phone to display custom screens after pressing the DND key, that list the account(s) on the phone. The user can select a specific account for DND, turn DND ON for all accounts, or turn DND OFF for all accounts.

Notes:
1. If there is only one account configured on the phone, then the mode setting is ignored and the phone behaves as if the mode was set to “Phone”.
2. Using the Aastra Web UI, if you change the DND Key Mode to “phone”, all accounts synchronize to the current setting of Account 1.</td></tr>
<tr><td>3</td><td>Click 
 to save your changes.

The change takes effect immediately without a reboot.</td></tr>
</table>

<table>
<tr><th colspan="2">Aastra Web UI</th></tr>
<tr><td>4</td><td>Click on Basic Settings->Account Configuration.

</td></tr>
<tr><td>5</td><td>For each account, enable DND by placing a check mark in the box. Disable DND by unchecking the box.

Notes:
1. If you selected “Account” or “Custom” mode in step 2, you can enable/disable each account or all accounts as applicable.
If you selected “Phone” mode, the first account allows you to change the DND status for all accounts.

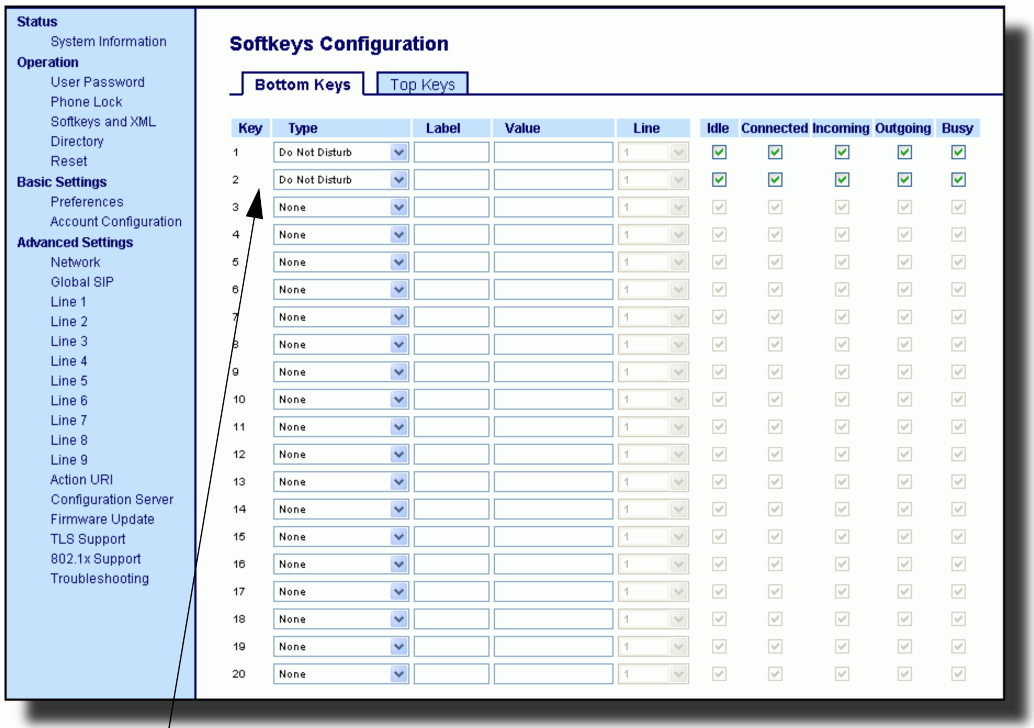
2. Number and name of accounts that display to this screen are dependant on the number and name of accounts configured on the phone. In the screen in step 4, Screenname1 is configured on Line 1, Screenname2 is configured on Line 2, and Screenname3 is configured on Line 3. The name for the account is dependant on the name specified for the “Screen Name” parameter at the path Advanced Settings->LineN. If you do not specify a value for the “Screen Name” parameter, the account name is based on the “Phone Number” parameter at the path Advanced Settings->LineN. If neither the “Screen Name” nor the “Phone Number” parameters are specified, the account name shows “1”, “2”, “3”, etc. only.</td></tr>
<tr><td>6</td><td>Click 

to save your changes.

The change takes effect immediately without a reboot.</td></tr>
</table>

Aastra Web UI

7 Click on **Operation->Softkeys and XML**;
or
Click on **Operation->Programmable Keys;**
or
Click on **Operation->Expansion Module**.



Softkeys 1 and 2 configured for DND

Note: If there is no DND key configured or if it is removed, DND is disabled on the IP Phone.

8 Click [Save Settings] to save your changes.

Using DND Modes via the IP Phone UI

If you add a DND key using the configuration files or the Aastra Web UI, you can toggle the DND state using the IP Phone UI. Use the following procedure to enable/disable DND on the IP Phone.

The following procedure assumes you have already configured a DND key AND assumes there are three accounts configured on the phone.

Notes:

1. If there is no DND key configured or if it is removed, DND is disabled on the IP Phone.

2. If there is only one account configured on the phone, then the mode setting is ignored and the phone behaves as if the mode was set to “Phone”.

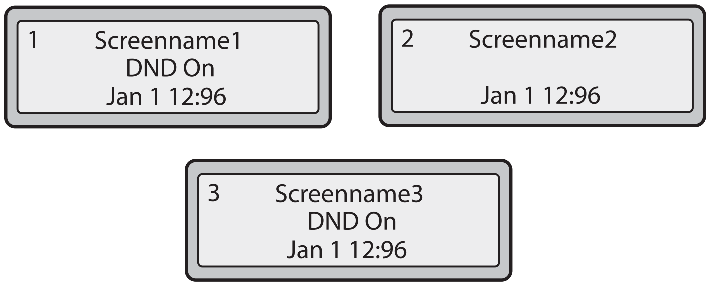
3. Using the Aastra Web UI, if you change the DND key mode to “phone”, all accounts synchronize to the current setting of Account 1.

Aastra IP Phone UI

Step	Action

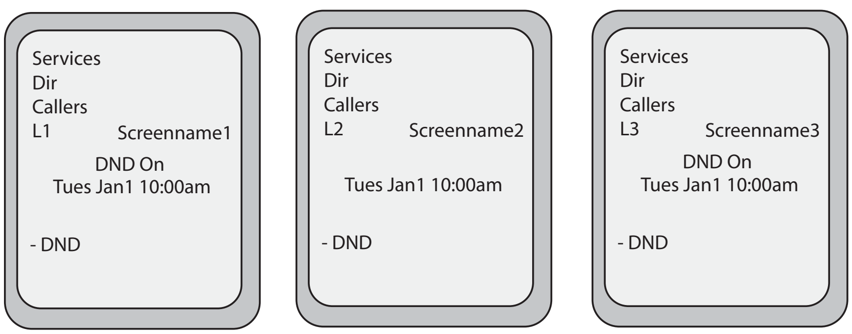
DND in Account Mode

On 3-Line LCD phones:
With the account in focus on the IP Phone UI, press the **DND** key to toggle DND ON or OFF for the account. Use the RIGHT and LEFT arrow keys to scroll through each account.



In the above example, Screenname1, Screenname2, and Screenname3, are three accounts configured on the phone. Only Screenname1 and 3 have DND ON. Screenname 2 has DND OFF.

On the 8 and 11-Line LCD phones:
With the account in focus on the IP Phone UI, press the **DND** key to toggle DND ON or OFF for the account. Use the RIGHT and LEFT arrow keys to scroll through each account.

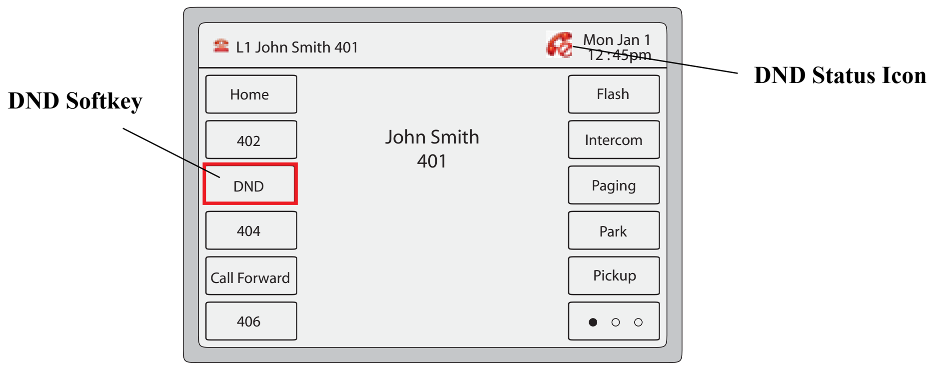


In the above example, Screenname1, Screenname2, and Screenname3, are three accounts configured on the phone. Only Screenname1 and 3 have DND ON. Screenname 2 has DND OFF.

Aastra IP Phone UI

Step	Action

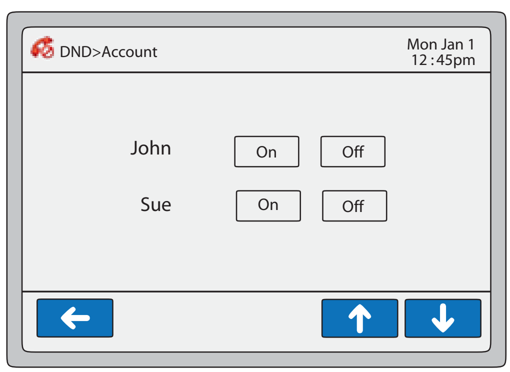
On the 6739i:
With the account in focus on the IP Phone UI, press the **DND** key to toggle DND ON or OFF for the account.



In the above example, pressing the DND key on the line in focus highlights the softkey in red to show that DND is enabled. The MWI LED illuminates ON. A DND icon appears in the status bar on the upper right of the screen. Pressing the DND key again disables DND on the line in focus, turns off the MWI LED, and the DND status icon disappears.

To enable/disable DND for another account:

1 Press the **<Services>** button.
2 Press the **<DND>** button.



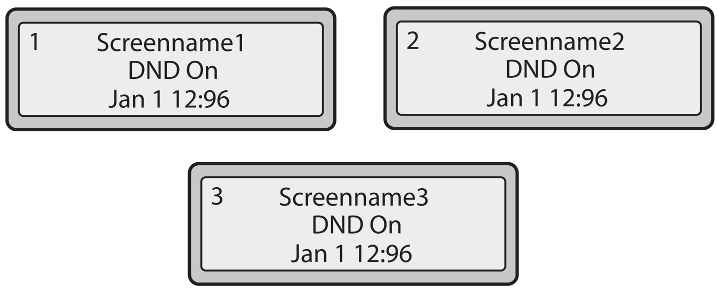
3 Press **<On>** or **<Off>** to enable/disable DND for a specific account. Use the scroll keys if applicable to scroll through accounts.

Aastra IP Phone UI

Step	Action

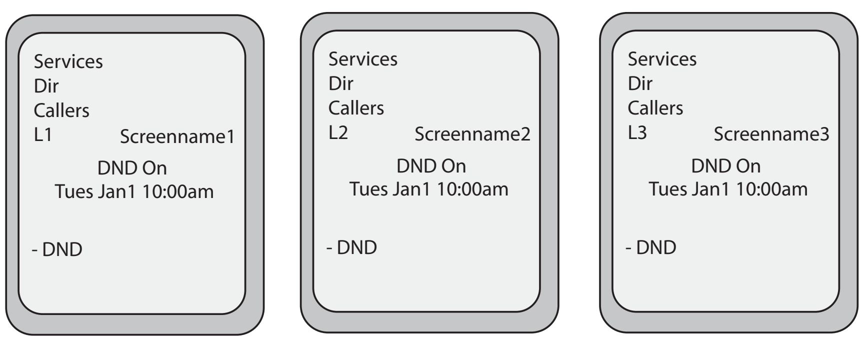
DND in Phone Mode

On 3-Line LCD phones:
Press the **DND** key to toggle DND ON or OFF for all accounts on the phone. Toggling to ON enables DND on all accounts on the phone. Toggling to OFF disables DND on all accounts on the phone. Use the RIGHT and LEFT arrow keys to scroll through each account.



In the above example, toggling Screenname1 to DND ON, enabled DND for Screenname2 and 3 also.

On the 8 and 11-Line LCD phones:
Press the **DND** key to toggle DND ON or OFF for all accounts on the phone. Toggling to ON enables DND on all accounts on the phone. Toggling to OFF disables DND on all accounts on the phone. Use the RIGHT and LEFT arrow keys to scroll through each account.

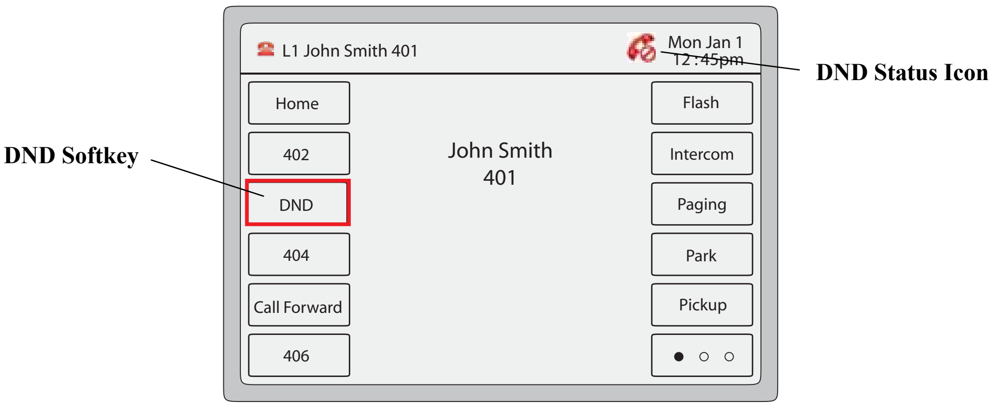


In the above example, toggling Screenname1 to DND ON, enabled DND for Screenname2 and 3 also.
Note: Enabling DND in "**Phone**" mode toggles all accounts on the phone to DND ON.

Aastra IP Phone UI

Step	Action

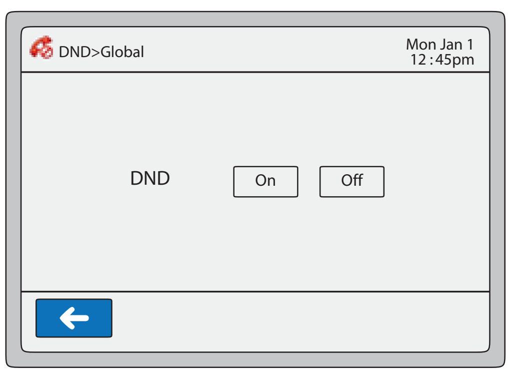
On the 6739i:
Press the **DND** key to toggle DND ON or OFF for all accounts on the phone.



In the above example, pressing the DND key highlights the softkey in red to show that DND is enabled for all accounts on the phone.The MWI LED illuminates ON. A DND icon appears in the status bar on the upper right of the screen. Pressing the DND key again disables DND on the phone, turn the MWI LED OFF, and the DND status icon disappears.

To enable/disable DND for all accounts on the phone:

1 Press the **<Services>** button.
2 Press the **<DND>** button.



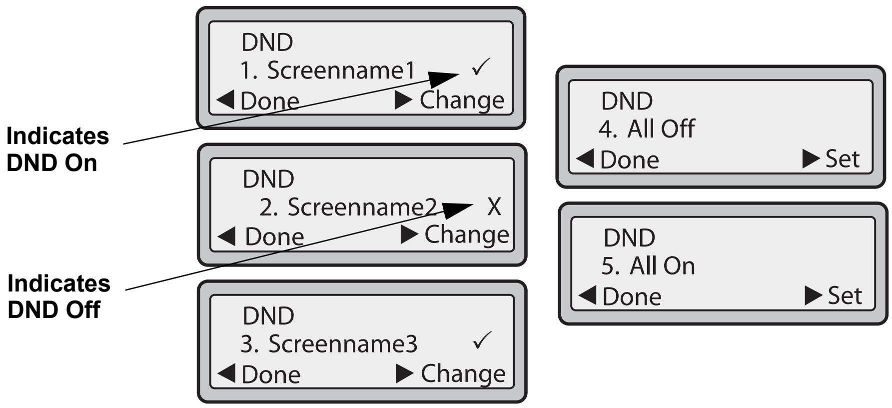
3 Press **<On>** or **<Off>** to enable/disable DND for all accounts.

Aastra IP Phone UI

Step	Action

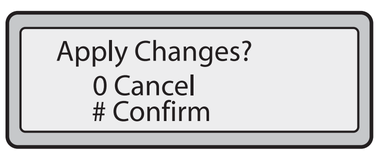
DND in Custom Mode

On 3-Line LCD phones:
Press the **DND** key on the phone. The screen displays a list of the accounts on the phone and allows you to enable/disable a specific account or all accounts. Use the UP and DOWN arrow keys to scroll through the accounts.

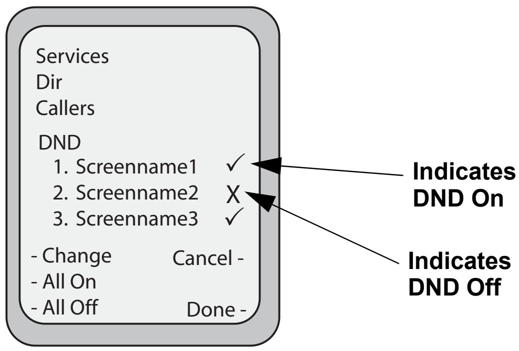


In the above example, Screenname1 and 3 have **DND ON** as indicated by a checkmark (✓). Screenname2 has DND off as indicated by an **X**. Items 4 and 5 allow you to disable or enable DND on all accounts, respectively.

You use the **CHANGE** key to enable or disable DND for a specific account. You use the **SET** key to enable/disable DND for all accounts. After making the change, you must press **DONE** and then **Confirm** (**#**) to save the change. Pressing **Cancel** (**0**) cancels the attempted change. The following screen displays after pressing a **DONE** key:



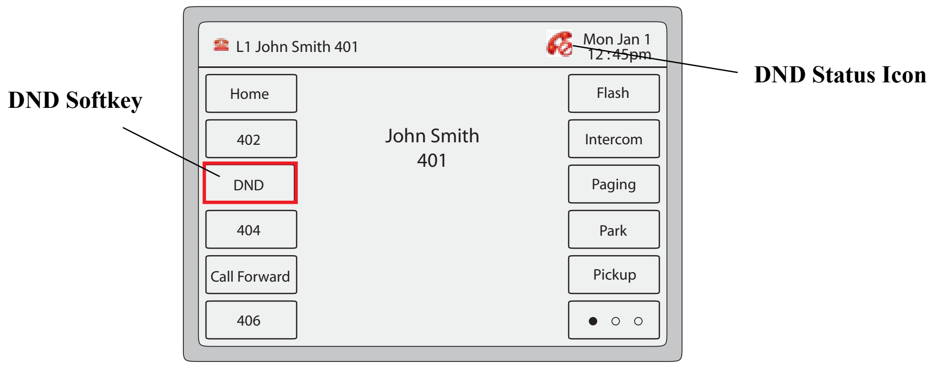
Aastra IP Phone UI

Step	Action
DND in Custom Mode (continued)	
	On the 8 and 11-Line LCD phones: Press the **DND** key on the phone. The screen displays a list of the accounts on the phone and allows you to enable/disable a specific account or all accounts. Use the UP and DOWN arrow keys to scroll through the accounts.  In the above example, Screenname1 and 3 have **DND ON** as indicated by a checkmark (✓). Screenname2 has DND off as indicated by an **X**. The **ALL ON** and **ALL OFF** softkeys allow you to enable or disable DND on all accounts, respectively. You use the **CHANGE** key to enable or disable DND for a specific account selected. After making the change, you must press **DONE** to save the change.

Aastra IP Phone UI

Step	Action

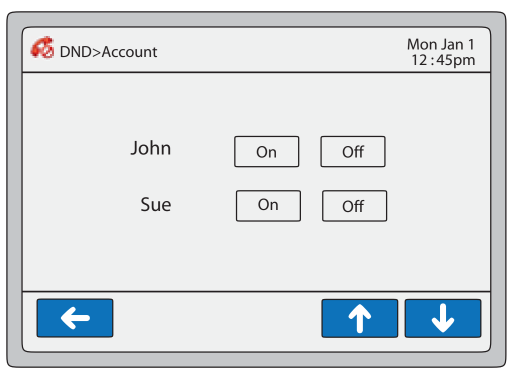
On the 6739i:
With the account in focus on the IP Phone UI, press the **DND** key to toggle DND ON or OFF for the account.



In the above example, pressing the DND key on the line in focus highlights the softkey in red to show that DND is enabled. The MWI LED illuminates ON. A DND icon appears in the status bar on the upper right of the screen. Pressing the DND key again disables DND on the line in focus, turns off the MWI LED, and the DND status icon disappears.

To enable/disable DND for another account:

1. Press the **<Services>** button.
2. Press the **<DND>** button.



3. Press **<On>** or **<Off>** to enable/disable DND for a specific account. Use the scroll keys if applicable to scroll through accounts.

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.

Bridged Line Appearance (BLA)

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

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

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

Note: 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.

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
9143i	9
9480i	9
9480i CT	9
6730i	6
6731i	6
6739i	9
6753i	9
6755i	9
6757i	9
6757i CT	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

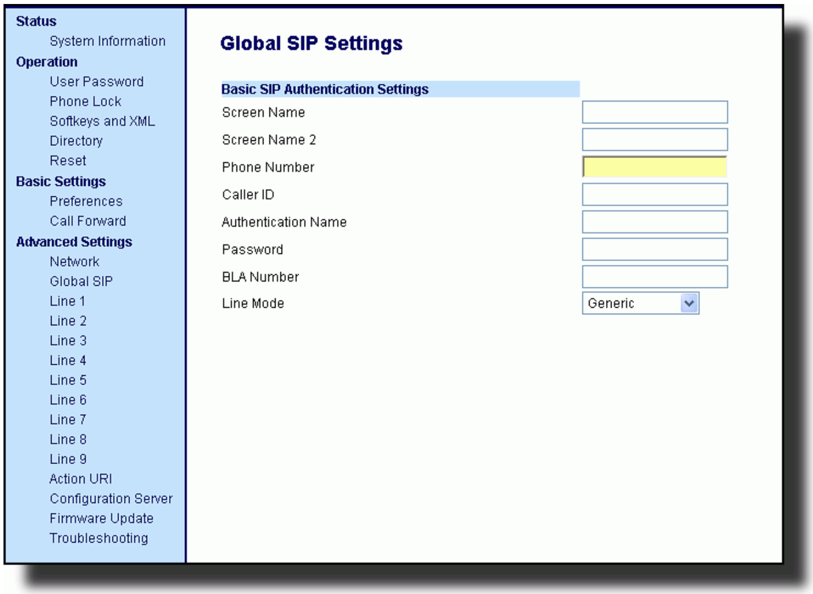
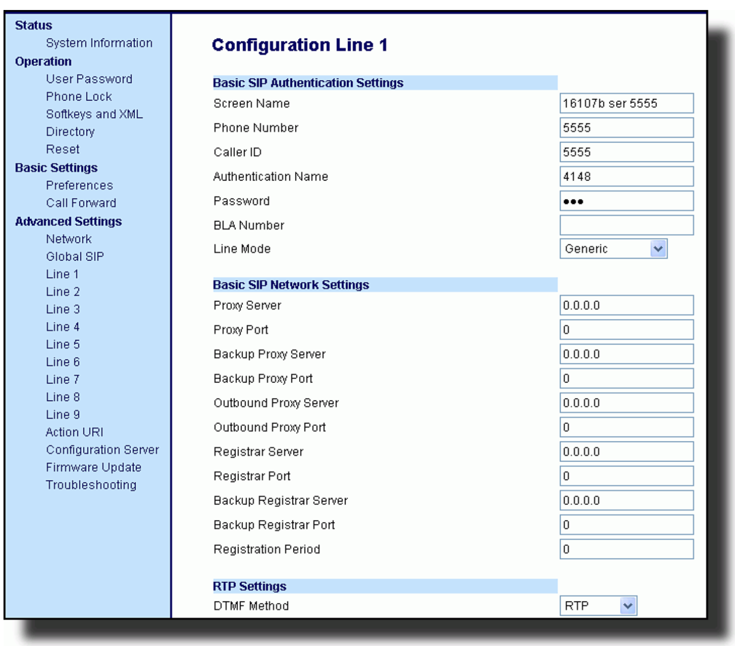
Configuration Files

For specific **global** parameters you can set in the configuration files, see Appendix A, the section, "SIP Basic, Global Settings" on page A-74.

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-86.

	Aastra Web UI
1	***For global configuration of BLA:*** Click on **Advanced Settings->Global SIP->Basic SIP Authentication Settings.** 
	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.

Aastra Web UI	
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.

BLA Subscription Period

The phones include a SIP BLA subscription period parameter that allows an Administrator to set the amount of time, in seconds, of the BLA subscription period.

Reference

For more information about setting the BLA Subscription Period, see the section, "BLA Subscription Period" on page 5-175.

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
```

P-Preferred Identity Header for BLA Accounts

The IP Phones support the BLA specification, draft-anil-sipping-bla-02, which states that the P-Preferrred-Identity header (RFC3325) gets added to the INVITE message to indicate the Caller-ID that is used for the call.

Note: The P-Preferred-Identity for BLA accounts is also sent for hold/ unhold messages.

BLA Support for Message Waiting Indicator (MWI)

The IP Phones have an option for a Busy Line Appearance (BLA) configured line to send a SUBSCRIBE SIP message for a Message Waiting Indicator (MWI).

Notes:

1. If you change the setting on this parameter, you must reboot the phone for it to take affect.

2. Both the "**sip explicit mwi subscription**" and "**sip mwi for bla account**" parameters must be enabled in order for the MWI subscription for BLA to occur.

3. The MWI re-subscription for the BLA account uses the value set for the "**sip explicit mwi subscription period**" parameter to re-subscribe.

4. Whether or not the "**sip mwi for bla account**" parameter is enabled, the priority for displaying MWI does not change.

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

Limitations

The following are limitations of the BLA Support for MWI feature:

- The phone shows MWI for the first matching identity if more than one line with different user names has the same BLA account.
- If a normal line has the same user name as the BLA user of another line, the phone shows MWI only for the normal line.

Use the following procedure to configure BLA support for MWI.

Configuring BLA Support for MWI Using the Configuration Files

Configuration Files

For specific parameters you can set in the configuration files, see Appendix A, the section, "BLA Support for MWI" on page A-97.

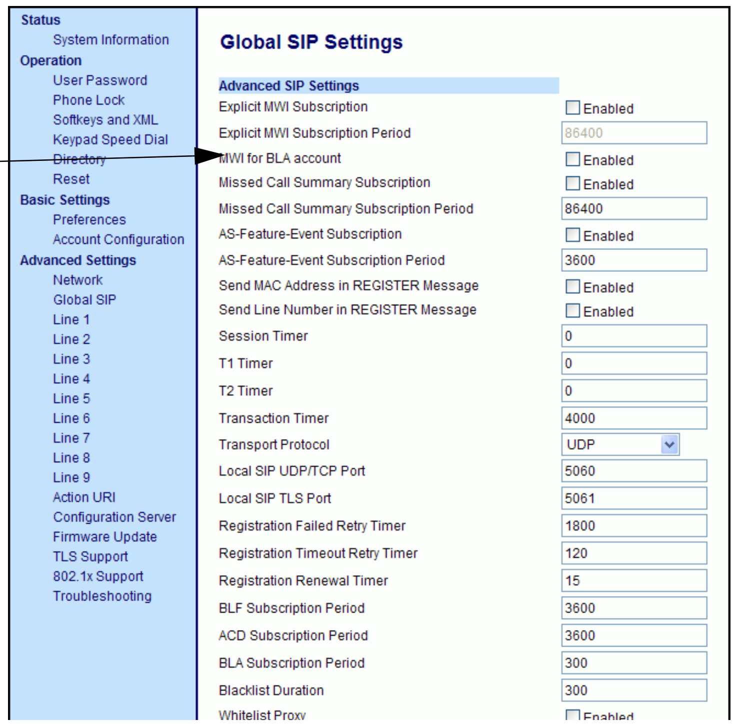
Configuring BLA Support for MWI using the Aastra Web UI

Use the following procedure to configure BLA support for MWI.

Aastra Web UI

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

MWI for BLA Account Parameter



2 The "**MWI for BLA Account"** field is disabled by default. To enable this feature, place a checkmark in the "**Enabled**" box.

Notes:
1. If you change the setting on this parameter, you must reboot the phone for it to take affect.
2. Both the "**sip explicit mwi subscription**" and "**sip mwi for bla account**" parameters must be enabled in order for the MWI subscription for BLA to occur.
3. The MWI re-subscription for the BLA account uses the value set for the "**sip explicit mwi subscription period**" parameter to re-subscribe.
4. Whether or not the "**sip mwi for bla account**" parameter is enabled, the priority for displaying MWI does not change.

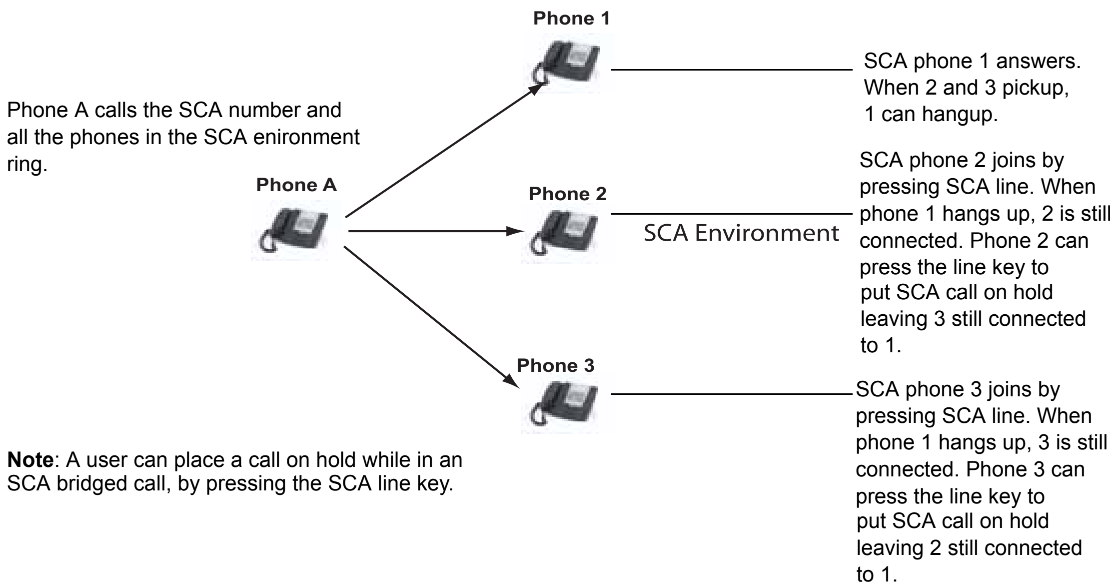
3 Click **Save Settings** to save your changes and reboot the phone for the change to take affect.

Shared Call Appearance (SCA) Call Bridging

Shared Call Appearance (SCA) is when incoming calls are presented to multiple phones simultaneously. For example, it is the ability to assign the boss' extension to a button on the secretary's phone. Calls can be transferred between two phones with the same extension button by simply putting the call on hold at one phone and picking it up on the other. Status LEDs light and flash in unison, allowing all people sharing the extension to see the status at a glance.

The IP Phones include an enhanced SCA for the servers that support call bridging and allows two or more SCA users to be connected in a call with a third party.

Refer to the following example.



Using the example above, when a call comes into Phone 1, Phone 2 and Phone 3 can pickup the same call by pressing the SCA line key. Phone 2 and 3 display the call they are bridging into on the LCD of the phones. Existing SCA parties in a bridge or one-to-one call hear an audible beep when another party has joined the call.

Note: Enabling/disabling the beep is configurable on the server-side.

If a phone is configured for SCA bridging and it attempts to join a call, but the account on the server does not have this functionality enabled, an error message displays to the LCD on the phone.

The SCA call bridging feature is disabled by default on all phones. You can enable this feature on a global or per-line basis using the configuration files only.

Note: A “Call-Info” header is included in the requests as well as the 200ok response to an INVITE, RE-INVITE, and UPDATE messages for SCA lines.

Keys States and LED Behavior

There are two new call states on the phones that support SCA bridging:

- **Bridge-active** - A bridged call is in progress
- **Bridge-held** - The 3rd-party (i.e., non-SCA party) in the bridge is on hold.

The following tables provide the key states and LED behavior in an SCA bridge call for users involved in an SCA call and users not involved in the SCA call.

Line Keys and Idle Screens

State	Call LED	Call Caller ID	Non-Call LED	Non-Call Caller ID
Idle	N/A	N/A	Off	N/A
Seized	Solid Green	None	Solid Red	None
Progressing (outgoing call)	Green	Called Part7	Solid Red	None
Alerting (incoming call)	Blinking Unselected Red	N/A		
Active	Solid Green	Far-end	Solid Red	Far-end
Held	Slow Flashing Green	Far-end	Slow Flashing Red	Far-end
Hold private	Slow Flashing Green	Far-end	Solid Red	Far-end
Bridge-active	Solid Green	Far-end	Solid Red	Far-end
Bridge-held	Slow Flashing Green	Far-end	Solid Red	Far-end

Softkey Line Keys

State	Call Icon	Call LED	Non-Call Icon	Non-Call LED
Idle	Small circle	None	Small circle	None
Seized	N/A	N/A	Sold Circle	Solid Red
Progressing (outgoing call)	Empty circle	Solid Red	Sold Circle	Solid Red
Alerting (incoming call)	Empty blinking circle	Flashing Red	N/A	N/A
Active	Empty circle	Solid Red	Sold Circle	Solid Red
Held	Reverse empty blinking circle	Slow Flashing Red	Sold Reverse Circle	Slow Flashing Red
Hold private	Reverse empty blinking circle	Slow Flashing Red	Sold Circle	Solid Red
Bridge-active	Empty circle	Solid Red	Sold Circle	Solid Red
Bridge-held	Reverse empty blinking circle	Slow Flashing Red	Sold Circle	Solid Red

9143i Phone LED States

State	Call LED	Non-Call LED
Idle	N/A	N/A
Seized	Solid Red	Solid Red
Progressing (outgoing call)	Solid Red	Solid Red
Alerting (incoming call)	Flashing Red	Flashing Red
Active	Solid Red	Solid Red
Held	Slow Flashing Red	Slow Flashing Red
Hold private	***Slow Flashing Red***	***Solid Red***
Bridge-active	***Solid Red***	***Solid Red***
Bridge-held	***Slow Flashing Red***	***Solid Red***

Line key Phone Behavior

State	**Call Line Key Pressed**	**Non-Call Line Key Pressed**
Idle	N/A	Attempt to seize the line
Seized	***Hang up***	Ignore
Progressing	Hang up	Ignore
Alerting	answer	N/A
Active	***Hold***	***Bridge***
Held	***Retrieve***	***Bridge***
Hold private	***Retrieve***	***Ignore***
Bridge-active	***Hold***	***Bridge***
Bridge-held	***Retrieve***	***Bridge***

Enabling/Disabling SCA Call Bridging Feature

Use the following procedure to enable/disable SCA call bridging on the IP Phones.

Configuration Files

For specific parameters you can set in the configuration files, see Appendix A, the section, "Shared Call Appearance (SCA) Call Bridging" on page A-98.

Park/Pick Up Softkey

The IP phones (including the CT handsets) have a park and pickup call feature that allows you to park a call and pickup a call when required. Administrators can configure the park and pickup call keys using the configuration files or the Aastra Web UI. Users can make changes to customize the label and the state of the park/pick up keys using the Web UI.

➡ **Note:** On the 6739i, you can configure Park/Pickup softkeys using the IP Phone UI also. *For more information, see the Aastra Model 6739i IP Phone User Guide.*

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

The following paragraphs describe the configuration of a park and pickup softkey on the IP phone.

Park/Pickup Programmable Configuration (using a softkey)

The programmable method of configuration creates park and pickup keys (softkeys, programmable keys, expansion module keys) that you can configure on the IP phones.

For all 8 and 11-Line LCD phones you can set a key as "Park" or "Pickup" and then:

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

For 3-Line LCD display phones, you can set a programmable key as "Park" or "Pickup".

On 8 and 11-Line LCD Phone UIs

- 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 Model CTs, the customized labels apply to the base unit only. On the Model CT handsets, pressing the **Features** key displays the default labels of "Park" and "Pickup".

➡ **Note:** On the 9480i/9480i CT and 6757i/6757i 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 3-Line LCD Phone UIs

- 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.

Configuring Park/Pickup Key Using Configuration Files

In the configuration files, you configure the park/pickup keys 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.

Examples for Models with 8 and 11-Line LCDs

Server	Park Configuration	Pickup Configuration
Asterisk	softkeyN type: **park** softkeyN label: **parkCall** softkeyN states: **connected*** sip lineN park pickup config: **70;70;asterisk**	softkeyN type: **pickup** softkeyN label: **pickupCall** softkeyN states: **idle, outgoing**** sip lineN park pickup config: **70;70;asterisk**
Sylantro	softkeyN type: **park** softkeyN label: **parkCall** softkeyN states: **connected*** sip lineN park pickup config: **98;99;sylantro**	softkeyN type: **pickup** softkeyN label: **pickupCall** softkeyN states: **idle, outgoing**** sip lineN park pickup config: **98;99;sylantro**
BroadWorks	softkeyN type: **park** softkeyN label: **parkCall** softkeyN states: **connected*** sip lineN park pickup config: **68;88;broadworks**	softkeyN type: **pickup** softkeyN label: **pickupCall** softkeyN states: **idle, outgoing**** sip lineN park pickup config: **68;88;broadworks**

*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".

Examples for Models with 3-Line LCDs

Server	Park Configuration	Pickup Configuration
Asterisk	prgkeyN type: **park** sip lineN park pickup config: **70;70;asterisk**	prgkeyN type: **pickup** sip lineN park pickup config: **70;70;asterisk**
Sylantro	prgkeyN type: **park** sip lineN park pickup config: **98;99;sylantro**	prgkeyN type: **pickup** sip lineN park pickup config: **98;99;sylantro**
BroadWorks	prgkeyN type: **park** sip lineN park pickup config: **68;88;broadworks**	prgkeyN type: **pickup** sip lineN park pickup config: **68;88;broadworks**

Note: The 6730i, 6731i, 6753i, 6755i and 9143i do not allow for the configuration of labels and states for programmable keys.

Configuration Files

For specific parameters you can set in the configuration files, see Appendix A, the section, “Softkey Settings for 8 and 11-Line LCD phones” on page A-203 and “Programmable Key Settings for 9143i, 6730i, 6731i, 6753i, and 6755i” on page A-210.

Configuring a Park/Pickup Key Using Aastra Web UI

For 8 and 11-Line LCD phones, CT handsets and 3-line LCD phones, you first configure the park and pickup keys at **Advanced Settings -> Line 1-9** by entering the appropriate value based on the server in your network.

Note: Applicable values depend on the server in your network (Asterisk, BroadWorks, and Sylantro). See the table below for applicable values.

For 8 and 11-Line LCD phones, you can enter a key label and change the sofkey states at **Operation->Softkeys and XML.** The default state of the park configuration is "**connected**". The default state of the Pickup configuration is "**idle, outgoing**".

For CT handsets, you can enter a key label at **Operation->Handset Keys**. If park or pickup are enabled on more than one line on the base unit, the CT handsets use the first programmable configuration.

For example, if line 1 and line 6 are configured for park, the CT handsets use the configuration set for line 1 to park a call.

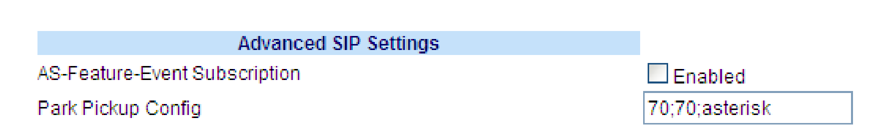
On 3-Line LCD phones, you can enter a key label at **Operation->Softkeys and XML.**

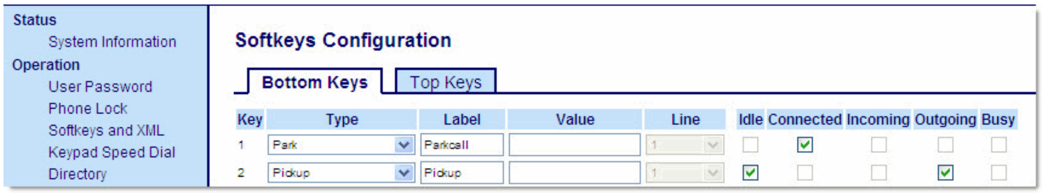
Park/Pickup Call Server Configuration Values

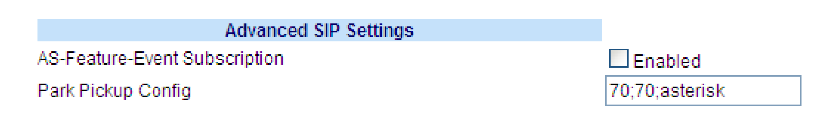
Server	Park Values*	Pickup Values*
Aasterisk	70	70
Sylantro	98	99
BroadWorks	68	88

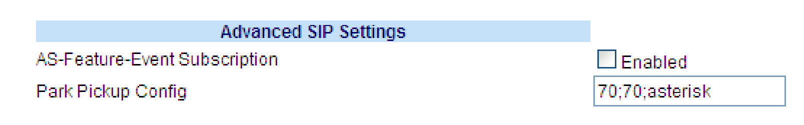
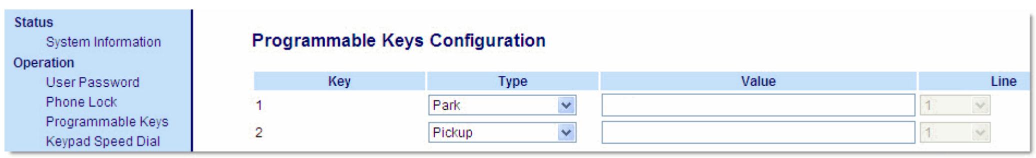
Use the following procedure to configure the park/pickup call feature using the Aastra Web UI.

Aastra Web UI

For 8 and 11-Line LCD Phones	
1	Click on **Advanced Settings ->Line 1** (you can select any line)
2	Under **Advanced SIP Settings** in the "**Park Pickup Config**" 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 5-209. **Note:** Leave the park/pickup configuration field blank to disable the park and pickup feature.

	Aastra Web UI
3	Click on **Operation->Softkeys and XML.** 
4	Pick a softkey to configure for parking a call.
5	In the "**Type**" field, select **Park**.
6	In the "**Label**" field, enter a label for the park softkey. **Note:** The" **Value**" and "**Line**" fields are already configured from the "**Park Pickup Config**" field.
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. **Note:** The" **Value**" and "**Line**" fields are already configured from the "**Park Pickup Config**" field.
11	The pickup softkey has a default state of "**Idle**" and "**Outgoing**". Leave these states enabled or to disable, uncheck the check boxes.
12	Click Save Settings to save your changes.
13	Click on **Reset**, then click Restart to restart the IP phone and apply the changes.

	Aastra Web UI
For CT Handset:s	
1	Click on **Advanced Settings -> Line 1** (you can select any line)
2	Under Advanced SIP Settings in the "**Park Pickup Config**" 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 5-209. **Note:** Leave the park/pickup configuration field blank to disable the park and pickup feature.
3	Click on **Operation->Handset Keys.** 
4	Pick a handset key to configure for parking a call.
5	In the "**Type**" field, select **Park**.
6	In the "**Label**" field, enter a label for the park softkey.
7	Pick another handset key to configure for picking up a call.
8	In the "**Type**" field, select **Pickup**.
9	In the "**Label**" field, enter a label for the pickup softkey.
10	Click Save Settings to save your changes.
11	Click on **Reset**, then click Restart to restart the IP phone and apply the changes.

Aastra Web UI	
For 3-Line LCD phones:	
1	Click on **Advanced Settings -> Line 1** (you can select any line)
2	Under Advanced SIP Settings in the "**Park Pickup Config**" 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 5-209. **Note:** Leave the park/pickup configuration field blank to disable the park and pickup feature.
3	Click on **Operation->Programmable Keys.** ***6731i Screen*** 
4	Pick a key to configure for parking a call.
5	In the "**Type**" field, select **Park**. **Note:** The" **Value**" and "**Line**" fields are already configured from the "**Park Pickup Config**" field.
6	Pick a key to configure for picking up a call.
7	In the "**Type**" field, select **Pickup**. **Note:** The" **Value**" and "**Line**" fields are already configured from the "**Park Pickup Config**" field.
8	Click Save Settings to save your changes.
9	Click on **Operation->Reset**.
10	Click on **Reset**, then 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. 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.
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) (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.

Note: On the 6739i, you can configure a Last Call Return softkey using the IP Phone UI also. *For more information, see the Aastra Model 6739i IP Phone User Guide.*

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

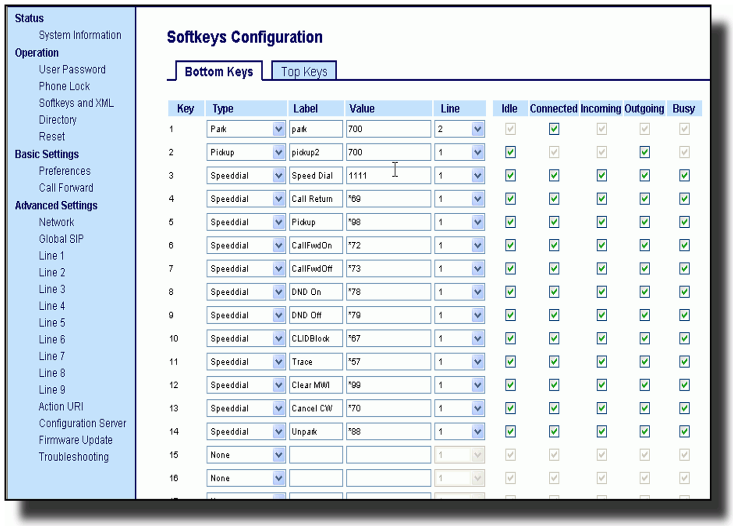
Use the following procedures to configure LCR on the IP phones.

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-202.

Aastra Web UI

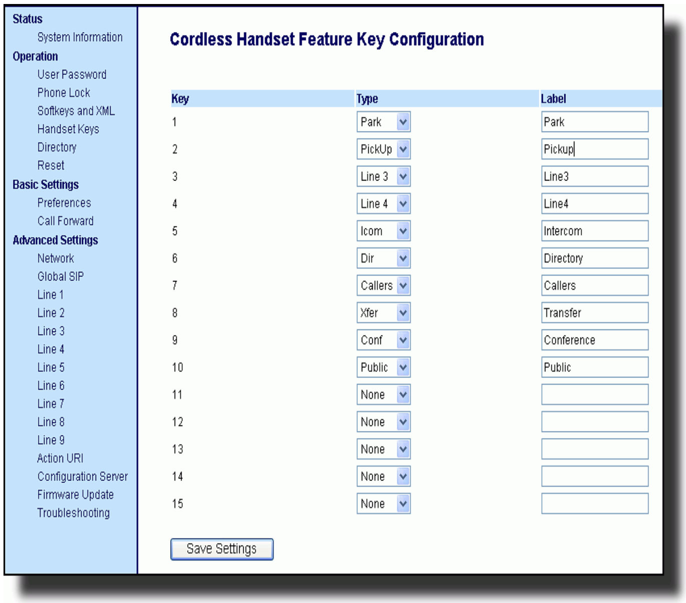
For the 6757i/6757i CT:

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  to save your changes.

Aastra Web UI

For the 6757i CT Handset:

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



2 Pick a handset key to configure for Last Call Return.

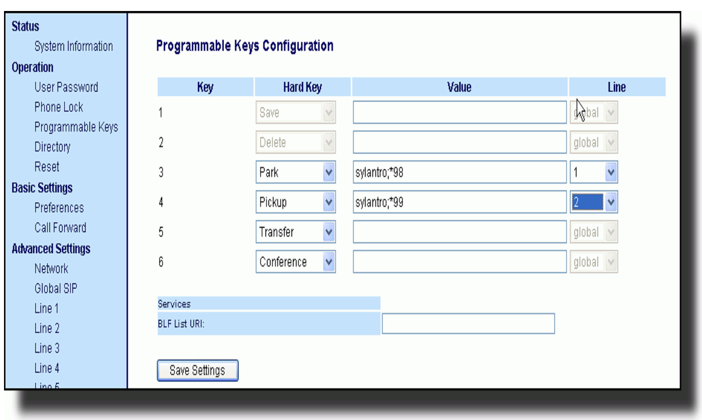
3 In the "**Key Function**" field, select **lcr**.

4 Click  to save your changes.

For 3-Line LCD phones:

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

6753i Screen



2 Pick a key to configure for Last Call Return.

3 In the "**Type**" field, select **lcr**.

	Aastra Web UI
4	***For 3-Line LCD phones:*** In the "**Line**" field, select a line for which to apply the lcr configuration.
5	Click Save Settings to save your changes.

Call Forwarding

Call Forward (CFWD) 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.

An Administrator or User can configure CFWD on the phone-side by setting a mode for the phone to use (**Account**, **Phone**, or **Custom**). Once the mode is set, you can use the IP Phone UI to use the CFWD feature at *Options->Call Forward* or by pressing a configured Call Forward softkey/programmable key/extension module key.

The following describes the behavior for each CFWD mode.

- **Account mode -** The account mode allows you to configure CFWD on a per account basis. Pressing a configured CFWD key applies to the account in focus.
- **Phone mode -** The Phone mode allows you to set the same CFWD configuration for all accounts (**All**, **Busy**, and/or **No Answer**). When you configure the initial account, the phone applies the configuration to all other accounts. (In the Aastra Web UI, only the account you configured is enabled. All other accounts are grayed out but set to the same configuration.) Using the Aastra Web UI, if you make changes to that initial account, the changes apply to all accounts on the phone.
- **Custom mode -** The Custom mode allows you to configure CFWD for a specific account or all accounts. You can configure a specific mode (**All**, **Busy**, and/or **No Answer**) for each account independently or all accounts. On 3-Line LCD phones, you can set all accounts to **ALL On** or **ALL Off**. On the 8 and 11-Line LCD phones, you can set all accounts to **All On**, **All Off**, or copy the configuration for the account in focus to all other accounts using a **CopytoAll** softkey.

Note: If there is only one account configured on the phone, then the mode setting is ignored and the phone behaves as if the mode was set to "Phone".

You can enable different call forwarding rules/modes independently (for example, you can set different phone numbers for Busy, All, and NoAns modes and then turn them on/off individually).

The following table describes the key and Message Waiting Indicator (MWI) LEDs when you enable CFWD on the IP Phone.

Key LED Behavior for All Modes	MWI LED Behavior for All Modes
CFWD key LED RED if CFWD All, CFWD Busy, or CFWD No Answer is enabled for the account in focus. **CFWD key LED OFF** if any CFWD mode is disabled.	**MWI LED ON** if current account in focus has CFWD ALL enabled. **MWI LED OFF** if CFWD All is disabled.

You can enable/disable CFWD and set a CFWD key using the configuration files or the Aastra Web UI. You can set CFWD mode (**Account**, **Phone**, **Custom**) using the configuration files, Aastra Web UI or IP Phone UI.

Important Notes

- In the configuration files, the "**call forward key mode**" parameter in the section, "Configuring Call Forwarding" on page 5-219 is in addition to the previous call forward parameter (**call forward disabled**). You can still use the previous call forwarding parameter if desired in the configuration files.
- In the IP Phone UI, you can access the Call Forwarding features at the path *Options->Call Forward* or by pressing a configured CFWD key.
- **If you make changes to the configuration for CFWD via the IP Phone UI, you must refresh the Aastra Web UI screen to see the changes.**

Note: On the 6739i, you can configure a Call Forward softkey using the IP Phone UI also. *For more information, see the Aastra Model 6739i IP Phone User Guide.*

Configuring Call Forwarding

You use the following parameters to set CFWD on the IP Phone using the configuration files:

- **call forward key mode**
- **softkeyN type, topsoftkeyN type, prgkeyN type, or expmodX keyN type**
- **softkeyN states** (optional)

Notes:

1. If there is no CFWD key configured on the phone or it is removed, you can still set the CFWD modes via the IP Phone UI at the path *Options->Call Forward*.

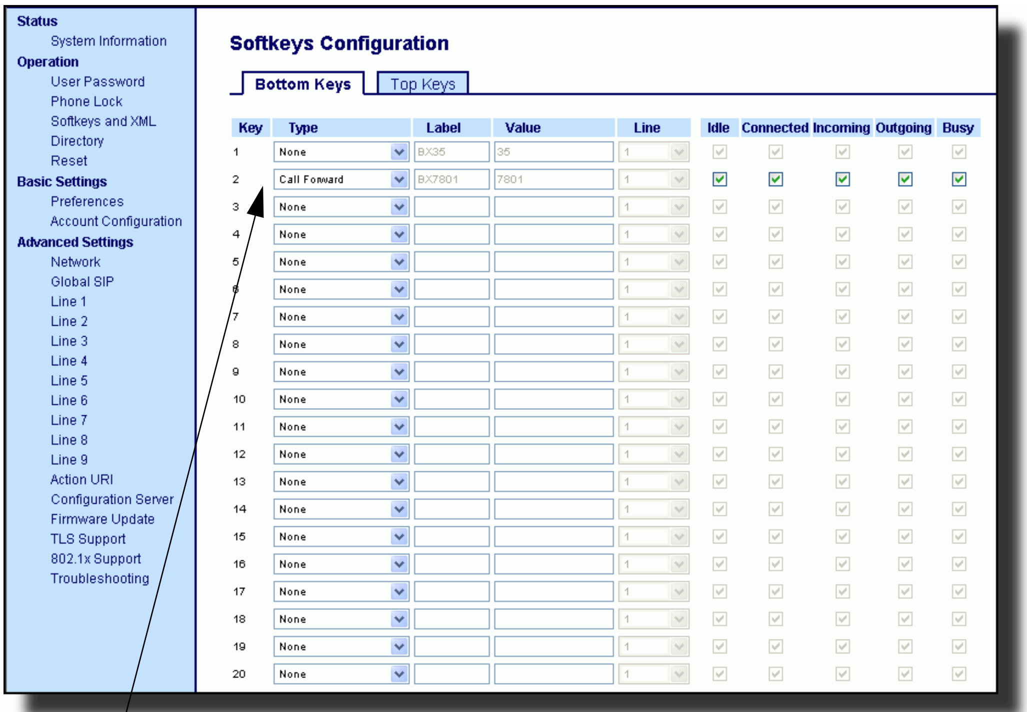
2. If there is only one account configured on the phone, then the mode setting is ignored and the phone behaves as if the mode was set to "Phone".

3. When configuring a CFWD mode (**All**, **Busy**, **No Answer**) for an account, you must configure a CFWD number for that mode in order for the mode to be enabled.

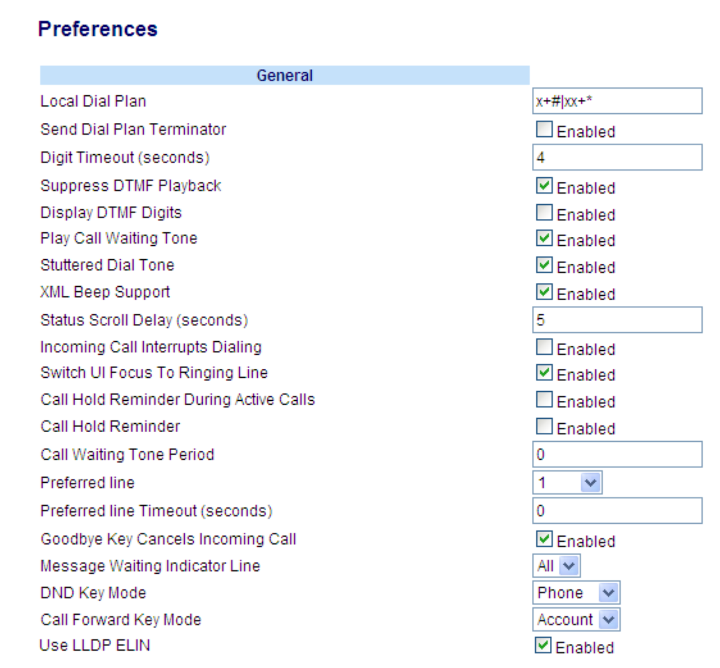
Use the following procedures to configure Call Forwarding on the IP phones.

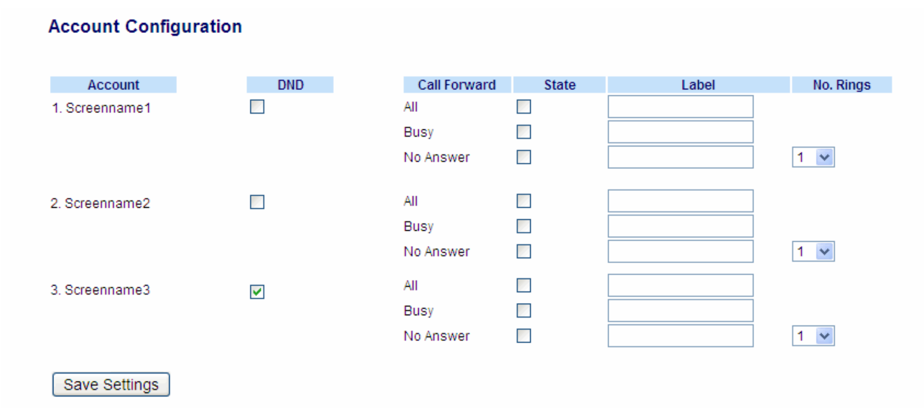
Configuration Files
For specific last call forwarding parameters you can set in the configuration files, see Appendix A, the sections, • "Priority Alert Settings" on page A-166. • "Softkey/Programmable Key/Feature Key/Expansion Module Key Parameters" on page A-202.

:

<table>
<tr><th colspan="2">Aastra Web UI</th></tr>
<tr><td>1</td><td>Click on Operation->Softkeys and XML;
or
Click on Operation->Programmable Keys;
or
Click on Operation->Expansion Module.


Softkey 2 configured for CFWD</td></tr>
<tr><td>2</td><td>Click Save Settings to save your changes.</td></tr>
</table>

	Aastra Web UI
3	Click on **Basic Settings->Preferences->General.**  **Note:** If there is no CFWD key configured on the phone or it is removed, you can still set the CFWD modes via the IP Phone UI at the path *Options->Call Forward.*

	Aastra Web UI
4	In the “**Call Forward Key Mode**” field, select a call forward mode to use on the phone. Valid values are: Account, Phone, Custom. Default is **Account**. • **account** The account mode allows you to configure CFWD on a per account basis. Pressing a configured CFWD key applies to the account in focus. • **phone** The Phone mode allows you to set the same CFWD configuration for all accounts (**All**, **Busy**, and/or **No Answer**). When you configure the initial account, the phone applies the configuration to all other accounts. (In the Aastra Web UI, only the account you configured is enabled. All other accounts are grayed out but set to the same configuration.) Using the Aastra Web UI, if you make changes to that initial account, the changes apply to all accounts on the phone. • **custom** The Custom mode allows you to configure CFWD for a specific account or all accounts. You can configure a specific state (**All**, **Busy**, and/or **No Answer**) for each account independently or all accounts. On 3-Line LCD phones, you can set all accounts to **ALL On** or **ALL Off**. On the 8 and 11-Line LCD phones, you can set all accounts to **All On**, **All Off**, or copy the configuration for the account in focus to all other accounts using a **CopytoAll** softkey. **Notes**: **1.** If there is no CFWD key configured on the phone or it is removed, you can still set the CFWD modes via the IP Phone UI at the path *Options->Call Forward.* **2.** If there is only one account configured on the phone, then the mode setting is ignored and the phone behaves as if the mode was set to “Phone”. **3.** When configuring a CFWD state (**All**, **Busy**, **No Answer**) for an account, you must configure a CFWD number for that state in order for the state to be enabled.
5	Click  to save your changes. The change takes effect immediately without a reboot.
6	Click on **Basic Settings->Account Configuration.** 

Aastra Web UI	
7	For each account, enable CFWD state by placing a check mark in one or more of the following "**State**" fields: • **All** • **Busy** • **No Answer** The "**All**" option forwards all incoming calls for this account to the specified phone number regardless of the state of the phone. The phone can be in the Busy or No Answer states, or can be in the idle state. The phone still forwards all calls to the specified number. The "**Busy**" option call forwards incoming calls only if the account is in the busy state. The calls are forwarded to the specified phone number. The "**No Answer**" option call forwards incoming calls only if the account rings but is not answered in the defined number of rings. The call gets forwarded to the specified number. **Note:** You can use the "**Busy**" and "**No Answer**" states together using different forwarding phone numbers. If these states are enabled for an account (the "**All**" state is disabled), and the phone is in the busy state when a call comes in, the phone can forward the call to the specified phone number (for example, voicemail). If there is no answer on the phone after the specified number of rings, the phone can forward the call to a different specified number, such as a cell phone number.
8	For each account, in the "**Number**" field, enter the phone number for which you want the incoming calls to forward to if the phone is in the specified state. If using the "**Account**" mode or "**Custom**" mode, you can enter different phone numbers for each account. **Notes**: **1.** If you selected "**Account**" mode in step 4, you can enable/disable each account or all accounts as applicable. You can enter different phone number for each enabled state. If you selected "**Custom**" mode, you can enable/disable each account or all accounts as applicable. You can enter different phone numbers for each enabled state. If you selected "**Phone**" mode, all accounts are set to the same CFWD configuration (**All**, **Busy**, and/or **No Answer**) as Account 1 on the phone. (In the Aastra Web UI, only Account 1 is enabled. All other accounts are grayed out but use the same configuration as Account 1.) Using the Aastra Web UI, if you make changes to Account 1, the changes apply to all accounts on the phone. Using the IP Phone UI, if you make changes to any other account other then Account 1, the changes also apply to all accounts on the phone. When enabling a CFWD state, you must specify a phone number for the phone to CFWD to. The number you specify applies to all accounts of the same mode. **2.** Number and name of accounts that display to this screen are dependant on the number and name of accounts configured on the phone. In the screen in step 6, Screenname1 is configured on Line 1, Screenname2 is configured on Line 2, and Screenname3 is configured on Line 3. The name for the account is dependant on the name specified for the "**Screen Name**" parameter at the path **Advanced Settings->LineN**. If you do not specify a value for the "Screen Name" parameter, the account name is based on the "**Phone Number**" parameter at the path **Advanced Settings->LineN**. If neither the "Screen Name" nor the "Phone Number" parameters are specified, the account name shows "1", "2", "3", etc. only.

	Aastra Web UI
9	For the **No Answer** state, in the "**No. Rings**" field, enter the number of times that the account rings before forwarding the call to the specified number. Valid values are 1 through 20. Default is 1. **Note:** When using the "**Account**" mode or "**Custom**" mode, you can enter a different number of rings for each account. If you use the Aastra Web UI to change the Call Forward Key Mode to "**Phone**", all accounts synchronize to Account 1.
10	Click Save Settings to save your changes. The change takes effect immediately without a reboot.

Using CFWD Modes via the IP Phone UI

If you enable/disable CFWD using the configuration files or the Aastra Web UI, you can use the CFWD screens that display to the IP Phone UI. You can access the CFWD parameters by pressing a configured CFWD key (if previously configured) OR by pressing *Options->Call Forward* on the phone's front panel.

The following procedure assumes you have already configured a CFWD key AND assumes there are three accounts configured on the phone.

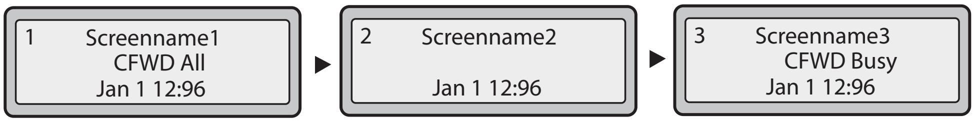
Notes:

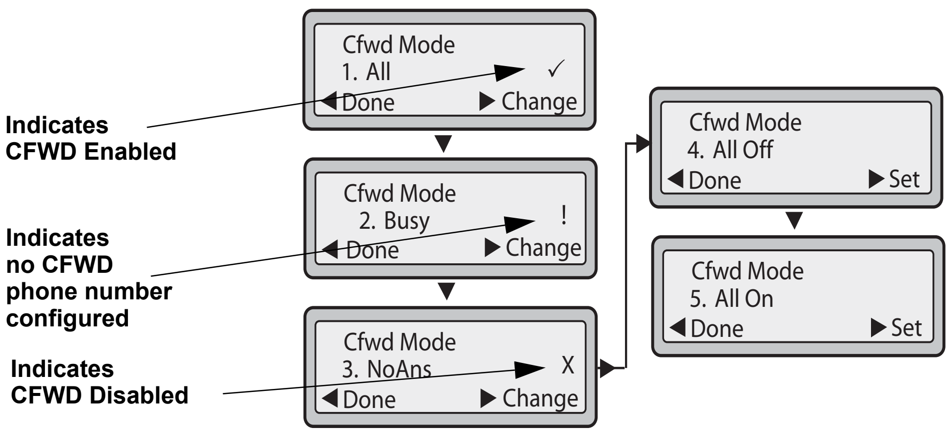
1. If there is no CFWD key configured on the phone or it is removed, you can still enable CFWD via the IP Phone UI at the path *Options->Call Forward*.

2. If there is only one account configured on the phone, then the mode setting is ignored and the phone behaves as if the mode was set to "Phone".

3. Using the Aastra Web UI, if you change the CFWD key mode to "**Phone**", all accounts synchronize to the current setting of Account 1.

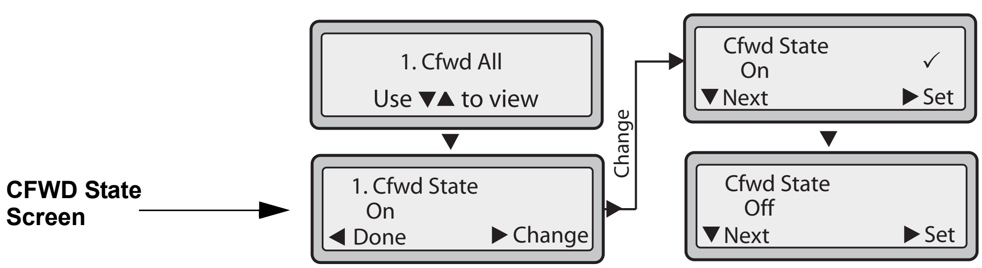
Aastra IP Phone UI

<table>
<tr><th>Step</th><th>Action</th></tr>
<tr><td colspan="2">CFWD in Account Mode (3-Line LCD Phones)</td></tr>
<tr><td>1</td><td>Use the RIGHT and LEFT arrow keys to scroll through each account.


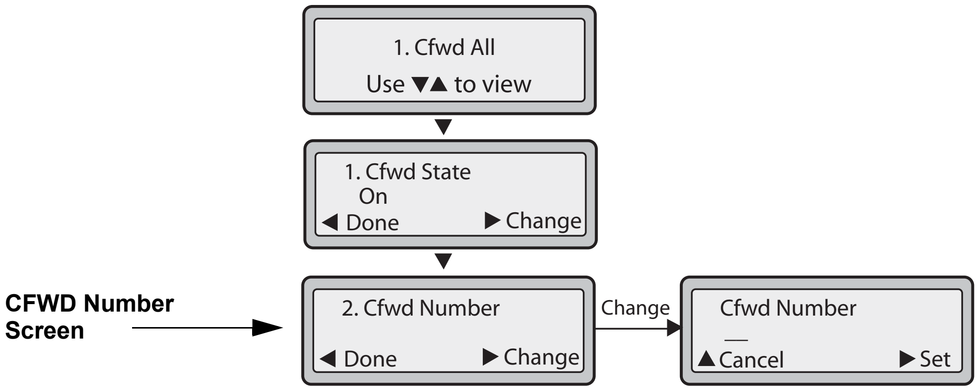
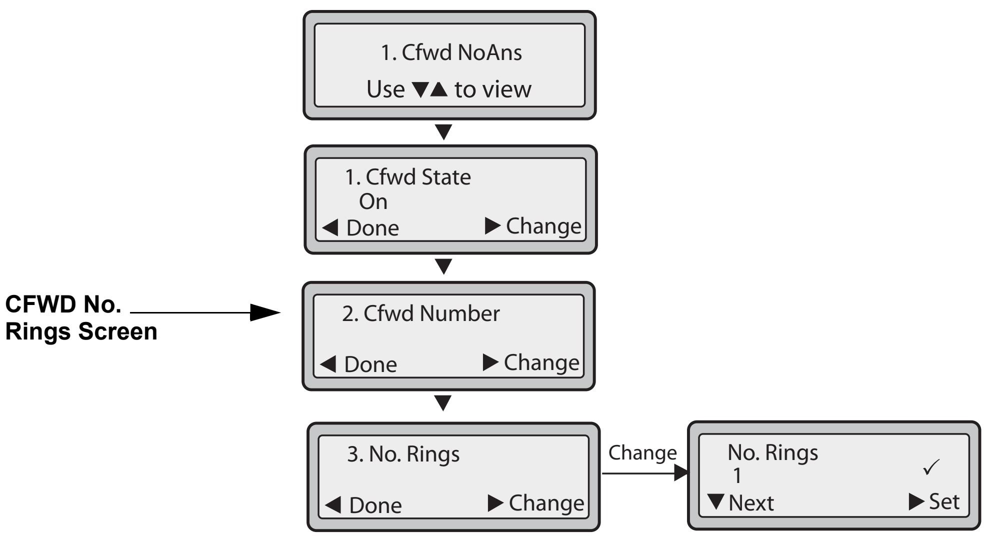
In the above example, Screenname1, Screenname2, and Screenname3, are three accounts configured on the phone. Screenname1 has “CFWD All” enabled, Screenname 2 has CFWD disabled as indicated by no message displayed, and Screenname3 has “CFWD Busy” enabled.</td></tr>
<tr><td>2</td><td>Press the Call Forward key. The Call Forward Mode screen displays. Use the UP and DOWN arrow keys to scroll through each state type.


In the above example, CFWD All is enabled as indicated by a checkmark (✓), CFWD Busy is enabled but no call forward phone number configured as indicated by a !, and CFWD NoAns is disabled, as indicated by an X.</td></tr>
</table>

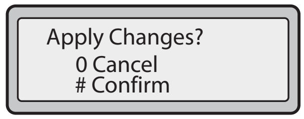
Aastra IP Phone UI

Step	Action
3	Select a state for the account(s) in focus using the **UP** and **DOWN** arrow keys. You can enable/disable any or all of the following states for an account: • **All** - Enables **CFWD All** for an account and forwards all incoming calls for that account, to the specified number. • **Busy** - Enables **CFWD Busy** for an account and forwards incoming calls to a specified number if DND has been enabled for that account OR if the account is currently engaged in another call. • **No Answer** - Enables **CFWD NoAns** for an account and forwards incoming calls to a specified number if the call has not been answered for the specified number of rings. **Note:** If **CFWD All** AND **CFWD Busy** AND **CFWD NoAns** are all enabled (and/or if the account has DND enabled), the **CFWD All** settings take precedence over **CFWD Busy** and **CFWD NoAns.** You can also use the following keys if required: **All Off** - Disables all CFWD states for the current account in focus. **All On** - Enables all CFWD states for the current account in focus.
4	Press the **CHANGE** key for the state you selected in step 3. Scroll to the CFWD State screen. This displays the current state of the mode you selected. In the following example, the CFWD All state is ON. 
5	Press the **CHANGE** key in the CFWD State screen. Press **NEXT** to toggle the state of the CFWD mode ON or OFF. In the example in Step 4, you press **NEXT** to change the option to **OFF**.
6	Press the **SET** key to save the change.

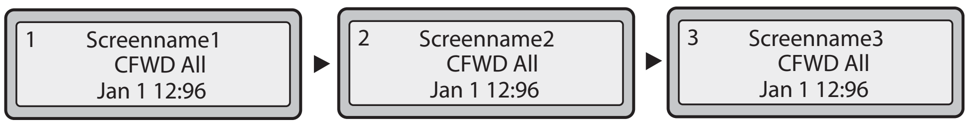
Aastra IP Phone UI

Step	Action
7	In the CFWD State screen, press the **DOWN** arrow key to scroll to the CFWD Number screen and press **CHANGE**. 
8	Enter a phone number to apply to the current state in focus. When the phone is in the state you specified, and a call comes into the phone, it forwards the call to the number you specify.
9	Press **SET** to save the change.
10	For the **CFWD No Answer** state, In the CFWD Number screen, press the **DOWN** arrow key to scroll to the CFWD No. Rings screen and press **CHANGE**. 
11	Press the **NEXT** key to select the number of rings to apply to the phone for call forwarding incoming calls. Valid values are 1 to 20. Default is 1. When the phone receives an incoming call, and call forward is configured on the phone, the phone rings the number of times you specify in the No. Rings screen, and then forwards the call if there is no answer.
12	Press **SET** to save the change.

Aastra IP Phone UI

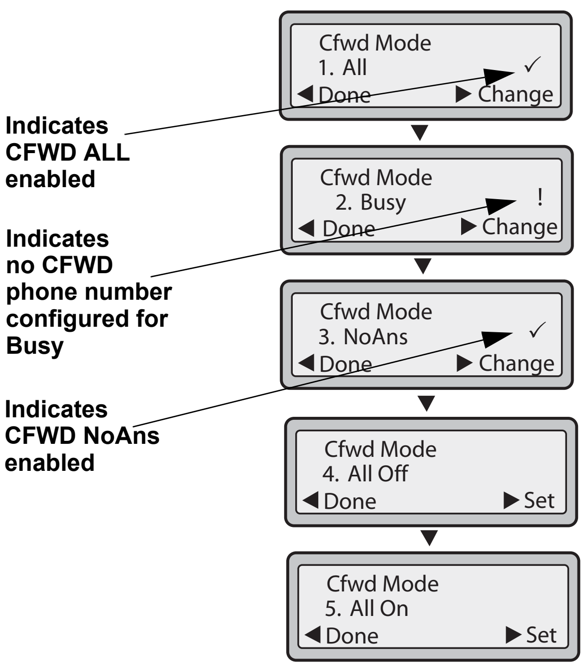
Step	Action
13	Press **DONE** to save CFWD All Number, CFWD All State, CFWD Busy Number, CFWD Busy State, CFWD No Answer Number, CFWD No Answer State, CFWD No Answer Rings. Each time you press **DONE**, the following “Apply Changes” screen displays. 
14	Press **#** to confirm the change(s) each time the “Apply Changes” screen displays. All changes are saved to the phone.

<table>
<tr><th colspan="2">Aastra IP Phone UI</th></tr>
<tr><th>Step</th><th>Action</th></tr>
<tr><td colspan="2">CFWD in Phone Mode (3-Line LCD Phones)</td></tr>
<tr><td>1</td><td>
Use the RIGHT and LEFT arrow keys to scroll through each account.



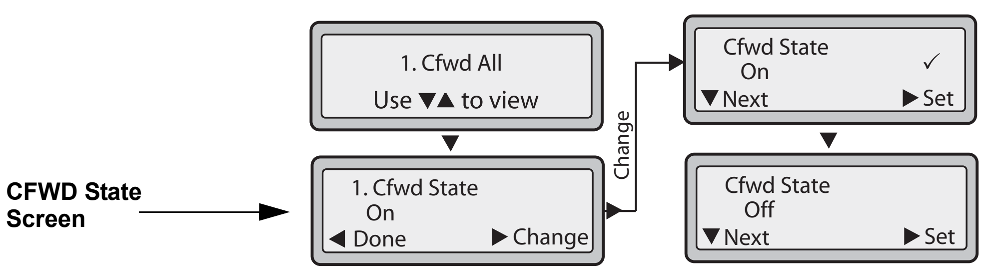
In the above example, Screenname1, Screenname2, and Screenname3, are three accounts configured on the phone. All three accounts have CFWD enabled as indicated by the “CFWD All” message.

Note: In ”Phone” mode, when you change the call forward configuration for an account, the change applies to all accounts.
</td></tr>
<tr><td>2</td><td>
Press the Call Forward key. The Call Forward Mode screen displays. Use the UP and DOWN arrow keys to scroll through each state type.



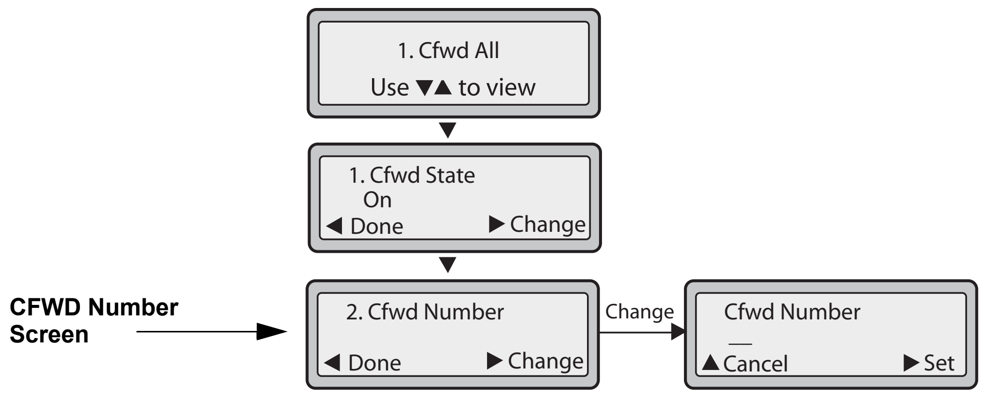
In the above example, the account has CFWD All and CFWD NoAns enabled as indicated by a checkmark (✓). The CFWD Busy setting is enabled but no call forward phone number is configured as indicated by a !
</td></tr>
</table>

Aastra IP Phone UI

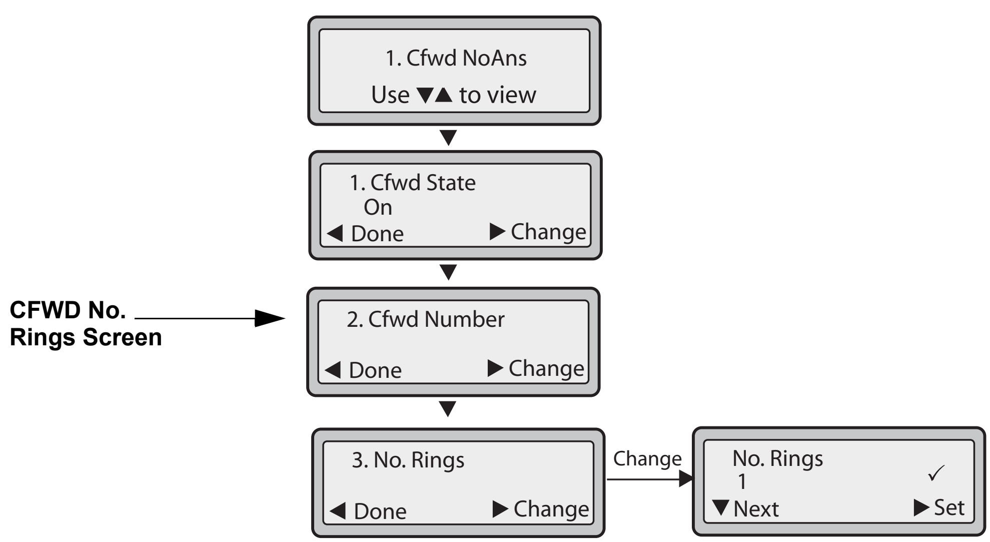
Step	Action
3	Select a state using the **UP** and **DOWN** arrow keys. You can enable/disable a specific account on the phone with any or all of the following states. However, the configuration you set will apply to all accounts on the phone. • **All** - Enables **CFWD All** on the phone and forwards all incoming calls to the specified number. • **Busy** - Enables **CFWD Busy** on the phone and forwards incoming calls to a specified number if DND has been enabled OR if the phone is currently engaged in another call. • **No Answer** - Enables **CFWD NoAns** on the phone and forwards incoming calls to a specified number if the call has not been answered for the specified number of rings. **Note:** If **CFWD All** AND **CFWD Busy** AND **CFWD NoAns** are all enabled (and/or if the account has DND enabled), the **CFWD All** settings take precedence over **CFWD Busy** and **CFWD NoAns.** You can also use the following keys if required: **All Off** - Disables all CFWD states for the phone. **All On** - Enables all CFWD states for the phone. **Note: In "Phone" mode, the initial configuration you set for an account applies to all the accounts on the phone.**
4	Press the **CHANGE** key for the mode you selected in step 2. Scroll to the CFWD State screen. This displays the current state of the mode you selected. In the following example, the CFWD All state is ON. 
5	Press the **CHANGE** key in the CFWD State screen. Press **NEXT** to toggle the state of the CFWD state ON or OFF. In the example in Step 4, you press **NEXT** to change the option to **OFF**.
6	Press the **SET** key to save the change.

Aastra IP Phone UI

<table>
<tr><th>Step</th><th>Action</th></tr>
<tr><td>7</td><td>In the CFWD State screen, press the DOWN arrow key to scroll to the CFWD Number screen and press CHANGE.

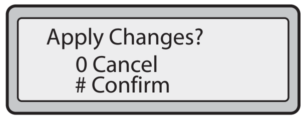


</td></tr>
<tr><td>8</td><td>Enter a phone number to apply to the current state in focus. When the phone is in the state you specified, and a call comes into the phone, it forwards the call to the number you specify.</td></tr>
<tr><td>9</td><td>Press SET to save the change.</td></tr>
<tr><td>10</td><td>For the CFWD No Answer state, in the CFWD Number screen, press the DOWN arrow key to scroll to the CFWD No. Rings screen and press CHANGE.



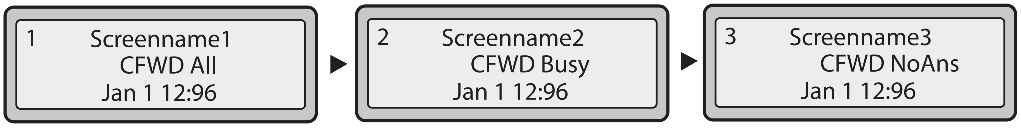
</td></tr>
<tr><td>11</td><td>Press the NEXT key to select the number of rings to apply to the phone for call forwarding incoming calls. Valid values are 1 to 20. Default is 1.

When the phone receives an incoming call, and call forward is configured on the phone, the phone rings the number of times you specify in the No. Rings screen, and then forwards the call.</td></tr>
<tr><td>12</td><td>Press SET to save the change.</td></tr>
</table>

Aastra IP Phone UI	
Step	**Action**
13	Press **DONE** to save CFWD All Number, CFWD All State, CFWD Busy Number, CFWD Busy State, CFWD No Answer Number, CFWD No Answer State, CFWD No Answer Rings. Each time you press **DONE**, the following “Apply Changes” screen displays. 
14	Press **#** to confirm the change(s) each time the “Apply Changes” screen displays. All the same changes are saved to all accounts on the phone.

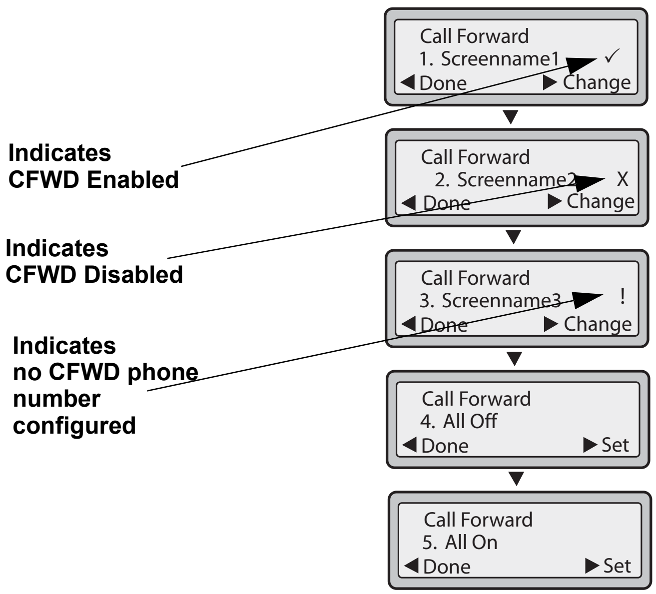
Aastra IP Phone UI

Step	Action
CFWD in Custom Mode (3-Line LCD Phones)	
1	Use the **RIGHT** and **LEFT** arrow keys to scroll through each account.



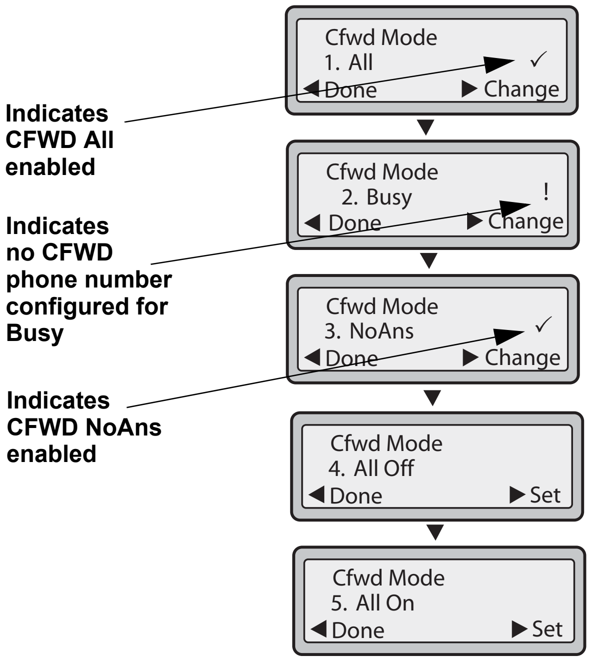
In the above example, Screenname1, Screenname2, and Screenname3, are three accounts configured on the phone. All three accounts have CFWD enabled as indicated by the "**CFWD All**", "**CFWD Busy**", and "**CFWD NoAns**" messages.

Press the **Call Forward** key. The CFWD Account screens display. Use the **UP** and **DOWN** arrow keys to scroll through each account.

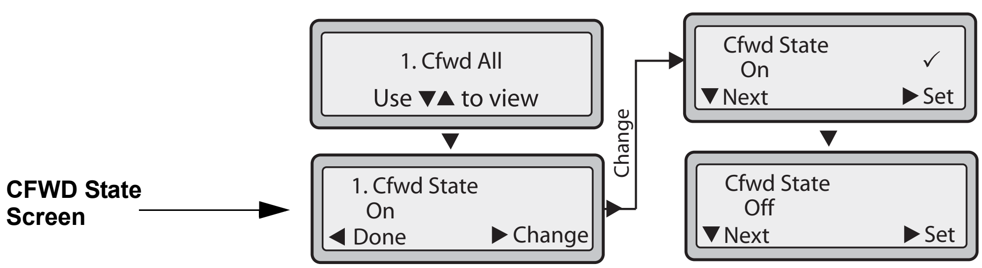


In the above example, Screenname1 has one or more CFWD states enabled as indicated by a checkmark (✓). Screenname2 has one or more CFWD states disabled as indicated by an **X**. Screenname3 has one or more CFWD states configured but a specific state has no call forward phone number configured as indicated by a **!**. Items 4 and 5 allow you to disable or enable CFWD on all accounts, respectively.

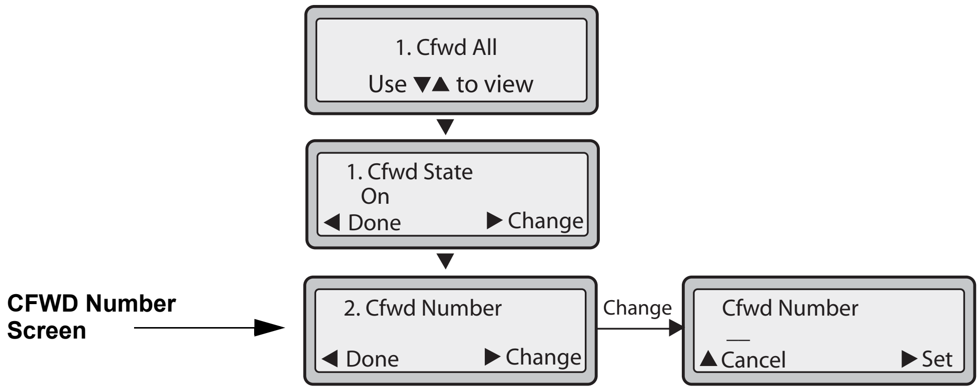
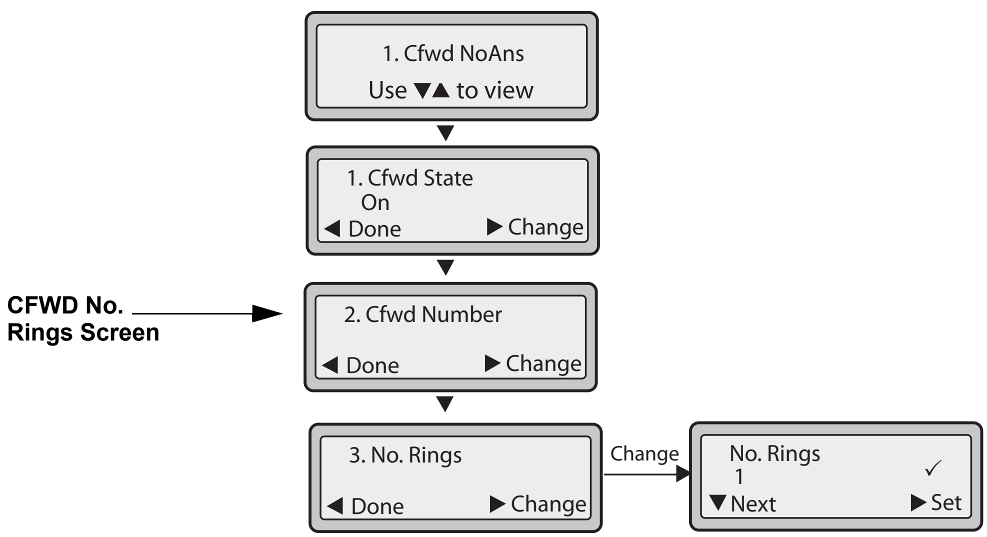
Aastra IP Phone UI

Step	Action
2	Select an account to configure. Press the **CHANGE** key. The CFWD Mode screen displays. Use the **UP** and **DOWN** arrow keys to scroll through each state screen.  In the above example, the account has **CFWD All** and **CFWD NoAns** enabled as indicated by a checkmark (✓). The **CFWD Busy** setting is enabled for the account but has no call forward phone number configured as indicated by a **!**.

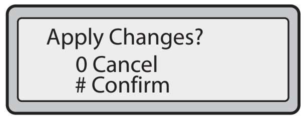
Aastra IP Phone UI

Step	Action
3	Select a state for the account(s) in focus using the **UP** and **DOWN** arrow keys. You can enable/disable any or all of the following states for a specific account or for all accounts (with individual configurations): • **All** - Enables **CFWD All** for an account and forwards all incoming calls for that account, to the specified number. The phone number can be different between accounts. • **Busy** - Enables **CFWD Busy** for an account and forwards incoming calls to a specified number if DND has been enabled for that account OR if that account is currently engaged in another call. The phone number can be different between accounts. • **No Answer** - Enables **CFWD NoAns** for an account and forwards incoming calls to a specified number if the call has not been answered for a specified number of rings. The phone number can be different between accounts. **Note:** If **CFWD All** AND **CFWD Busy** AND **CFWD NoAns** are all enabled (and/or if the account has DND enabled), the **CFWD All** settings take precedence over **CFWD Busy** and **CFWD NoAns.** You can also use the following keys if required: **All Off** - Disables all CFWD states for the current account in focus or all accounts. **All On** - Enables all CFWD states for the current account in focus or all accounts.
4	Press the **CHANGE** key for the mode you selected in step 2. Scroll to the CFWD State screen. This displays the current state of the mode you selected. In the following example, the CFWD All state is ON. 
5	Press the **CHANGE** key in the CFWD State screen. Press **NEXT** to toggle the state of the CFWD state ON or OFF. In the example in Step 4, you press **NEXT** to change the option to **OFF**.
6	Press the **SET** key to save the change.

Aastra IP Phone UI

Step	Action
7	In the CFWD State screen, press the **DOWN** arrow key to scroll to the CFWD Number screen and press **CHANGE**. 
8	Enter a phone number to apply to the current state in focus. When the phone is in the state you specified, and a call comes into the phone, it forwards the call to the number you specify.
9	Press **SET** to save the change.
10	For the **CFWD No Answer** state, in the CFWD Number screen, press the **DOWN** arrow key to scroll to the CFWD No. Rings screen and press **CHANGE**. 
11	Press the **NEXT** key to select the number of rings to apply to the phone for call forwarding incoming calls. Valid values are 1 to 20. Default is 1. When the phone receives an incoming call, and call forward is configured on the phone, the phone rings the number of times you specify in the No. Rings screen, and then forwards the call.
12	Press **SET** to save the change.

Aastra IP Phone UI

Step	Action
13	Press **DONE** to save CFWD All Number, CFWD All State, CFWD Busy Number, CFWD Busy State, CFWD No Answer Number, CFWD No Answer State, CFWD No Answer Rings. Each time you press **DONE**, the following “Apply Changes” screen displays. 
14	Press **#** to confirm the change(s) each time the “Apply Changes” screen displays. All changes are saved to the phone for all accounts.

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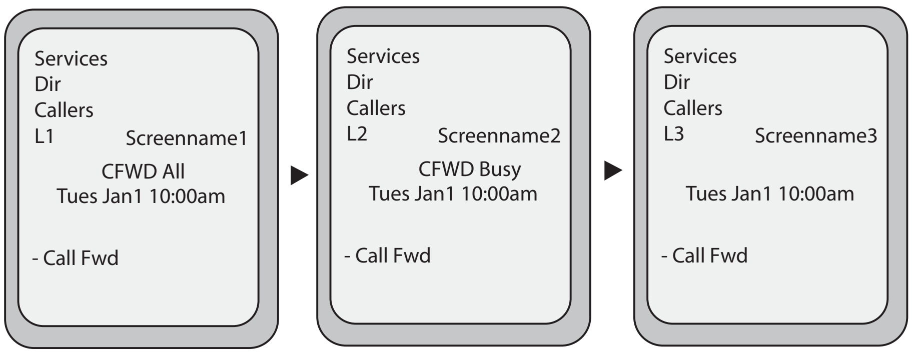
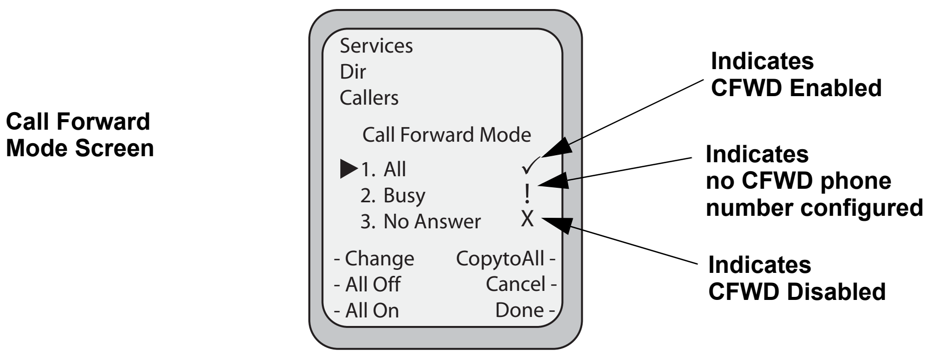
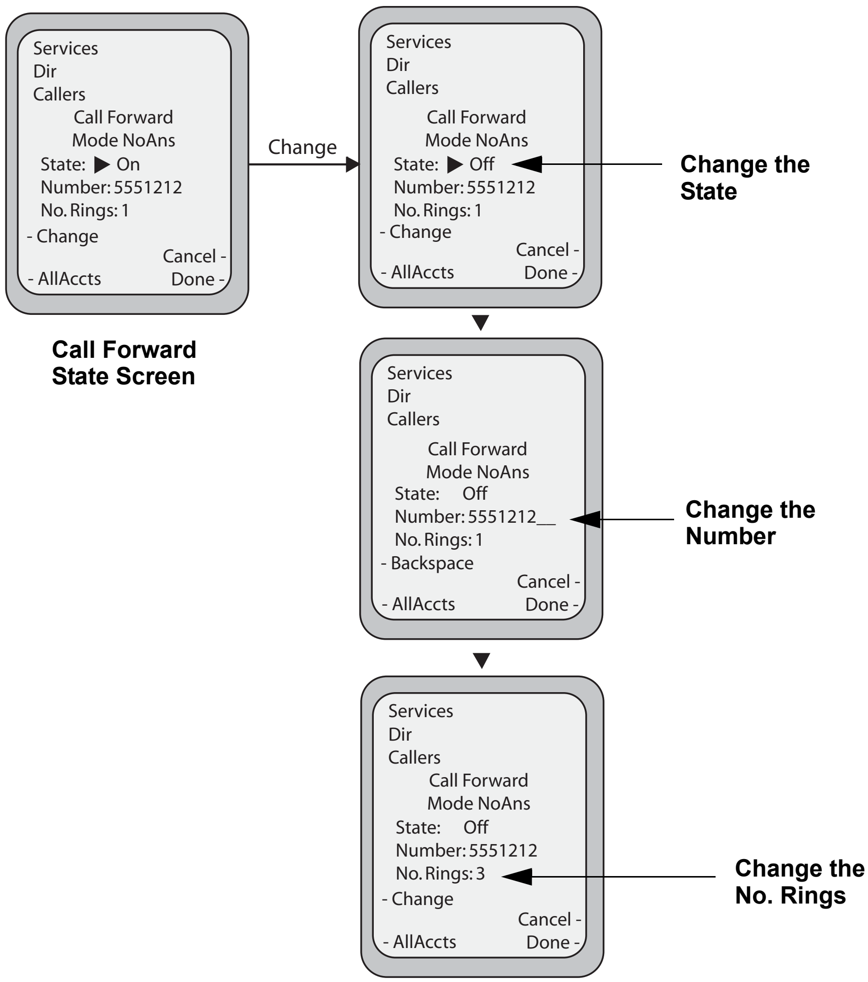
Aastra IP Phone UI	
Step	**Action**
CFWD in Account Mode (8 and 11-Line LCD Phones)	
1	Use the **RIGHT** and **LEFT** arrow keys to scroll through each account.  In the above example, Screenname1, Screenname2, and Screenname3, are three accounts configured on the phone. Screenname1 has "**CFWD All**" enabled, Screenname2 has "**CFWD Busy**" enabled, and Screenname3 has CFWD disabled as indicated by no message displayed.
2	Press the **Call Forward** key. The Call Forward Mode screen displays for the account you selected. Use the **UP** and **DOWN** arrow keys to scroll through each state type.  In the above example, **CFWD All** is enabled as indicated by a checkmark (✓), **CFWD Busy** is enabled but no call forward phone number is configured as indicated by a **!**, and **CFWD NoAns** is disabled as indicated by an **X.**

Aastra IP Phone UI

Step	Action
3	Select a state for the account(s) in focus using the **UP** and **DOWN** arrow keys. You can enable/disable any or all of the following states for an account: • **All** - Enables **CFWD All** for an account and forwards all incoming calls for that account, to the specified number. • **Busy** - Enables **CFWD Busy** for an account and forwards incoming calls to a specified number if DND has been enabled for that account OR if the account is currently engaged in another call. • **No Answer** - Enables **CFWD NoAns** for an account and forwards incoming calls to a specified number if the call has not been answered for the specified number of rings. **Note:** If **CFWD All** AND **CFWD Busy** AND **CFWD NoAns** are all enabled (and/or if the account has DND enabled), the **CFWD All** settings take precedence over **CFWD Busy** and **CFWD NoAns.** You can also use the following keys if required: **All Off Key**- Disables all CFWD states for the current account in focus. **All On Key** - Enables all CFWD states for the current account in focus. **CopytoAll Key** - Copies the call forward phone number and state of the Call Forward mode (All, Busy, No Answer) in focus to every Call Forward mode of that account. For example, if you have the cursor pointing at the "**All**" state and it is enabled and has a call forward phone number configured, pressing the **CopytoAll** Key enables the **Busy** state and the **NoAns** state and assigns the same phone number to both states. **Cancel Key** - Cancels any configuration you may have made without saving. To cancel a configuration, you must press this **CANCEL** key before pressing the **DONE** key.

Aastra IP Phone UI

<table>
<tr><th>Step</th><th>Action</th></tr>
<tr><td>4</td><td>For the CFWD No Answer state, press the CHANGE key for the mode you selected in step 2. This displays the Call Forward State screen. In the following example, the CFWD All state is ON.


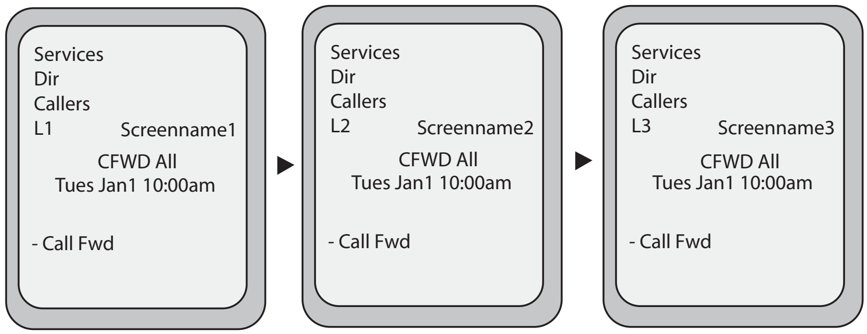
Note: You can press the AllAccts key to copy the settings in the current Call Forward Settings screen for a specific call forward mode, to every account on the phone. Every account will have the same settings for that call forward mode.</td></tr>
<tr><td>5</td><td>Press the CHANGE key in the CFWD State screen. With the cursor in the “State” field, toggle the state ON and OFF by pressing the CHANGE key.

Note: You can press the AllAccts key to copy the settings in the current Call Forward Settings screen for a specific call forward mode, to every account on the phone. Every account will have the same settings for that call forward mode.</td></tr>
</table>

Aastra IP Phone UI	
Step	**Action**
6	Use the **DOWN** arrow key to scroll to the "**Number**" field. Enter a phone number to apply to the current state in focus. When the phone is in the state you specified, and a call comes into the phone, it forwards the call to the number you specify. Use the **BACKSPACE** key if required to delete characters. **Note:** You can press the **AllAccts** key to copy the settings in the current Call Forward Settings screen for a specific call forward mode, to every account on the phone. Every account will have the same settings for that call forward mode.
7	Use the **DOWN** arrow key to scroll to the "**No. Rings**" field. Press the **CHANGE** key to select the number of rings to apply to the phone for call forwarding incoming calls. Valid values are 1 to 20. Default is 1. When the phone receives an incoming call, and call forward is configured on the phone, the phone rings the number of times you specify in the No. Rings screen, and then forwards the call. **Note:** You can press the **AllAccts** key to copy the settings in the current Call Forward Settings screen for a specific call forward mode, to every account on the phone. Every account will have the same settings for that call forward mode.
8	Press **DONE** in the CFWD State Screen to save all changes. Press **DONE** in the CFWD Mode Screen to save all changes.

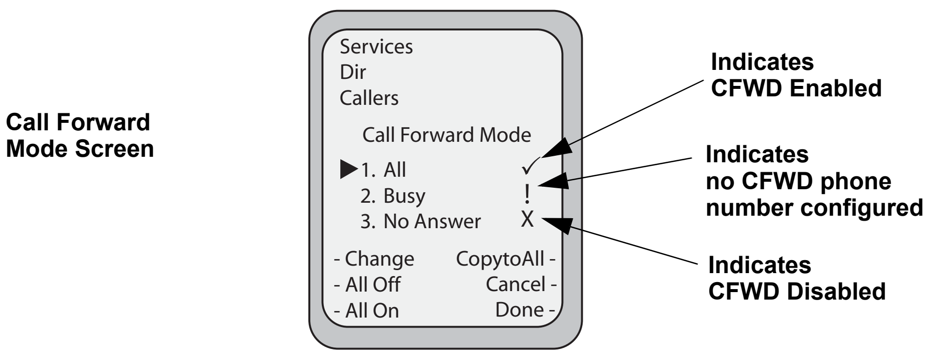
Aastra IP Phone UI

<table>
<tr><th>Step</th><th>Action</th></tr>
<tr><td colspan="2">CFWD in Phone Mode (8 and 11-Line LCD Phones)</td></tr>
<tr><td>1</td><td>Use the RIGHT and LEFT arrow keys to scroll through each account.



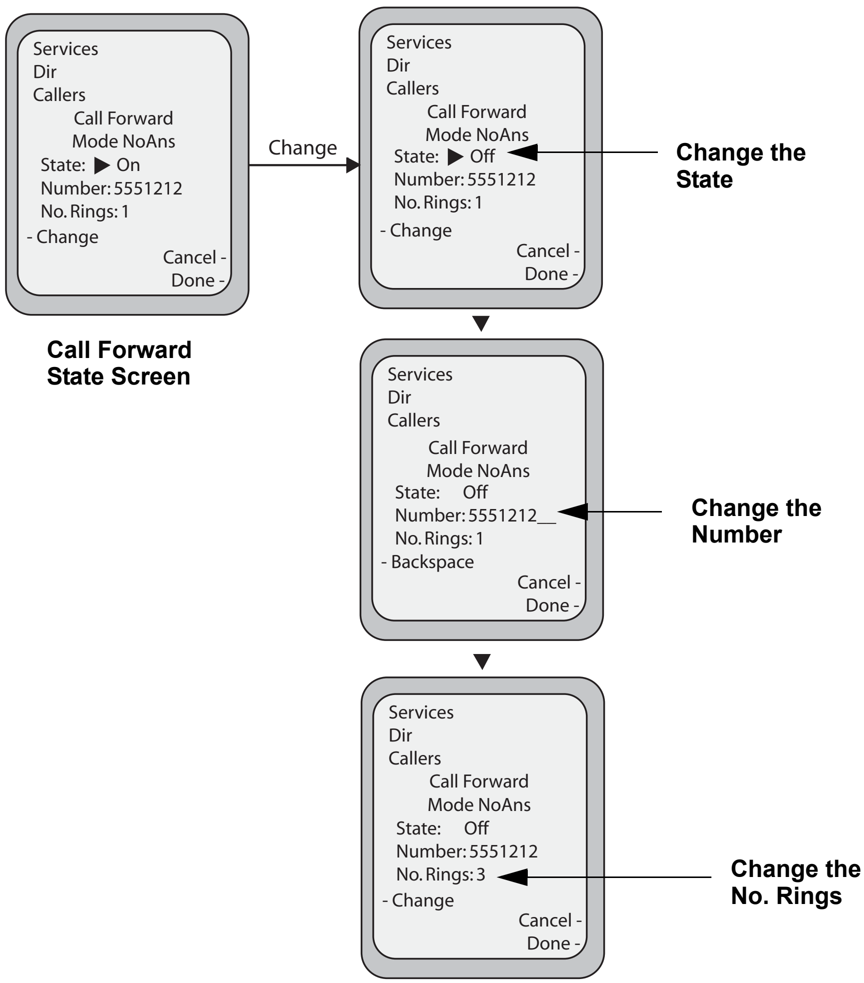
In the above example, Screenname1, Screenname2, and Screenname3, are three accounts configured on the phone. Screenname1, 2 and 3 have “CFWD All” enabled.

Note: In “Phone” mode, the initial configuration you set for an account applies to all the accounts on the phone.</td></tr>
<tr><td>2</td><td>Press the Call Forward key. The Call Forward Mode screen displays. Use the UP and DOWN arrow keys to scroll through each state type.



In the above example, CFWD All is enabled as indicated by a checkmark (✓), CFWD Busy is enabled but no call forward phone number is configured as indicated by a !, and CFWD No Answer is disabled, as indicated by an X.</td></tr>
</table>

Aastra IP Phone UI	
Step	**Action**
3	Select a state for the phone using the **UP** and **DOWN** arrow keys. You can enable/disable a specific account on the phone with any or all of the following states. However, the configuration you set will apply to all accounts on the phone. • **All** - Enables **CFWD All** on the phone and forwards all incoming calls to the specified number. • **Busy** - Enables **CFWD Busy** on the phone and forwards incoming calls to a specified number if DND has been enabled OR if the phone is currently engaged in another call. • **No Answer** - Enables **CFWD NoAns** on the phone and forwards incoming calls to a specified number if the call has not been answered for the specified number of rings. **Note:** If **CFWD All** AND **CFWD Busy** AND **CFWD NoAns** are all enabled (and/or if the account has DND enabled), the **CFWD All** settings take precedence over **CFWD Busy** and **CFWD NoAns.** You can also use the following keys if required: **All Off Key**- Disables all CFWD modes. **All On Key** - Disables all CFWD modes. **CopytoAll Key** - Copies the call forward phone number and state of the Call Forward mode (All, Busy, No Answer) in focus to every Call Forward mode. For example, if you have the cursor pointing at the "**All**" state and it is enabled and has a call forward phone number configured, pressing the **CopytoAll** Key enables the **Busy** state and the **NoAns** state and assigns the same phone number to both states. **Cancel Key** - Cancels any configuration you may have made without saving. To cancel a configuration, you must press this **CANCEL** key before pressing the **DONE** key.

Aastra IP Phone UI	
Step	**Action**
4	In the **CFWD No Answer** state, press the **CHANGE** key for the mode you selected in step 2. This displays the Call Forward State screen. In the following example, the CFWD All state is ON. 
5	Press the **CHANGE** key in the CFWD State screen. With the cursor in the "**State**" field, toggle the state ON and OFF by pressing the **CHANGE** key.
6	Use the **DOWN** arrow key to scroll to the "**Number**" field. Enter a phone number to apply to the current state in focus. When the phone is in the state you specified, and a call comes into the phone, it forwards the call to the number you specify. Use the **BACKSPACE** key if required to delete characters.
7	Use the **DOWN** arrow key to scroll to the "**No. Rings**" field. Press the **CHANGE** key to select the number of rings to apply to the phone for call forwarding incoming calls. Valid values are 1 to 20. Default is 1. When the phone receives an incoming call, and call forward is configured on the phone, the phone rings the number of times you specify in the No. Rings screen, and then forwards the call.

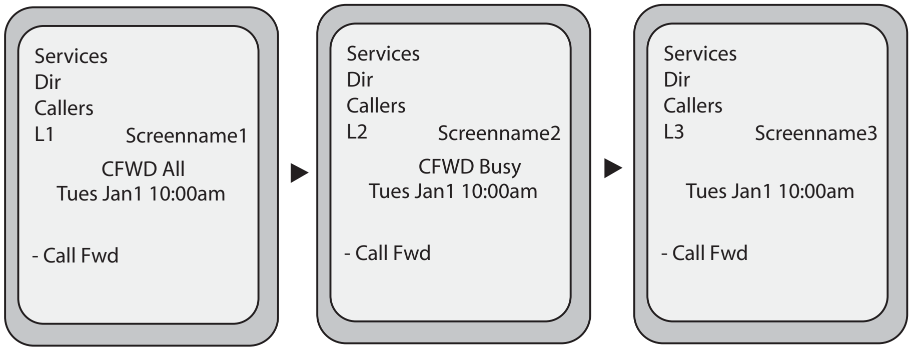
Aastra IP Phone UI

Step	Action
8	Press **DONE** in the CFWD State Screen to save all changes. Press **DONE** in the CFWD Mode Screen to save all changes. **Note: In "Phone" mode, the configuration applies to all the accounts on the phone.**

Aastra IP Phone UI

<table>
<tr><th>Step</th><th>Action</th></tr>
<tr><td colspan="2">CFWD in Custom Mode (8 and 11-Line LCD Phones)</td></tr>
<tr><td>1</td><td>

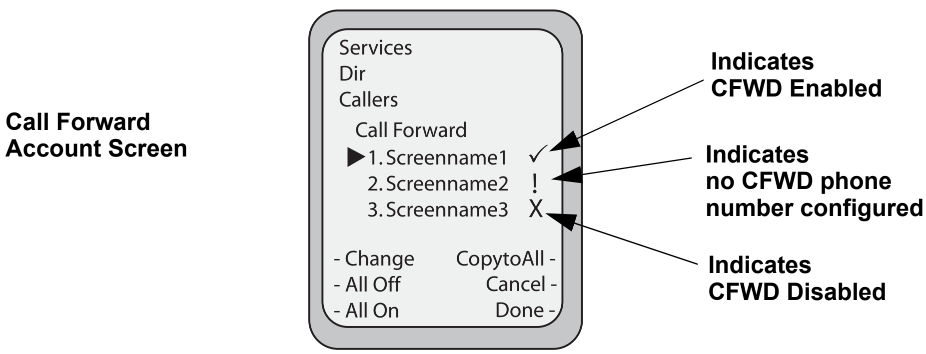
Use the **RIGHT** and **LEFT** arrow keys to scroll through each account.



In the above example, Screenname1, Screenname2, and Screenname3, are three accounts configured on the phone. Screenname1 has “**CFWD All**” enabled, Screenname2 has “**CFWD Busy**” enabled, and Screenname3 has CFWD disabled as indicated by no message displayed.

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

With the account in focus on the IP Phone UI, press the **Call Forward** key. The Call Forward Account screen displays which lists all the accounts on the phone. Use the **UP** and **DOWN** arrow keys to scroll through each account.



In the above example, Screenname1 has one or more CFWD states enabled as indicated by a checkmark (✓), Screenname 2 has one or more CFWD states enabled but a specific state has no call forward phone number configured as indicated by a **!**, and Screenname3 has one or more CFWD states disabled as indicated by an **X.**

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

Aastra IP Phone UI

<table>
<tr><th>Step</th><th>Action</th></tr>
<tr><td>3</td><td>Select an account using the UP and DOWN arrow keys.

You can also use the following keys if required:

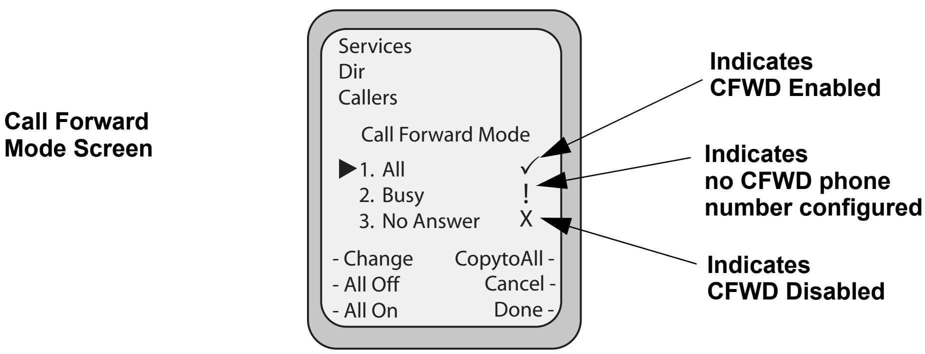
All Off Key- Disables CFWD for all accounts on the phone.

All On Key - Enables CFWD for all accounts on the phone.

CopytoAll Key - Copies all settings for the account you select, to all other accounts on the Call Forward Account screen. For example, if you have the cursor pointing at Screenname1, and you press the CopytoAll key, all of the CFWD settings for Screenname1 are copied to Screenname2 and Screenname3.

Cancel Key - Cancels any configuration you may have made without saving. To cancel a configuration, you must press this CANCEL key before pressing the DONE key.</td></tr>
<tr><td>4</td><td>After selecting an account, press CHANGE.

The Call Forward Mode screen displays for the account you selected. Use the UP and DOWN arrow keys to scroll through each state type.

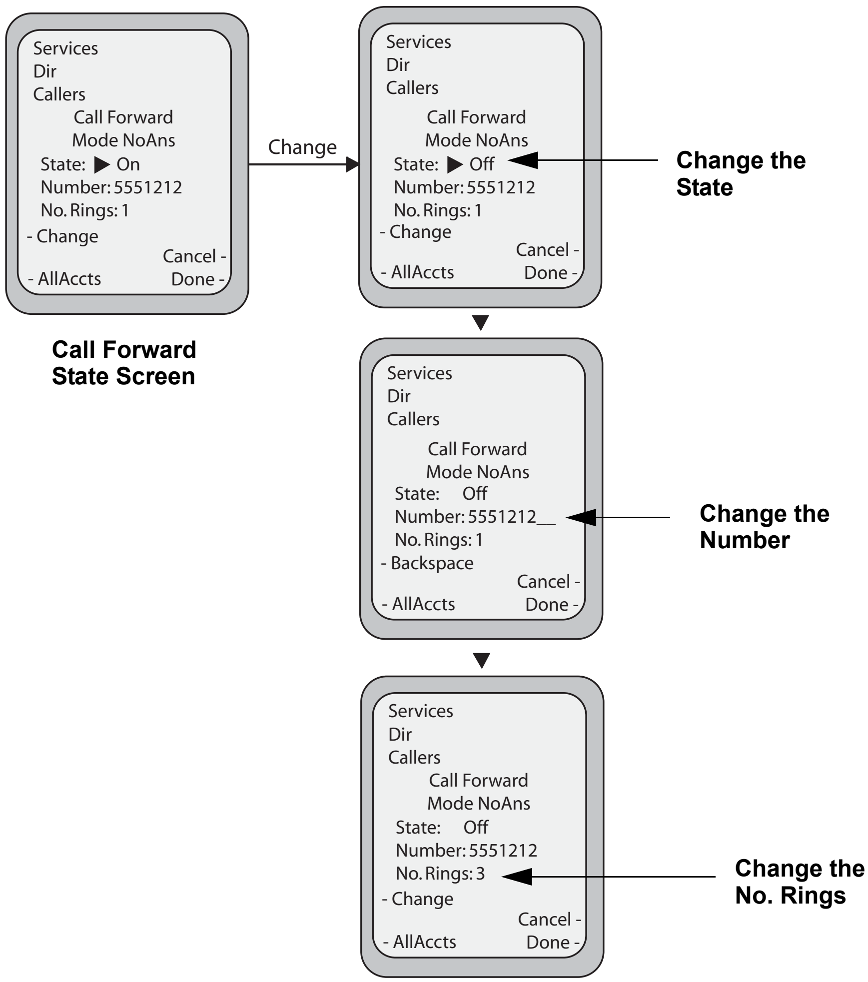


In the above example, CFWD All is enabled as indicated by a checkmark (✓), CFWD Busy is enabled but no call forward phone number is configured as indicated by a !, and CFWD No Answer is disabled, as indicated by an X.</td></tr>
</table>

Aastra IP Phone UI	
Step	**Action**
5	Select a state for the selected account(s) using the **UP** and **DOWN** arrow keys. You can enable/disable any or all of the following states for a specific account or for all accounts (with individual configurations): • **All** - Enables **CFWD All** for an account and forwards all incoming calls for that account, to the specified number. The phone number can be different between accounts. • **Busy** - Enables **CFWD Busy** for an account and forwards incoming calls to a specified number if DND has been enabled for that account OR if that account is currently engaged in another call. The phone number can be different between accounts. • **No Answer** - Enables **CFWD NoAns** for an account and forwards incoming calls to a specified number if the call has not been answered for a specified number of rings. The phone number can be different between accounts. **Note:** If **CFWD All** AND **CFWD Busy** AND **CFWD NoAns** are all enabled (and/or if the account has DND enabled), the **CFWD All** settings take precedence over **CFWD Busy** and **CFWD NoAns.** You can also use the following keys if required: **All Off Key**- Disables all CFWD states for the selected account. **All On Key** - Enables all CFWD states for the selected account. **CopytoAll Key** - Copies the call forward phone number and state of the selected Call Forward mode (All, Busy, No Answer) to every Call Forward mode of that account. For example, if you have the cursor pointing at the "**All**" state and it is enabled and has a call forward phone number configured, pressing the **CopytoAll** Key enables the **Busy** state and the **NoAns** state and assigns the same phone number to both states. **Cancel Key** - Cancels any configuration you may have made without saving. To cancel a configuration, you must press this **CANCEL** key before pressing the **DONE** key.

Aastra IP Phone UI

<table>
<tr><th>Step</th><th>Action</th></tr>
<tr><td>6</td><td>Press the CHANGE key for the mode you selected in step 4. This displays the Call Forward State screen. In the following example, the CFWD All state is ON.



Call Forward State Screen

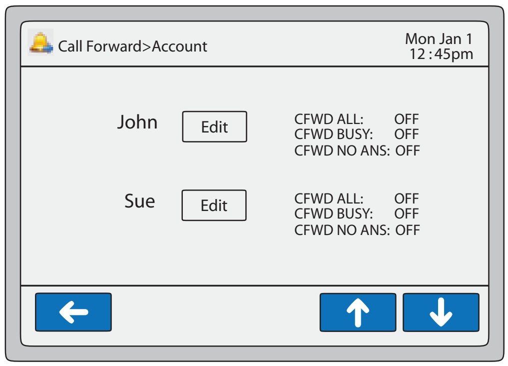
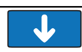
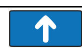
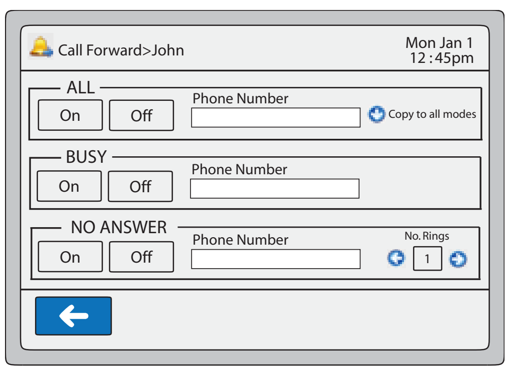
Note: You can press the AllAccts key to copy the settings in the current Call Forward Settings screen for a specific call forward mode, to every account on the phone. Every account will have the same settings for that call forward mode.</td></tr>
<tr><td>7</td><td>Press the CHANGE key in the CFWD State screen. With the cursor in the “State” field, toggle the state ON and OFF by pressing the CHANGE key.

Note: You can press the AllAccts key to copy the settings in the current Call Forward Settings screen for a specific call forward mode, to every account on the phone. Every account will have the same settings for that call forward mode.</td></tr>
</table>

Aastra IP Phone UI

Step	Action
8	Use the **DOWN** arrow key to scroll to the "**Number**" field. Enter a phone number to apply to the current state in focus. When the phone is in the state you specified, and a call comes into the phone, it forwards the call to the number you specify. Use the **BACKSPACE** key if required to delete characters. **Note:** You can press the **AllAccts** key to copy the settings in the current Call Forward Settings screen for a specific call forward mode, to every account on the phone. Every account will have the same settings for that call forward mode.
9	Use the **DOWN** arrow key to scroll to the "**No. Rings**" field. Press the **CHANGE** key to select the number of rings to apply to the phone for call forwarding incoming calls. Valid values are 1 to 20. Default is 1. When the phone receives an incoming call, and call forward is configured on the phone, the phone rings the number of times you specify in the No. Rings screen, and then forwards the call. **Note:** You can press the **AllAccts** key to copy the settings in the current Call Forward Settings screen for a specific call forward mode, to every account on the phone. Every account will have the same settings for that call forward mode.
10	Press **DONE** in the CFWD State screen to save all changes. Press **DONE** in the CFWD Mode screen to save all changes. Press **DONE** in the CFWD Account screen to save all changes.

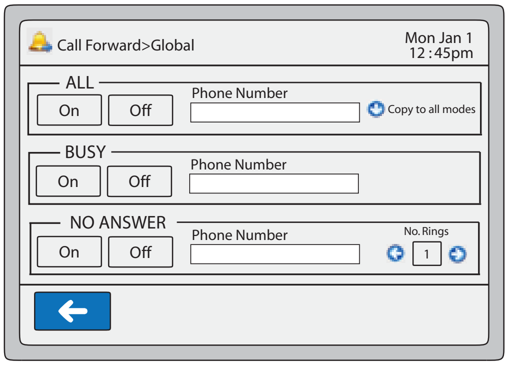
Aastra IP Phone UI

Step	Action
CFWD in Account Mode (6739i)	
1	Press the **<Services>** Key, then press the <**Call Forward**> button. or Press a pre-configured **Call Forward** softkey. The following screen displays.  **Call Forward Account Mode**
2	Use the  and  keys to scroll the list of accounts and select an account to apply CFWD.
3	Press **<Edit>**. The screen displays for the account you selected. The following example shows the CFWD screen for John's account. 
4	Select the parameter you want to enable (ALL, BUSY, and/or NO ANSWER). Selecting the ALL parameter call forwards all calls to another number regardless of the state of your phone.
5	Press the <**On**> button to turn the CFWD parameter ON or the <**Off**> button to turn the CFWD parameter OFF.
6	Press the "**Phone Number**" field. A pop-up keyboard displays.
7	Using the keyboard, enter the phone number that your phone will call forward to and press <**Enter**>.

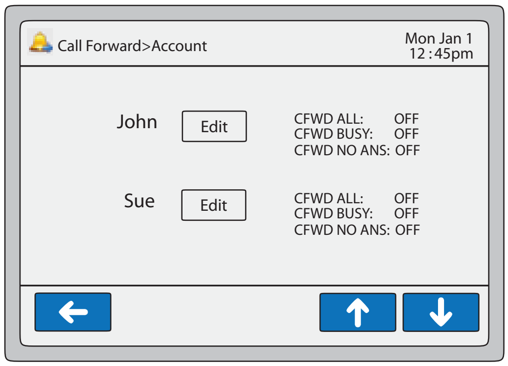
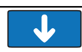
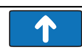
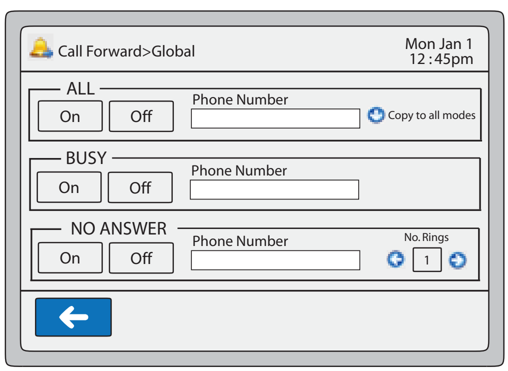
Aastra IP Phone UI

Step	Action
8	If you enter a phone number for the ALL parameter, you can press the **Copy to all modes** to automatically copy the same phone number to the BUSY and NO ANSWER phone number fields.
9	For the NO ANSWER parameter, in the "**No. Rings"** field, select the number of rings you want your phone to perform before call forwarding a call. **Note:** If **All** AND **Busy** AND **NoAns** are all enabled, and if the account has DND enabled, the **CFWD All** settings take precedence over **CFWD Busy** and **CFWD NoAns**.
10	Press the to return to the previous menu or press the to return to the idle screen.

Aastra IP Phone UI

Step	Action
CFWD in Phone Mode (6739i)	
1	Press the **<Services>** Key, then press the <**Call Forward**> button. or Press a pre-configured **Call Forward** softkey. The following screen displays.  **Call Forward Phone Mode**
2	Select the parameter you want to enable (ALL, BUSY, and/or NO ANSWER). Selecting the ALL parameter call forwards all calls to another number regardless of the state of your phone.
3	Press the <**On**> button to turn the CFWD parameter ON or the <**Off**> button to turn the CFWD parameter OFF.
4	Press the "**Phone Number**" field. A pop-up keyboard displays.
5	Using the keyboard, enter the phone number that your phone will call forward to and press <**Enter**>.
6	If you enter a phone number for the ALL parameter, you can press the **Copy to all modes** to automatically copy the same phone number to the BUSY and NO ANSWER phone number fields.
7	For the NO ANSWER parameter, in the "**No. Rings"** field, select the number of rings you want your phone to perform before call forwarding a call. **Note:** If All AND Busy AND NoAns are all enabled, and if the account has DND enabled, the CFWD All settings take precedence over CFWD Busy and CFWD NoAns.
8	Press the to return to the previous menu or press the to return to the idle screen.

Aastra IP Phone UI

Step	Action
CFWD in Custom Mode (6739i)	
1	Press the **<Services>** Key, then press the <**Call Forward**> button. or Press a pre-configured **Call Forward** softkey. The following screen displays.  **Call Forward Custom Mode**
2	Use the  and  keys to scroll the list of accounts and select an account to apply CFWD.
3	Press **<Edit>**. The screen displays for the account you selected. The following example shows the CFWD screen for John's account. 
4	Select the parameter you want to enable (ALL, BUSY, and/or NO ANSWER). Selecting the ALL parameter call forwards all calls to another number regardless of the state of your phone.
5	Press the <**On**> button to turn the CFWD parameter ON or the <**Off**> button to turn the CFWD parameter OFF.
6	Press the "**Phone Number**" field. A pop-up keyboard displays.
7	Using the keyboard, enter the phone number that your phone will call forward to and press <**Enter**>.

Aastra IP Phone UI

Step	Action
8	If you enter a phone number for the ALL parameter, you can press the **Copy to all modes** to automatically copy the same phone number to the BUSY and NO ANSWER phone number fields.
9	For the NO ANSWER parameter, in the "**No. Rings"** field, select the number of rings you want your phone to perform before call forwarding a call.
10	Press the to return to the previous menu.
11	To configure CFWD for additional accounts, repeat steps 2 through 9. **Note:** If All AND Busy AND NoAns are all enabled, and if the account has DND enabled, the CFWD All settings take precedence over CFWD Busy and CFWD NoAns.
12	Press the to return to the previous menu or press the to return to the idle screen.

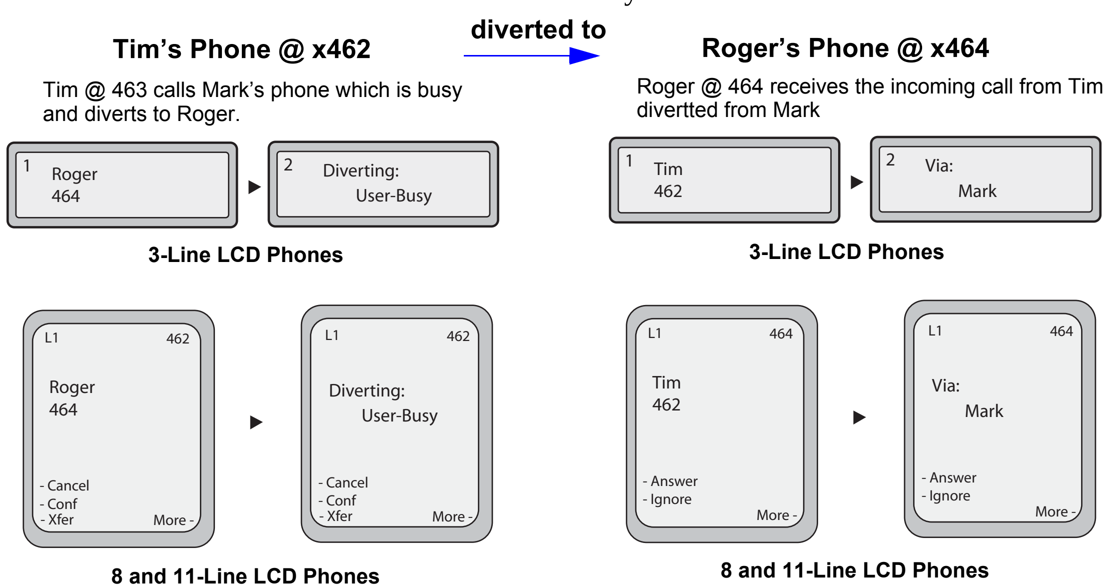
SIP Phone Diversion Display

When an outgoing call from the phone is being diverted to another destination (i.e., via call forward), the phone displays the Caller ID (phone number and/or caller name) of the new destination and the reason for the call diversion. Similarly, at the new destination, the Caller ID of the original call destination now displays.

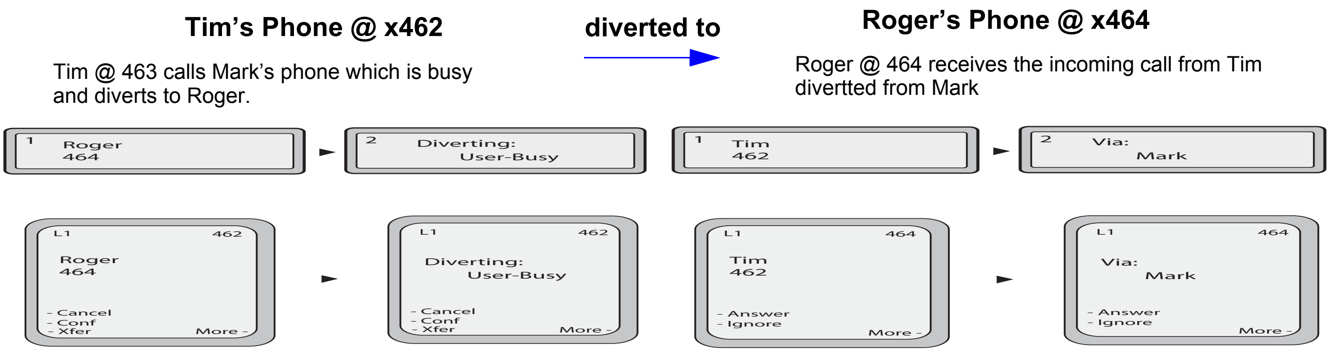
Call Diversion Example	
1	Tim calls Mark at x400.
2	Mark’s phone is busy.
3	Mark’s phone diverts the incoming call to another destination (Roger @ x 464).
4	Tim’s phone displays name and extension of where the call is being diverted to and reason for diverting the call. The screen scrolls between Screen 1 and Screen 2.
5	Roger’s phone accepts the call and displays the name and number of the phone the incoming call (Tim) and the name (or number) of the original destination (Mark). The screens scroll between Screen 1 and Screen 2..

See examples of the phones’ LCD below.

The diversion LCD screens scroll every 3 seconds.



For the 6739i:



Note: If proxy servers exist in the network, it is possible that multiple diversions can take place on the phones. When multiple diversion headers are returned in a single 302 response back to the originating phone, the phone that originated the call (i.e., Tim's phone in above example) displays the URI of the newest (first encountered) Diversion header, but displays the REASON of the oldest (last encountered) Diversion Header. The phone that receives the diverted call (i.e., Roger's phone in example above) displays the information of the oldest diverted call (last encountered).

You can enable or disable this feature on a global or per-line basis using the configuration files only.

Configuring SIP Diversion Display on the Phone

Use the following procedures to configure SIP diversion display on the IP phones.

Configuration Files
For specific parameters you can set in the configuration files, see Appendix A, the section, "SIP Diversion Display" on page A-173.

Limitations

- The diversion header assumes that the the ID of the 'diverted' caller is passed in a URI style manner.
- This feature relies on the server supporting and generating the Diversion header; the phone does not generate the header itself.
- Diversion header parameters such as counter, limit, privacy, screen, and extension are not recognized or supported by the phone. However, they are still passed along during the diversion process.

Displaying Call Destination for Incoming Calls

The IP Phones allow an Administrator to enable and disable the call destination name in the “TO” header of the INVITE message for incoming calls. When this feature is enabled, the call destination name displays on the LCD of the phone. This allows the user to easily determine the intended destination of an incoming call.

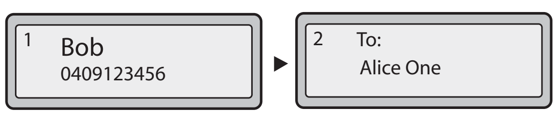
Behavior of the Phone

When this feature is enabled, the phone behaves as follows:

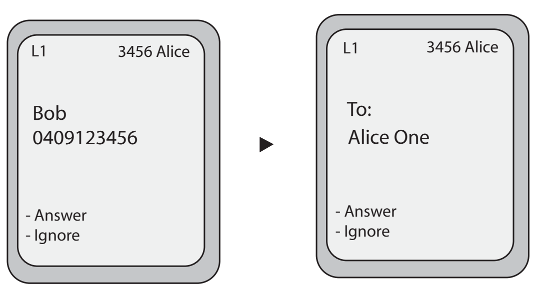
IF	THEN
A value exists for the Display Name field in the “TO” header of the INVITE message for incoming calls...	the phone displays the call destination name.
Display Name field is empty...	the phone uses the name specified for the “Screen name 1” parameter.
“Screen name 1” parameter is empty...	the phone uses the name specified for the “Display Name” parameter.
“Display Name” parameter is empty...	the phone uses the name specified for the “SIP User Name” parameter.
“SIP User Name” parameter is empty...	the phone uses the name specified for the “Call Destination Number” parameter.

The call destination information displays on multiple screens that scroll every 3 seconds. The following example shows call destination information on the phones.

Call Destination Enabled on the Phone

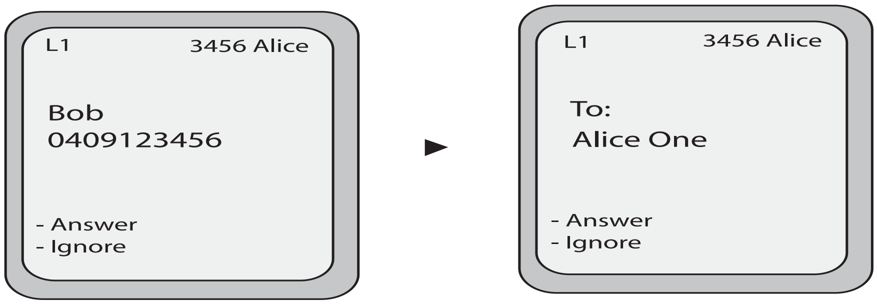


Incoming Phone UI on 3-Line LCD Phones

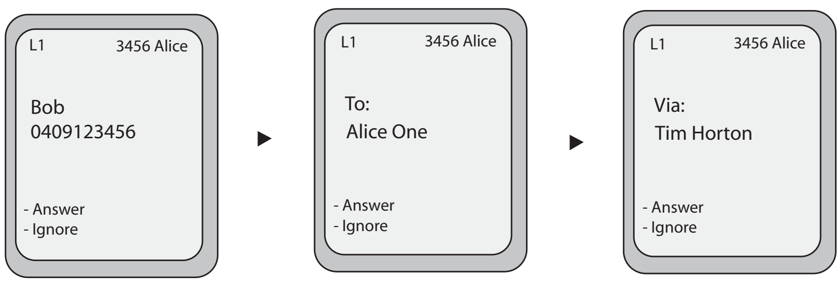


Incoming Phone UI on 8 and 11-Line LCD Phones

Call Destination Enabled on 6739i

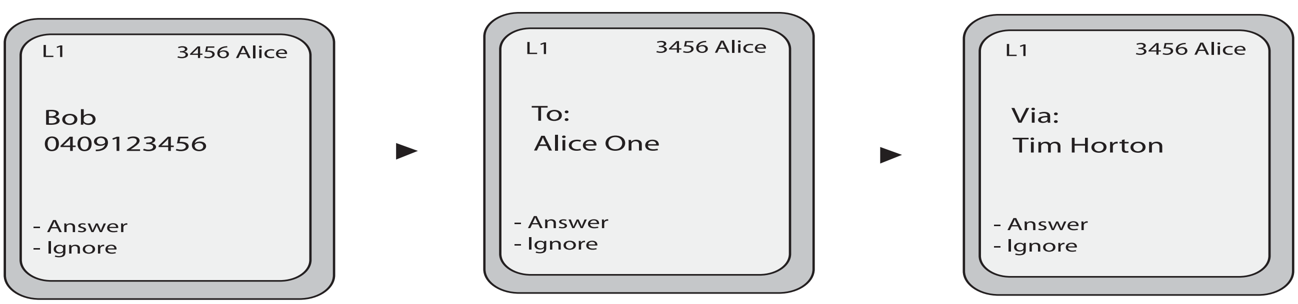


Since Call Diversion is enabled on the phone by default, the following example shows the behavior when call destination is also enabled.



Page 1 Caller Info | **Page 2 Call Destination** | **Page 3 Call Diversion**

For 6739i:



Page 1 Caller Info | **Page 2 Call Destination** | **Page 3 Call Diversion**

Note: As shown above, when both call diversion and call destination are enabled, the formation displays to the phone's screens in the following order:

Screen 1 Caller info
Screen 2 Call destination
Screen 3 Call diversion

Configuring the Display of Call Destination for Incoming Calls

Use the following procedures to configure the display of call destination for incoming calls on the IP Phones.

Configuration Files

For the specific parameter you can set in the configuration files, see Appendix A, the section, "Display of Call Destination for Incoming Calls" on page A-174.

Limitations

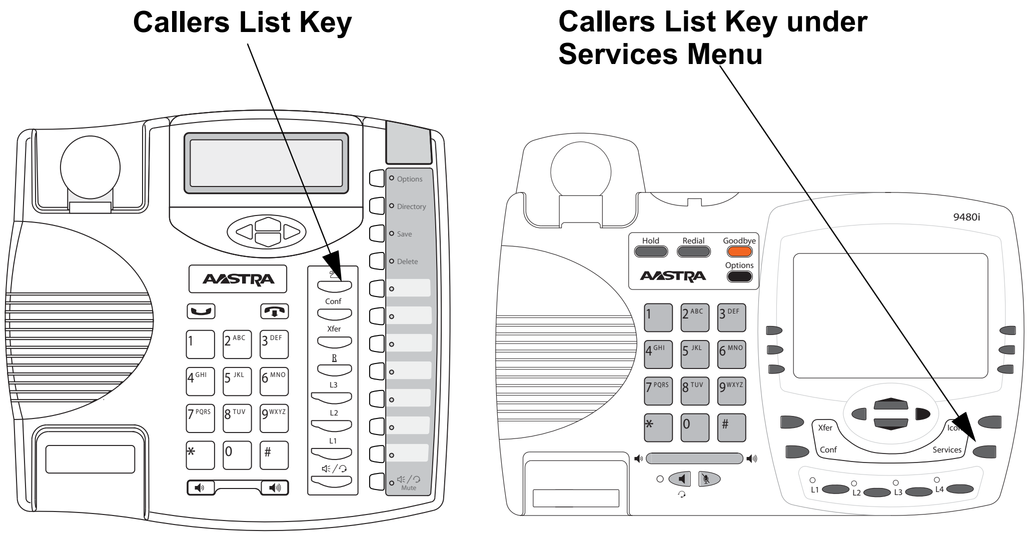
The following are limitations of this feature:

- Any call destination name exceeding the screen length is truncated by the phone.
- The CT cordless handsets do not support this feature.
- Page scrolling every 3 seconds is hard-coded and not configurable.

Callers List

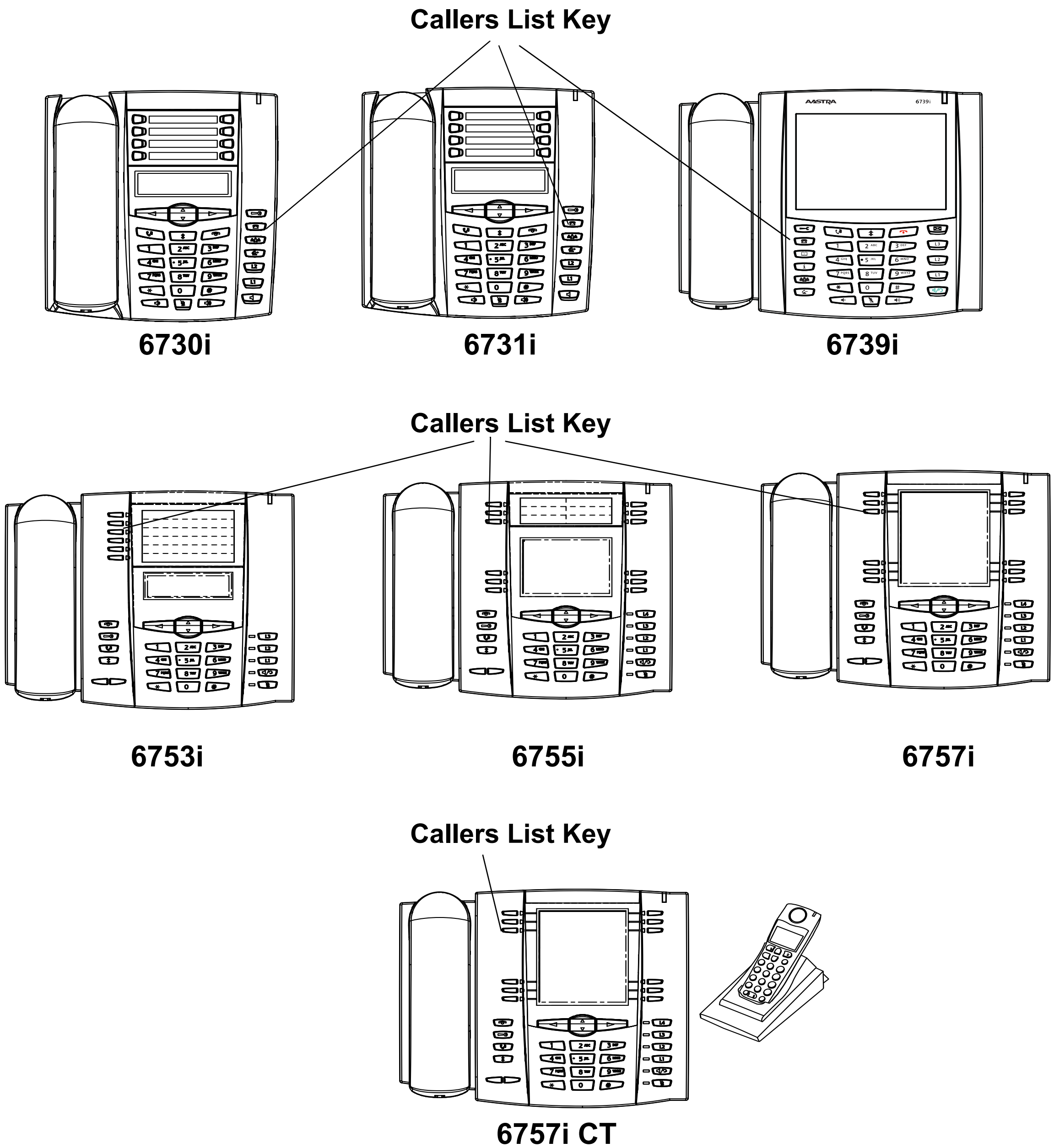
The IP phones have a "Callers List" feature that store the name, phone number, and incremental calls, for each call received by the phone.

The following illustrations show the default location of the Callers List Key on each type of phone model.



9143i

9480i/9480i CT

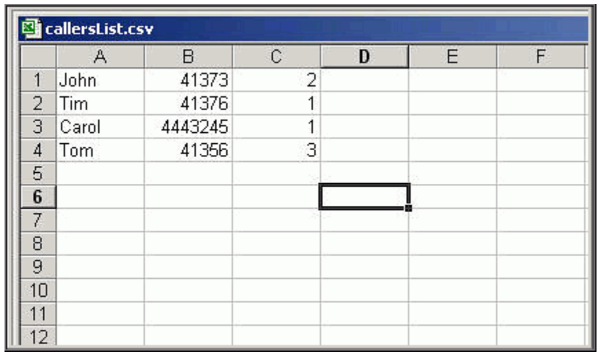


You can enable and disable the Callers List feature using the configuration files. When disabled, the Callers List does not display on the IP phone UI and the Caller List key is ignored when pressed.

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 6757i and 6757i 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-138.

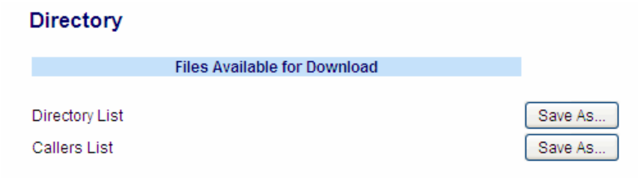
Using the Callers List

Use the following procedure to access and use the Callers List.

IP Phone UI	
Step	**Action**
1	***On all phones:*** Press the **"Callers List"** key on the phone to enter the Callers List. ***On the 9480i, 9480i CT:*** Press the **"Services"** key on the phone, and then select "**Callers List**" from the list of options.
2	Use the r and s to scroll through the line items in the Callers List. To the left of a line item, a b 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 dial a displayed entry from the Callers List, pick up the handset, press the **d / f** handsfree key, or press a line key. **Note**: After selecting a number from the Callers List, you can edit the number if necessary.
4	To delete all entries in the Callers list, press the 3 **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 3 **Delete** key.
5	To cancel a delete function, press the r or the s **Scroll** keys.
6	To save a line item to a programmable key for speed dialing, press the 3 **Save** key and enter the line number at the "Save to?" prompt that is already configured for speed dialing at a programmable key.
7	To exit the Callers List, press the "**Callers List**" key again or the "**Goodbye"** key.
For the 6739i:	
1	Press the key on the phone to enter the Callers List.
2	Use the and keys to navigate to the Caller record for which you want to call. A GREEN checkmark displays next to the entries that were answered. A RED exclamation point displays next to the entries that were missed calls.
3	To dial from the Callers List, press the Caller record and then press Dial. The phone automatically dials the number from the record you selected. **Note**: After selecting a number from the Callers List, you can edit the number if necessary.
4	To delete a record from the Callers List, press the Caller record and then press Delete.
5	To save a Caller record to the directory, press the Caller record and then press the Add to Directory. button. The Directory displays. Check that all of the information from the Callers record was entered correctly in the Directory List entry.

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 may have a Callers List key and a Services key (as a hard key or softkey/ programmable key) depending on your model phone. 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-139.

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.

Note: The phones also include a "Missed-Calls-Message" field in a "message summary" event of a SIP NOTIFY message.

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-144.

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.

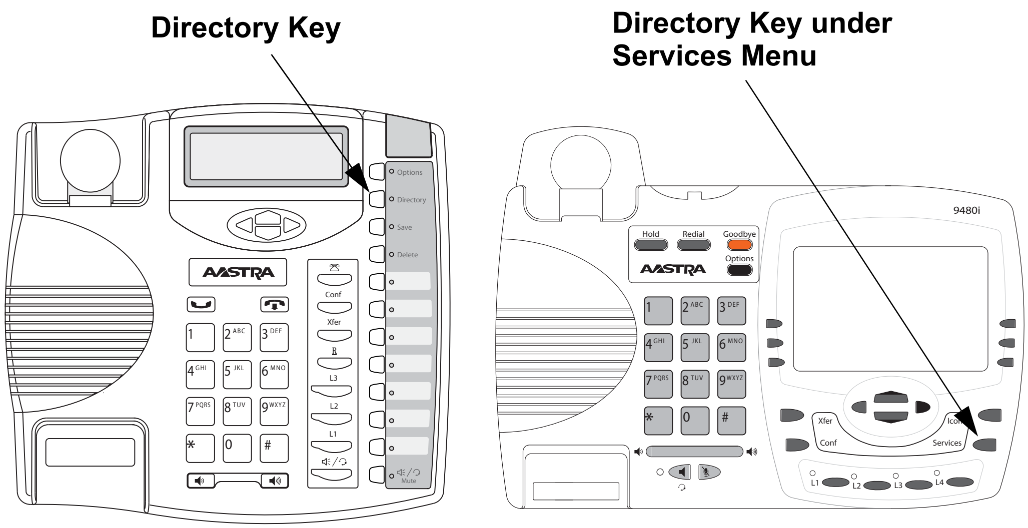
IP Phone UI

Step	Action
1	Press the **"Callers List"** key on the phone to enter the Callers List.
2	Use the r and s to scroll through the line items in the Callers List to find the line items that have the b 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 3 , **Clear,** or **Delete** key (depending on your phone model). **Note:** To cancel a delete function, press the r or the s **Scroll** keys. The line item deletes from the Callers List.
For the 6739i:	
1	Press the key on the phone to enter the Callers List.
2	Use the and keys to navigate to the Caller record marked with a indicating missed calls.
3	To delete the missed call from the Callers List, press the Caller record and then press Delete .

Directory List

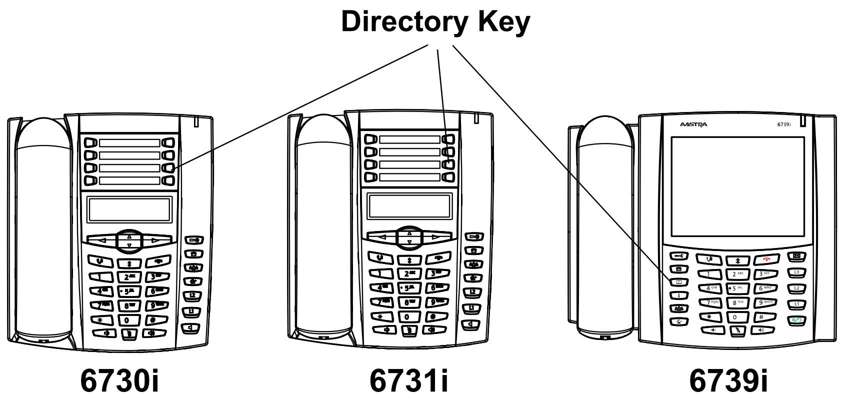
The IP phones have a "**Directory List**" feature that allows you to store frequently used names and numbers on the phone. You can also dial directly from the directory entry.

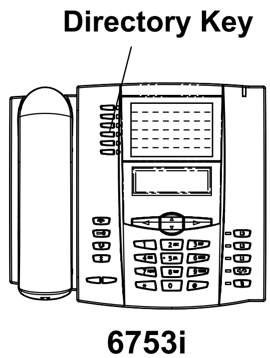
The following illustrations show the default location of the Directory Key on each type of phone model.

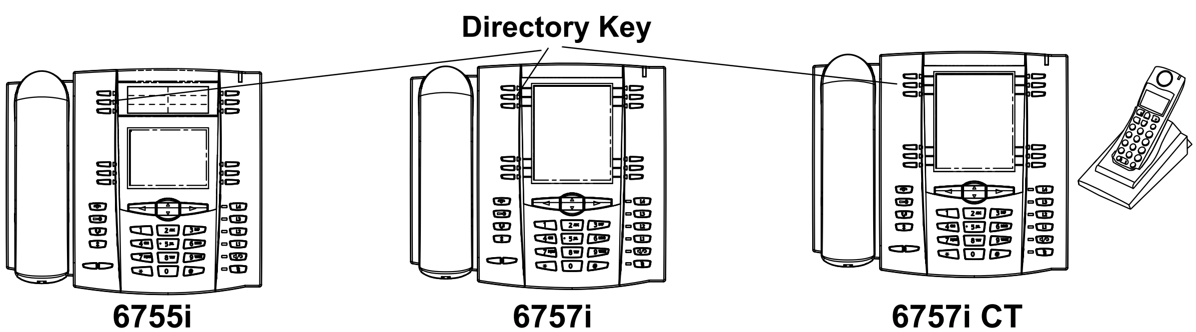


9143i

9480i/9480i CT


Directory Key
6730i
6731i
6739i


Directory Key
6753i


Directory Key
6755i
6757i
6757i CT

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.

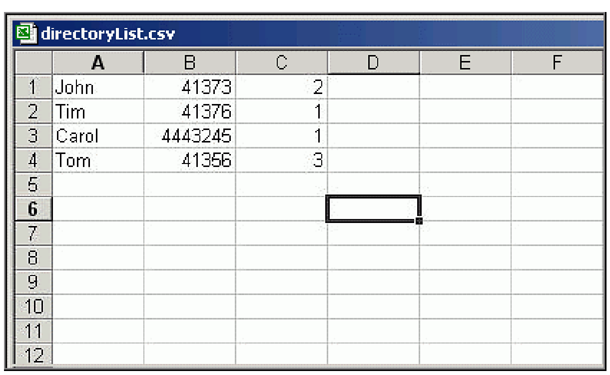
Reference

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 6757i 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 3-Line LCD phones, 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-136.

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 (6757i 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	***On the 9143i and all 6700i phones:*** Press the **DIRECTORY** key to enter the Directory List. ***On the 9480i, 9480i CT:*** Press the **Services** key, and then select "**Directory**" from the list of options. **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 r and s 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 **d / f** 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.
To edit an entry in the Directory List:	
5	a At a line item in the Directory List, press the 3 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.

IP Phone UI

Step	Action
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 r or the s 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 r or the s scroll keys.
To copy an entry from the Directory List to a speed dial key (for 3-Line LCD phones):	
8	At a line item in the Directory List, press the **SAVE** key. The "Save to?" prompt displays. Enter a number from 1 to 9 (associated with the keypad) where you want to save the item as a speed dial. **Note:** You must have a speed dial key previously configured on your phone to use this feature. To configure a speed dial key, see your Model-specific *User's Guide*.
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 CT handsets:	
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 6757i 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 **d / f** handsfree key, or press the **Dial** softkey.

Using the Directory on the 6739i

Accessing and Searching the Directory

Use the following procedures to access and search for entries in your Directory List.

IP Phone UI

1. Press the Directory Key. The Directory screen displays. The screen is blank if the Directory is empty.
2. Use the ↓ and ↑ keys to navigate the list to look for an entry.
3. Press the entry to display a single entry's details.

Using the Search Feature in the Directory

1. Press the button on the main Directory screen. A keyboard displays.
2. Enter the name or phone number of the entry for which you are searching. As you type characters on the keyboard, the entries beginning with those characters display on the screen. When you have completed your typing, the entry you are looking for, displays on the screen and the characters you typed display in a text box.

Dialing from the Directory

You can dial a phone number or extension directly from an entry in the Directory List.

IP Phone UI

1. Press the Directory Key. The Directory screen displays.
2. Use the ↓ and ↑ keys to navigate the list to look for an entry.
   or
   Press the button.
3. Select the entry you want to dial. All of the numbers display that are associated with the single entry (i.e., Cell, Office, Home, etc.).

> **Note:** A "**picture ID**" displays when you select an entry in the Directory if your System Administrator enabled this feature on your phone. Contact your System Administrator for more information about the "picture ID" feature.

4. Press a number button to dial the number or extension. The phone goes off-hook and automatically dials the number from the Directory. For example, pressing the "**Office**" button in the screen above dials "**2345**" automatically from the Directory. The outgoing call defaults to using Line 1.

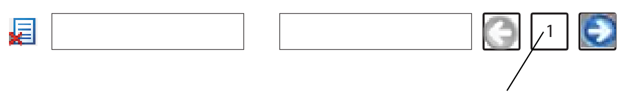
Adding an Entry

You can add up to 200 entries to your Directory. You can also store up to a maximum of 7 numbers associated with a single entry (i.e., office number, cell number, home number, etc.). You can apply custom labels to each entry as required.

IP Phone UI

1. Press the Directory Key. The Directory screen displays.
2. Press the Add button.
3. Press the "**First**" field, enter the first name of the person you are adding to your directory, and then press **<Enter>**. Use the pop-up keyboard to enter the first name.
4. Press the "**Last**" field, enter the last name of the person you are adding to your directory, and then press **<Enter>**.
5. Press Add . The "**Number/Label**" fields display on the screen.

Number **Label** **Line**



This number indicates the Line number.

6. Press the "**Number**" field, enter the number of the person you are adding to your directory, and then press **<Enter>**. Use the pop-up keyboard to enter the number.
7. Press the "**Label**" field, enter the label associated with the number you just entered, and then press **<Enter>**. Use the pop-up keyboard to enter the label.
8. Press the buttons to select a Line (1 through 9) to associate with the phone number and label you are entering. When you dial the current number from the directory, the outgoing call uses the line number you assigned it.
9. To add another phone number and label, press Add An additional "**Number/Label**" field displays.
10. Repeat steps **6** through **8** to enter another number and label for the current entry.
11. When you are finished entering **Numbers/Labels** for the current entry, press . This saves the entry and all associated numbers/labels to the Directory on your phone.

Note: When adding an entry, number, and label to the Directory, you can press the any time to cancel the entry. It is not saved to the Directory.

---

12. Press the to return to the previous menu or press the to return to the idle screen.

Editing an Entry

You can edit a Directory entry, number, and/or label as required from a single entry screen..

IP Phone UI

1. Press the Directory Key. The Directory screen displays.
2. Use the and keys to navigate the list to look for an entry.
   or
   Press the button.
3. Press the entry you want to edit. The single entry screen displays.
4. Press the number and/or label that you want to edit, and press the Edit button.
5. Edit the Number and/or Label as required using the touch keypad that displays, and press .
6. Press the to return to the previous menu or press the to return to the idle screen

Deleting an Entry

You can delete entries from the Directory List in the following ways:

- Delete a single entry and all associated numbers and labels
- Delete all entries in the Directory List
- Delete specific numbers and labels from an entry

Deleting a single entry or all entries

IP Phone UI

1. Press the Directory Key. The Directory screen displays.
2. Use the and keys to navigate the list to look for an entry.
   or
   Press the button.
3. Press the entry you want to delete. The single entry screen displays.
4. Press the Delete button. The following screen displays.
5. Press Delete to delete the entry and all associated numbers and labels from the Directory. The main Directory screen displays.
   or
   Press Delete All to delete **all** entries from the Directory. The main Directory screen displays.
6. Press the to return to the previous menu or press the to return to the idle screen

Deleting a number and/or label from an entry

IP Phone UI

1. Press the Directory Key. The Directory screen displays.
2. Use the and keys to navigate the list to look for an entry.
   or
   Press the button. For a procedure on using the search button, see "Using the Search Feature in the Directory" on page -277.
3. Press the entry for which you want to delete a number/label. The single entry screen displays.
4. Press the Edit button. All of the Numbers/Labels associated with this entry display on the screen.
5. Press the button next to the Number/Label to delete both the number and label from the entry.
   or
   Press the field (Number or Label) you want to delete. When the keyboard displays, press the **<Backspace>** key to delete the text in the field and press **<Enter>**.
6. Press the button to save your change(s).
7. Press the to return to the previous menu or press the to return to the idle.

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

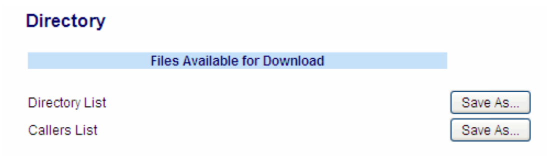
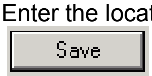
For specific parameters you can set in the configuration files for downloading the Directory List, see Appendix A, the section, "Directory Settings" on page A-136.

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.

	Aastra Web UI
1	Click on **Operation->Directory.** 
2	In the Directory List field, click on . A “File Download” message displays.
3	Click .
4	Enter the location on your computer where you want to download the Directory List and click . The *directorylist.csv* file downloads to your computer.
5	Use a spreadsheet application to open and view the Directory List.

Voicemail

The Voicemail feature on the 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 or 1-6 lines depending on your model phone).

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 IP phone that has 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

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 IP phone, 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-135.

Using Voicemail

Use the following procedure to access and use voicemail.

IP Phone UI

Step	Action
For all phones:	
1	Press **Services** on the phone to display the **Services** menu.
2	From the **Services** menu, select "**Voicemail**".
3	Use the r and s to scroll through the line items in the Voicemail List.
4	When you have selected a line item, press the **d / f** handsfree key, 4 **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.

Using Voicemail on the 6739i

IP Phone UI

1. Press the Key or the icon on the phone.
2. Using your finger, scroll through the line items in the Voicemail list.
3. When you have selected a line item, press the Key, or press a line/call appearance key 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.

Note: You can also access your Voicemail via the “**Services**” Key on your phone if this has been setup by your System Administrator.

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) **AastraIPPhoneFormattedTextScreen** (for Text screens)	Creates a screen of text that wraps appropriately. Creates a formatted screen of text (specifies text alignment, text size, text static or scrolling)
AastraIPPhoneInputScreen (for User Input screens) **AastraIPPhoneInputScreen Time and Date Attributes** (for User Input screens)	Creates screens for which the user can input text where applicable. 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.
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 5-287 for more information).
AastraIPPhoneImageScreen (Standard Bitmap Image)	Creates a display with a single bitmap image according to alignment, height, and width specifications.

XML Object	Description
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.

Reference

For more information about creating customized XML applications, contact Aastra Customer Support regarding the "*Aastra XML Developer's Guide*."

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 the *Aastra XML Developer's Guide*) Configuration files (See page 5-286) Aastra Web UI (See page 5-286)	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 5-287) Aastra Web UI (See page 5-287)	1 to 25 Default = 5
Timeout	Specifies a timeout value for the LCD screen display. **Use with:** XML object (See the *Aastra XML Developer's Guide*)	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 5-288)	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 the *Aastra XML Developer's Guide*)	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 the *Aastra XML Developer's Guide*)
		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 the *Aastra XML Developer's Guide*.

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 the *Aastra XML Developer's Guide*)
- **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 5-72.

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 5-73.

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 the *Aastra XML Developer's Guide* 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 the *Aastra XML Developer's Guide*.

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-145.

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 the *Aastra XML Developer's Guide*.

Example Configuration of XML Application

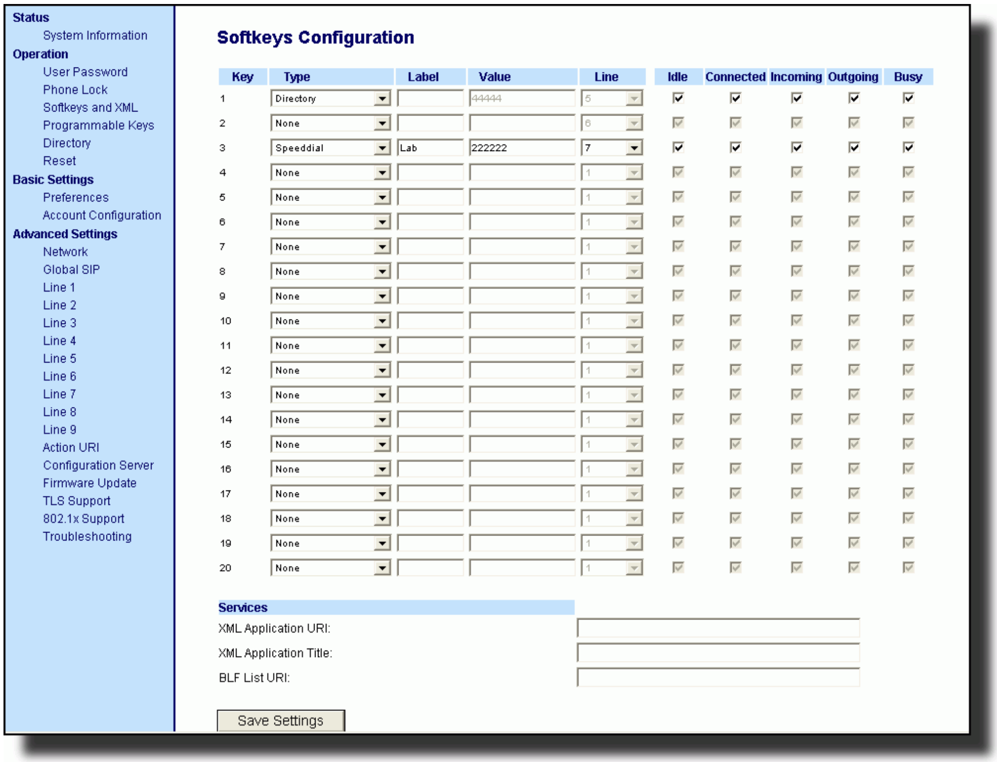
The following example shows the parameters you enter in the configuration files to configure an XML application:

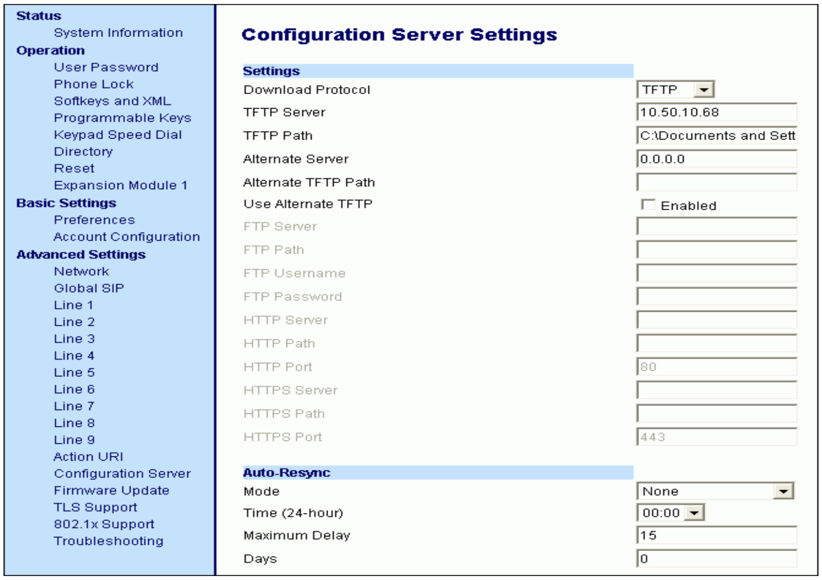
```
xml application URI: http://172.16.96.63/aastra/internet.php

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.

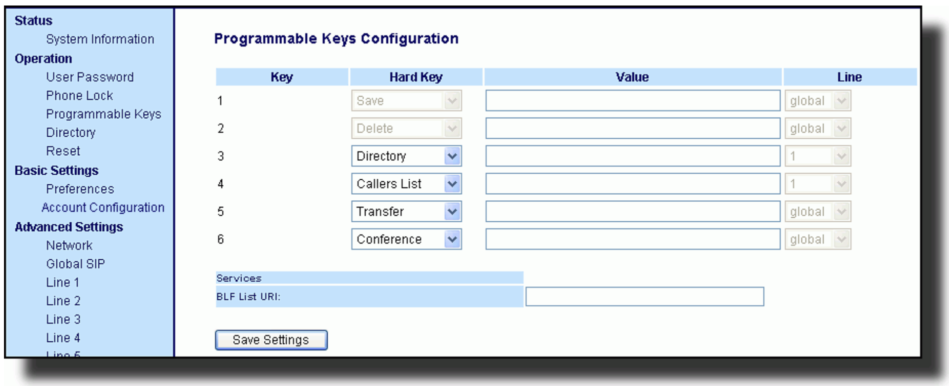
Configuration Files
For specific parameters you can set in the configuration files, see Appendix A, the section, "XML Settings" on page A-145.

	Aastra Web UI
	For 8 and 11-Line LCD phones:
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**".
8	Click  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.

	Aastra Web UI
9	Click on **Advanced Settings->Configuration Server**. 
10	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.
11	Click Save Settings to save your changes. **Note:** No posting is performed if a session times out.

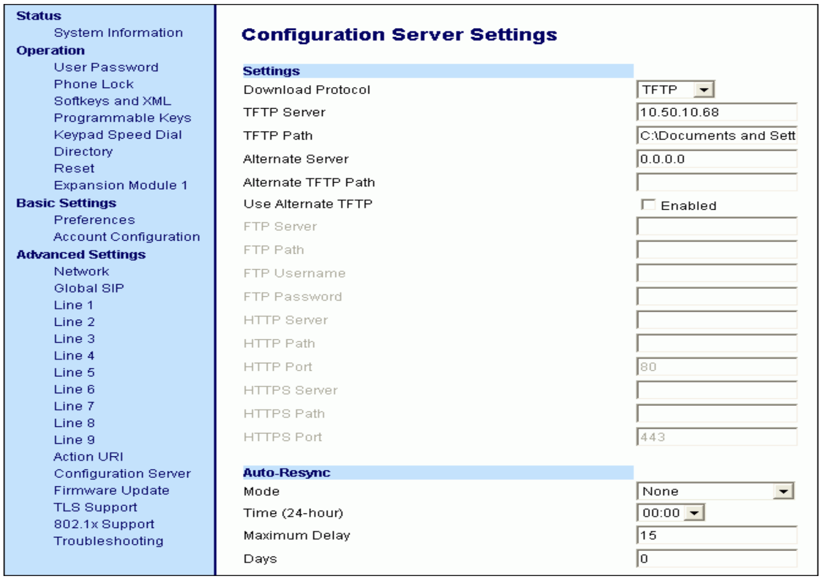
Aastra Web UI

<table>
<tr><td colspan="2">For 3-Line LCD phones:</td></tr>
<tr><td>1</td><td>Click on Operation->Programmable Keys.

6753i Screen

</td></tr>
<tr><td>2</td><td>For 3-Line LCD phones, select an available key.</td></tr>
<tr><td>3</td><td>In the "Hard Key" field, select XML from the list box.</td></tr>
<tr><td>4</td><td>In the "Value" field, enter the IP address or qualified domain name of the XML application.</td></tr>
<tr><td>5</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>
</table>

	Aastra Web UI
6	Click on **Advanced Settings->Configuration Server**. 
7	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.
8	Click Save Settings to save your changes. **Note:** No posting is performed if a session times out.

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.

Reference

For more information about customizing the phones using XML objects, contact Aastra Customer Support regarding the “Aastra XML Development Guide.”

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

IP Phone UI	
Step	**Action**
For 8 and 11-Line LCD phones:	
1	Press the **Services** key on the phone to display the **Services** menu.
2	Select "**Custom Features**".
3	Use the r and s 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 3-Line LCD phones:	
1	Press the programmable key configured on the phone for XML services. A "**Custom Features**" screen displays.
2	Use the r and s 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.

Step	Action
For the 6739i:	
1	Press **Services** key on the phone.
2	Press "**Custom Features**".
3	Use the and keys 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	Press the to return to the previous screen.
5	Press the button at any time to return to the idle screen.

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:

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 Action URI feature prevents the phones from hanging if the Action URIs should fail. The phones also support transparent, non-blocking, XML post execute item URIs.

The IP phone XML events that support this feature are defined in the following table.

Action URI	Description
Startup	Specifies the URI for which the phone executes a GET on when a startup event occurs.
Successful Registration	Specifies the URI for which the phone executes a GET on when a successful registration event occurs.
Registration Event	Specifies the URI for when registration events occur or when there are registration state changes. **Note:** This action URI is not called when the same event is repeated (for example, a timeout occurs again when registration is already in a timeout state.)
Incoming Call	Specifies the URI for which the phone executes a GET on when an incoming call event occurs.
Outgoing Call	Specifies the URI for which the phone executes a GET on when an outgoing call event occurs.
Offhook	Specifies the URI for which the phone executes a GET on when an offhook event occurs.
Onhook	Specifies the URI for which the phone executes a GET on when an onhook event occurs.
Connected	Specifies the URI for which the phone executes an HTTP GET when it goes into the "connected" state. This includes regular phone calls, intercom calls, paging calls, RTP streaming, and the playing of a WAV file. It is also triggered when the phone establishes the second leg of a 3-way call. For more information, see "Action URI Connected" on page 5-354.
Disconnected	Specifies the URI that the phone executes a GET on, when it transitions from the incoming, outgoing, calling, or connected state into the idle state. For more information, see "Action URI Disconnected" on page 5-307.
XML SIP Notify	Specifies the URI to be called when an empty XML SIP NOTIFY is received by the phone. For more information, see "XML SIP Notify Events" on page 5-310.
Poll	Specifies the URI to be called every "action uri poll interval" seconds. For more information, see "Polling Action URIs" on page 5-303.
Poll Interval	Specifies the interval, in seconds, between calls from the phone to the "action uri poll". For more information, see "Polling Action URIs" on page 5-303.

Note: For more information about the XML execute items, see the *Aastra XML Developer's Guide*.

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.

Action URIs and Associated Variables

Configuration File Parameters	Aastra Web UI Parameters at Advanced Settings->Action URI	Applicable Variables
action uri startup	Startup	$$REMOTENUMBER$$ $$DISPLAYNAME$$ $$SIPUSERNAME$$ $$SIPAUTHNAME$$ $$INCOMINGNAME$ $$PROXYURL$$ $$LINESTATE$$ $$LOCALIP$$ $$DISPLAYNAME$$ $$CALLDURATION$$ $$CALLDIRECTION$$
action uri registered	Successful Registration	$$SIPUSERNAME$$ $$SIPAUTHNAME$$ $$PROXYURL$$ $$LINESTATE$$ $$LOCALIP$$
action uri registration event	Registration Event	$$REGISTRATIONSTATE$$ $$REGISTRATIONCODE$$
action uri incoming	Incoming Call	$$REMOTENUMBER$$ $$DISPLAYNAME$$ $$SIPUSERNAME$$ $$INCOMINGNAME$ $$LINESTATE$$ $$CALLDURATION$$ $$CALLDIRECTION$$ $$LOCALIP$$
action uri outgoing	Outgoing Call	$$REMOTENUMBER$$ $$SIPUSERNAME$$ $$LINESTATE$$ $$LOCALIP$$
action uri offhook	Offhook	$$LINESTATE$$ $$LOCALIP$$
action uri onhook	Onhook	$$LOCALIP$$ $$CALLDURATION$$ $$CALLDIRECTION$$ $$LINESTATE$$

Configuration File Parameters	Aastra Web UI Parameters at Advanced Settings->Action URI	Applicable Variables
action uri connected	Connected	$$REMOTENUMBER$$ $$DISPLAYNAME$$ $$SIPUSERNAME$$ $$SIPAUTHNAME$$ $$INCOMINGNAME$ $$PROXYURL$$ $$LINESTATE$$ $$LOCALIP$$ $$DISPLAYNAME$$ $$CALLDURATION$$ $$CALLDIRECTION$$ $$REGISTRATIONSTATE$$ $$REGISTRATIONCODE$$$
action uri disconnected	Disconnected	$$LINESTATE$$ $$LOCALIP$$ For more information, see "Action URI Disconnected" on page -307.
action uri xml sip notify	XML SIP Notify	$$LOCALIP$$ For more information, see "XML SIP Notify Events" on page 5-310.
action uri poll	Poll	For more information, see "Polling Action URIs" on page 5-303.
action uri poll interval	Interval	For more information, see "Polling Action URIs" on page 5-303.

Variable Descriptions

The following table provides a description of each variable.

Variable	Description
$$SIPUSERNAME$$	Username associated with: • registered phone • incoming caller • outgoing caller
$$SIPAUTHNAME$$	Authentication name associated with: • registered phone
$$PROXYURL$$	Proxy URL associated with: • registered phone
$$LINESTATE$$	Current line state associated with: • registered phone • incoming caller • outgoing caller • offhook • onhook • disconnected

Variable	Description
$$LOCALIP$$ **Note:** This variable allows for enhanced information in call records and billing applications.	IP Address associated with: • registered phone • onhook
$$REMOTENUMBER$$	Remote number associated with: • incoming caller • outgoing caller
$$DISPLAYNAME$$	Display name associated with: • incoming caller
$$SIPUSERNAME$$	Username associated with: • registered phone • incoming caller • outgoing caller
$$INCOMINGNAME$$	Name associated with: • incoming caller
$$CALLDURATION$$ **Note:** This variable allows for enhanced information in call records and billing applications.	Duration of last call. This variable is associated with: • onhook
$$CALLDIRECTION$$ **Note:** This variable allows for enhanced information in call records and billing applications.	Specifies whether the current/last call was incoming or outgoing. This variable is associated with: • onhook
$$REGISTRATIONSTATE$$	Specifies the state of the phone's registration. Registration states can be: • registered • unregistered • expired • refused • timeout
$$REGISTRATIONCODE$$	Specifies the code generated during the registration process. Registration code can be: "xxx" where xxx is the 3 digit code; for example, "403". Possible codes are: • 001 (registration successful) • 403 (registration failed)

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`

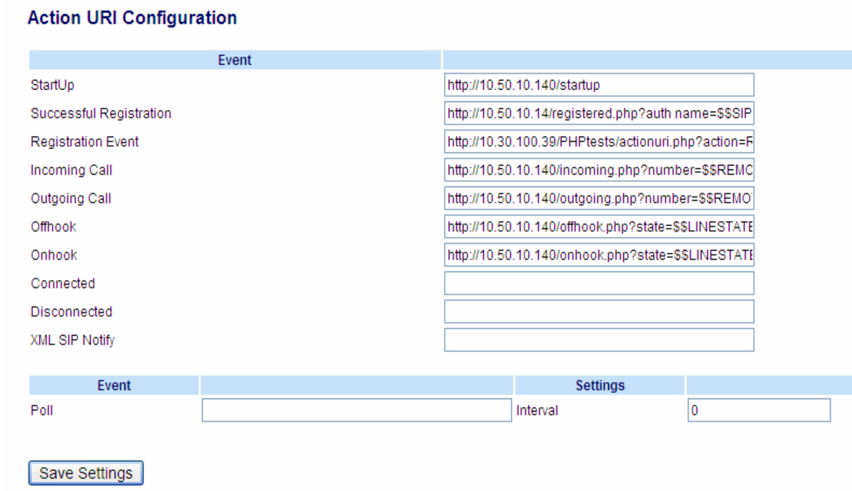
Note: If the phone can't find the Action URI you specify, it returns a "NULL" response. For example,
`http://10.50.10.140/outgoing.pl?number=`

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-148.

	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, use the associated variables indicated in the table "Action URIs and Associated Variables" on page 5-297. The "**Successful Registration**" parameter executes on the first successful registration of each unique line configured on the phone.

4	Enter an XML URI in the "**Registration Event**" field, for when the phone performs registration. For example:
	
	`http://10.30.100.39/PHPtests/`
	`actionuri.php?action=RegEvt&regstate=$$REGISTRATIONSTATE$$&regcode=$$REGI`
	`STRATIONCODE$$`
	
	This parameter specifies the URI that the phone executes a GET on, when a registration event change occurs.
	
	Note: For a registration event, use the associated variables indicated in the table "Action URIs and Associated Variables" on page 5-297.

	Aastra Web UI
5	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, use the associated variables indicated in the table "Action URIs and Associated Variables" on page 5-297.
6	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, use the associated variables indicated in the table "Action URIs and Associated Variables" on page 5-297.
7	Enter an XML URI for an offhook event in the "**Offhook**" field. For example: `http://10.50.10.140/offhook.php?state=$$LINESTATE$$` This parameter specifies the URI for which the phone executes a GET on when an offhook event occurs. **Note:** For an offhook event, use the associated variables indicated in the table "Action URIs and Associated Variables" on page 5-297.
8	Enter an XML URI for an onhook event in the "**Onhook**" field. For example: `http://10.50.10.140/onhook.php?state=$$LINESTATE$$` This parameter specifies the URI for which the phone executes a GET on when an onhook event occurs. **Note:** For an onhook event, use the associated variables indicated in the table "Action URIs and Associated Variables" on page 5-297.
9	To configure a Connected event, see the section, "Action URI Connected" on page 5-305.
10	To configure a Disconnected event, see the section, "Action URI Disconnected" on page 5-307.
11	To configure an XML SIP Notify event, see the section, "XML SIP Notify Events" on page 5-310.
12	(Optional) You can poll a URI at specific intervals on the phones. For more information about polling Action URIs, see "Polling Action URIs" on page 5-303.
13	Click Save Settings to save your changes.

Polling Action URIs

Another way to reach a phone behind a NAT/firewall is to have the phone make an XML call at periodic intervals. An Administrator can use the **action uri poll** parameter that commands the phone to perform an XML call at configurable intervals.

An Administrator can specify the URI to be called and specify the interval between polls using the configuration files or the Aastra Web UI. Configuration of this feature is dynamic (no reboot required).

Configuring Polling Action URI via the Configuration Files

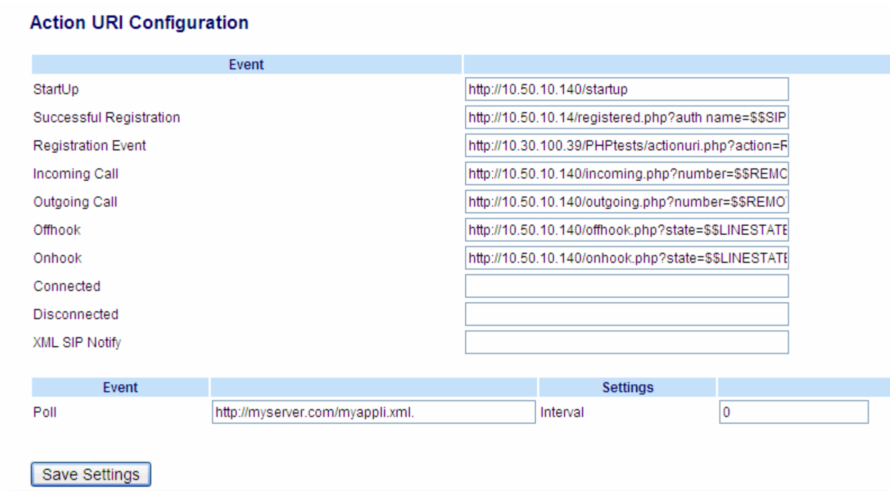
Use the following parameters to configure the polling Action URI on the IP Phones.

- **action uri poll**
- **action uri poll interval**

Configuration Files
For the specific parameter you can set in the configuration files, see Appendix A, the section, "Polling Action URI Settings" on page A-153.

Configuring Polling Action URI via the Aastra Web UI

Use the following procedure to configure polling Action URI using the Aastra Web UI.

	Aastra Web UI
1	Click on **Advanced Settings->Action URI**. 
2	In the "**Poll**" field, enter a URI to be called every "action uri pool interval" seconds. Enter the value in a URI format. For example, **http://myserver.com/myappli.xml**.
3	In the "**Interval**" field, enter the interval, in seconds, between calls from the phone to the "action uri poll". The value of "**0**" is disabled.
4	Click  to save your changes.

Action URI Connected

A parameter called "**action uri connected**" (configuration files) and "**Connected**" (Aastra Web UI) now allows XML scripts to determine when a call is connected. When enabled, the phone triggers an HTTP GET when it goes into the "connected" state. This includes regular phone calls, intercom calls, paging calls, RTP streaming, and the playing of a WAV file. It is also triggered when the phone establishes the second leg of a 3-way call.

This parameter can use the following variables:

- $$REMOTENUMBER$$
- $$DISPLAYNAME$$
- $$SIPUSERNAME$$
- $$SIPAUTHNAME$$
- $$INCOMINGNAME$
- $$PROXYURL$$
- $$LINESTATE$$
- $$CALLDIRECTION$$
- $$LOCALIP$$
- $$DISPLAYNAME$$
- $$CALLDURATION$$
- $$REGISTRATIONSTATE$$
- $$REGISTRATIONCODE$$

If the Administrator enables this feature (by specifying a connect URI), when a call is connected, the phone checks to see if the event has a Connect URI configured. If the phones finds a configured URI it executes an XML script or the variable if defined.

Example

In the configuration files, you can enter the following:

```
action uri connected: http://www.example.com/connect.php
```

An Administrator can enable the "Connected" Action URI feature using the configuration files or the Aastra Web UI.

Limitations

- During SLA calls, the phone uses the Action URI Connected parameter when the line is seized before the caller dials out.
- SCA and BLA calls on hold trigger the Action URI Connected parameter, since the retrieval is a 2nd call by the phone, and the phone cannot link the retrieved call with the earlier held call.

For more information about the XML API objects, see *Aastra's XML Developers Guide*.

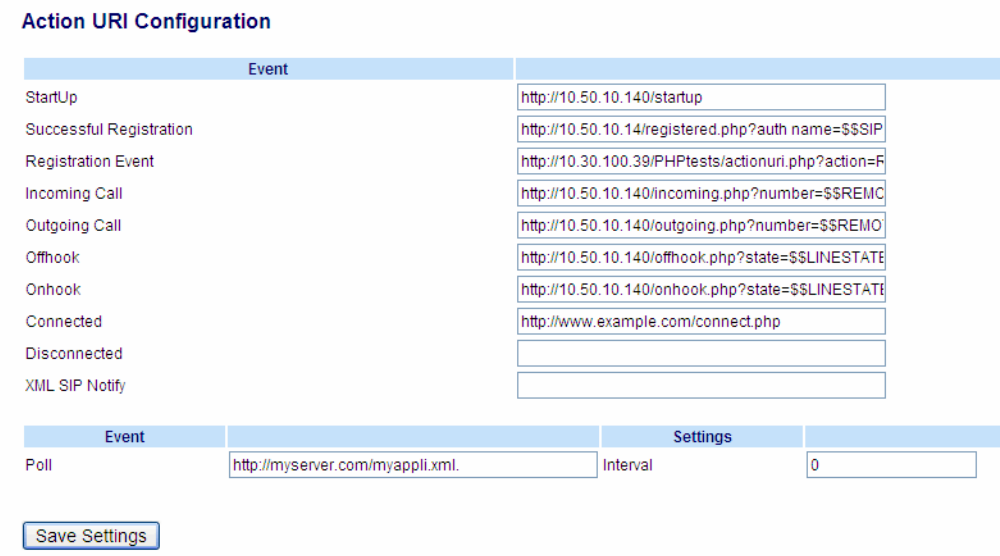
Configuring the Action URI Connected Feature

Use the following procedure to configure the Action URI Connected feature on the phone.

Configuration Files

For specific parameters you can set in the configuration files, see Appendix A, the section, "Action URI Settings" on page A-148.

<table>
<tr><th colspan="2">Aastra Web UI</th></tr>
<tr><td>1</td><td>Click on Advanced Settings->Action URI.



</td></tr>
<tr><td>2</td><td>In the “Connected” field, enter a valid URI for which the phone executes a GET on, when it goes into the “connected” state. Leaving this field empty disables the Action URI Connected feature. For example,

http://www.example.com/connect.php</td></tr>
<tr><td>3</td><td>Click 

 to save your settings.</td></tr>
</table>

Action URI Disconnected

The phones have a parameter, "**action uri disconnected**" that allow a disconnect event to occur when the phone transitions from any active state (outgoing, incoming, connected, or calling) to an idle state. This parameter can use the variable "**$$LINESTATE$$**".

Note: The **$$LINESTATE$$** variable is optional and not required when enabling the "**action uri disconnected**" parameter.

If the Administrator enables this feature (by specifying a disconnect URI), when a call is disconnected, the phone checks to see if the event has a Disconnect URI configured. If the phones finds a configured URI with a $$LINESTATE$$ variable, it replaces the $$LINESTATE$$ variable with the appropriate line state of the current active line. After all of the variables are bound, the phone executes a GET on the URI. The following table lists the applicable values for the $$LINESTATE$$ variable.

$$LINESTATE$$ Value	Description	Meaning in a Disconnected URI
IDLE	Phone is idle.	N/A
DIALING	Phone is offhook and ready to dial.	N/A
CALLING	A SIP INVITE was sent but no response was received.	Error occurred during the call.
OUTGOING	Remote party is ringing.	Call was cancelled.
INCOMING	Local phone is ringing.	Call was missed or cancelled.
CONNECTED	Parties are talking.	Call was successful.
CLEARING	Call was released but not acknowledged.	N/A

The Action URI Disconnect feature allows an Administrator to determine the reason for the disconnect if required.

Note: If you enable the Action URI Disconnect feature by specifying a URI, the URI is called when any disconnect event occurs including an intercom call or a conference setup.

Example

If you enter the following string on Phone A for the "**action uri disconnected"** parameter:

```
action uri disconnected: http://fargo.ana.aastra.com/
disconnected.xml?state=$$LINESTATE$$
```

and then Phone A calls Phone B, Phone B answers and then hangs up, Phone A executes a GET on:

`http://fargo.ana.aastra.com/disconnected.xml?state=CONNECTED`

which is what the remote server receives.

> **Note:** If the phone can't find the Action URI you specify, it returns a "NULL" response. For example,
> **`http://fargo.ana.aastra.com/disconnected.xml?state=`**

An Administrator can enable the disconnect feature using the configuration files or the Aastra Web UI.

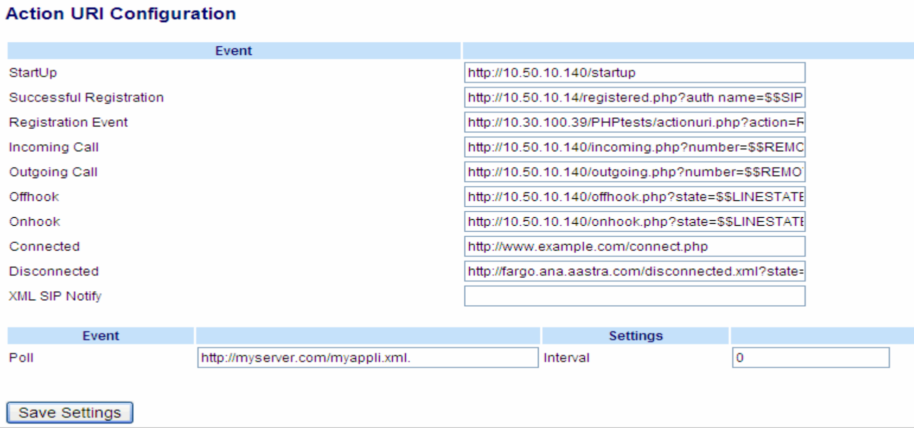
Configuring the Action URI Disconnected Feature

Use the following procedure to configure the Action URI Disconnected feature on the phone.

Configuration Files
For specific parameters you can set in the configuration files, see Appendix A, the section, "Action URI Settings" on page A-148.

Aastra Web UI

1 Click on **Advanced Settings->Action URI->Event.**



2 In the "**Disconnected**" field, enter a valid URI for which the phone executes a GET on, when it transitions from the incoming, outgoing, calling, or connected state into the idle state. Leaving this field empty disables the Action URI Disconnected feature. For example,

http://fargo.ana.aastra.com/disconnected.xml?state=$$LINESTATE$$

The following table lists the applicable values and descriptions for the $$LINESTATE$$.

$$LINESTATE$$ Value	Description	Meaning in a Disconnected URI
IDLE	Phone is idle.	N/A
DIALING	Phone is offhook and ready to dial.	N/A
CALLING	A SIP INVITE was sent but no response was received.	Error occurred during the call.
OUTGOING	Remote party is ringing.	Call was cancelled.
INCOMING	Local phone is ringing.	Call was missed or cancelled.
CONNECTED	Parties are talking.	Call was successful.
CLEARING	Call was released but not acknowledged.	N/A

3 Click  to save your settings.

XML SIP Notify Events

In order for an XML push to bypass the NAT/firewall, the phone supports a proprietary SIP NOTIFY event (aastra-xml) with or without XML content. An Administrator can enable/disable the SIP NOTIFY event using a specific parameter in the configuration files (**sip xml notify event**) or the Aastra Web UI (**XML SIP Notify**).

If XML content is provided in the SIP NOTIFY, it is processed directly by the phone as it is done for an XML PUSH.

If the content is empty in the SIP NOTIFY, the phone automatically triggers a new pre-configured action uri (**action uri xml sip notify**).

Example of a SIP NOTIFY with XML Content

```
NOTIFY sip:200@10.30.100.103:5060 SIP/2.0
Via: SIP/2.0/UDP 10.30.100.103:5060;branch=z9hG4bK7bbc1fac;rport
From: <sip:201@10.30.100.103:5060>;tag=81be2861f3
To: Jacky200 <sip:200@10.30.100.103:5060>
Contact: <sip:201@10.30.100.103>
Call-ID: 59638f5d95c9d301
CSeq: 4 NOTIFY
Max-Forwards: 70
Event: aastra-xml
Content-Type: application/xml
Content-Length: 115
<AastraIPPhoneExecute><ExecuteItem URI="http://10.30.100.39/XMLtests/
SampleTextScreen.xml"/></AastraIPPhoneExecute>
```

When the phone receives the SIP NOTIFY, the XML content is processed as any XML object. In the above example, the phone calls http://10.30.100.39/XMLtests/SampleTextScreen.xml after reception of the SIP NOTIFY.

Note: The phone supports all the current XML objects with all the existing limitations. For example if an AastraIPPhoneExecute is used, the embedded URI(s) can not be HTTPS based.

Example of a SIP NOTIFY without XML Content

```
NOTIFY sip:200@10.30.100.103:5060 SIP/2.0
Via: SIP/2.0/UDP 10.30.100.103:5060;branch=z9hG4bK7bbc1fac;rport
From: <sip:201@10.30.100.103:5060>;tag=81be2861f3
To: Jacky200 <sip:200@10.30.100.103:5060>
Contact: <sip:201@10.30.100.103>
Call-ID: 59638f5d95c9d301
CSeq: 4 NOTIFY
Max-Forwards: 70
Event: aastra-xml
Content-Type: application/xml
Content-Length: 0
```

When the phone receives the SIP NOTIFY, it will trigger the **action uri xml sip notify** parameter**,** if it has been previously configured using the configuration files or the Aastra Web UI. If the **action uri xml sip notify** parameter is not configured, the phone does not do anything.

On the phone side, a System Administrator can enable or disable this SIP NOTIFY feature using the configuration files or the phone Web UI.

Also to ensure that the SIP NOTIFY is coming from a trusted source, it is recommended that you enable the Whitelist feature (Whitelist Proxy parameter) on the IP phone. If enabled, and the phone receives a SIP NOTIFY from a server that is NOT on the whitelist, the phone rejects the message.

Enabling/Disabling XML SIP NOTIFY using the Configuration Files

To enable/disable the SIP NOTIFY event, you can set the following parameter in the configuration files:

- **sip xml notify event**

If the content is missing in the SIP NOTIFY message received by the phone, the phone automatically uses the value you specify for the following parameter:

- **action uri xml sip notify**

Configuration Files

For the specific parameter you can set in the configuration files, see Appendix A, the section, "Polling Action URI Settings" on page A-153.

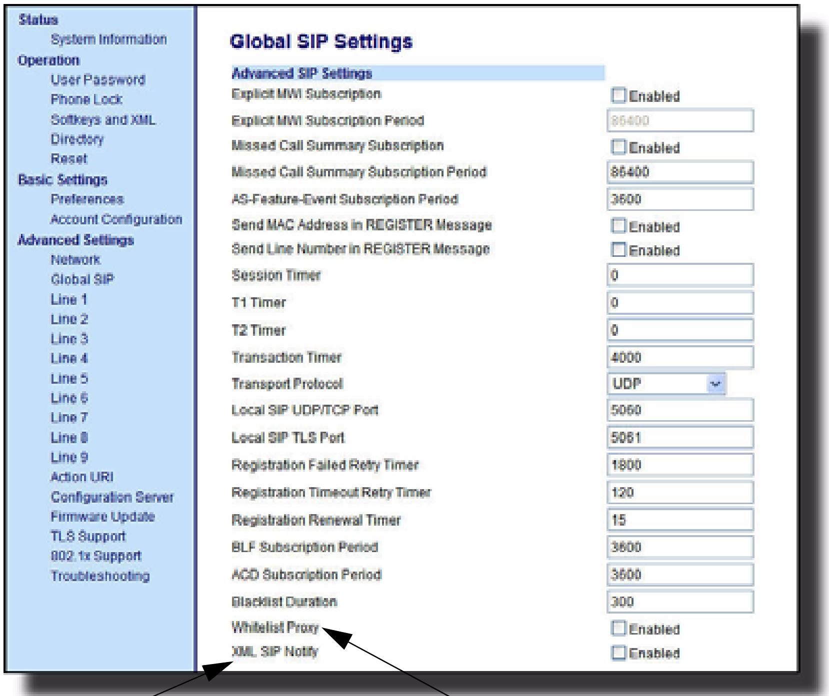
Enabling/Disabling XML SIP NOTIFY Using the Aastra Web UI

Use the following procedure to enable/disable the XML SIP NOTIFY feature in the Aastra Web UI.

Aastra Web UI

Enabling/Disabling the XML SIP NOTIFY Feature

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



XML SIP Notify Parameter **Enabling Whitelist Proxy is recommended**

2 The “**XML SIP Notify**" field is disabled by default. To enable this field, check the box.

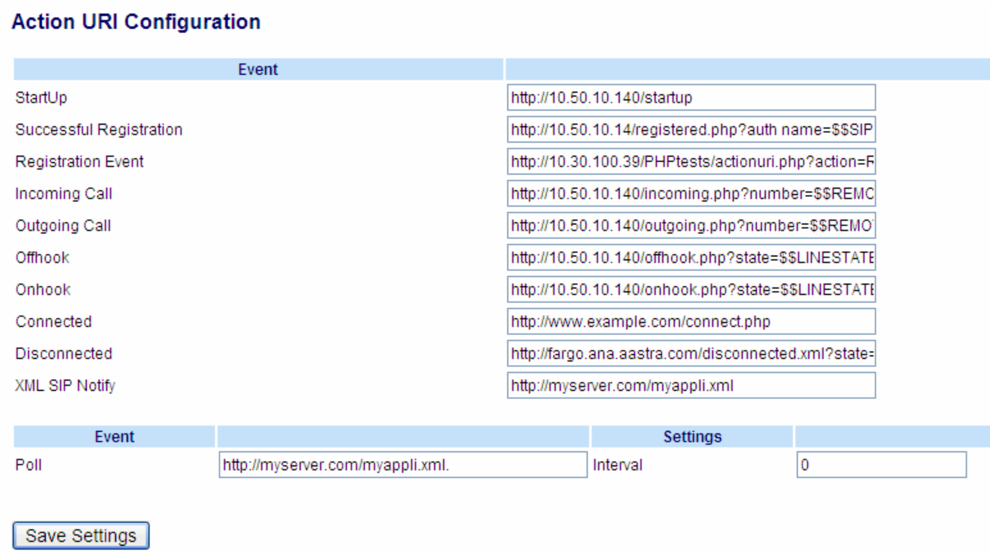
This field enables or disables the phone to accept or reject an aastra-xml SIP NOTIFY message.

Note: To ensure the SIP NOTIFY is coming from a trusted source, it is recommended that you enable the Whitelist feature (**Whitelist Proxy** parameter) on the IP phone. If enabled, and the phone receives a SIP NOTIFY from a server that is NOT on the whitelist (i.e. untrusted server), the phone rejects the message.

3 Click [Save Settings] to save your changes.

Configuring XML SIP NOTIFY using the Aastra Web UI if an Empty SIP NOTIFY Message Received by the Phone

Use the following procedure in the Aastra Web UI to configure the XML SIP NOTIFY parameter when the phone receives an empty SIP NOTIFY message.

	Aastra Web UI
1	If the content is missing in the SIP NOTIFY message received by the phone, the phone automatically uses the value you specify for the Action URI **XML SIP Notify** parameter at the path *Advanced Settings->Action URI*. Click on **Advanced Settings->Action URI.** 
2	Specify a URI to be called when an empty XML SIP NOTIFY is received by the phone. For example: **http://myserver.com/myappli.xml** **Note:** The **sip xml notify event** parameter at the path *Advanced Settings->Global SIP->Advanced SIP Settings* must be enabled.
3	Click  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:

- $$SIPUSERNAME$$
- $$SIPAUTHNAME$$
- $$PROXYURL$$
- $$LINESTATE$$
- $$LOCALIP$$
- $$REMOTENUMBER$$
- $$DISPLAYNAME$$
- $$SIPUSERNAME$$
- $$INCOMINGNAME$$
- $$CALLDURATION$$
- $$CALLDIRECTION$$

Note: For a description of each variable in the above list, see "Variable Descriptions" on page 5-298.

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

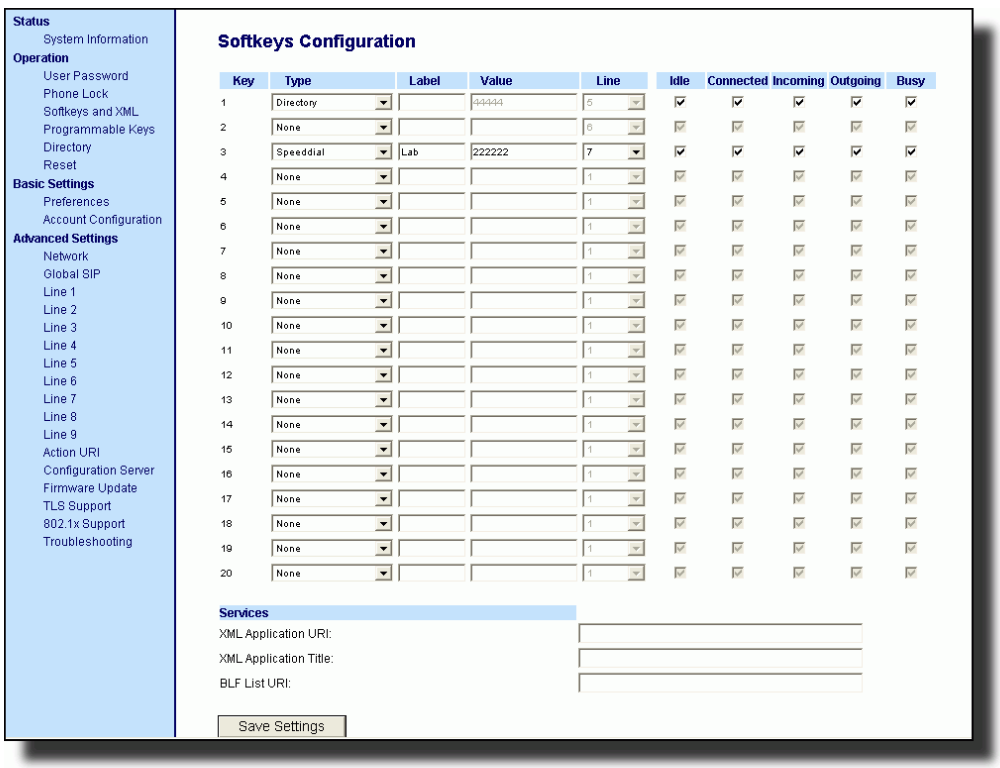
Use the following procedures to configure XML Softkey URIs using the configuration files or the Aastra Web UI.

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-202.

Aastra Web UI

For 8 and 11-Line LCD phones:

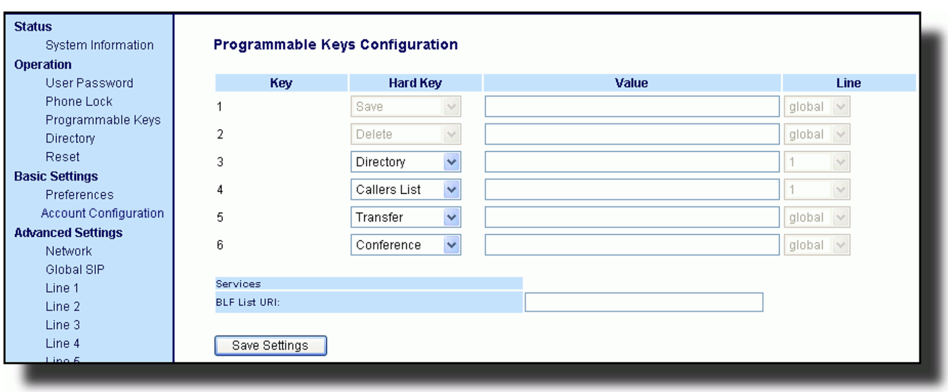
1	Click on **Operation->Softkeys and XML**. 
2	Select a key from keys 1 through 20.
3	In the "**Type**" field, select **XML** from the list box.
4	In the "**Label**" field, enter a label that displays on the IP phone for the XML softkey. For example, "**aastra**".

Aastra Web UI

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: • $$SIPUSERNAME$$ • $$SIPAUTHNAME$$ • $$PROXYURL$$ • $$LINESTATE$$ • $$LOCALIP$$ • $$REMOTENUMBER$$ • $$DISPLAYNAME$$ • $$SIPUSERNAME$$ • $$INCOMINGNAME$$ • $$CALLDURATION$$ • $$CALLDIRECTION$$
6	Click **Save Settings** to save your changes.

Aastra Web UI

For 3-Line LCD phones:

1	Click on **Operation->Programmable Keys.** **6753i Screen** 
2	On 3-Line LCD phones, select an available key.
3	In the "**Type**" field, select **XML** from the list box.
4	On 3-Line LCD phones, in the "**Label**" field, enter a label that displays on the IP phone for the XML softkey. For example, "**aastra**".

	Aastra Web UI
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: • $$SIPUSERNAME$$ • $$SIPAUTHNAME$$ • $$PROXYURL$$ • $$LINESTATE$$ • $$LOCALIP$$ • $$REMOTENUMBER$$ • $$DISPLAYNAME$$ • $$SIPUSERNAME$$ • $$INCOMINGNAME$$ • $$CALLDURATION$$ • $$CALLDIRECTION$$
6	Click Save Settings to save your changes.

XML Key Redirection

The IP phones allow the redirecting of phone-based hard keys to XML scripts. This allows the server to provide the phone with Redial, Transfer (Xfer), Conference (Conf), and Intercom (Icom) key features, and the Voicemail option feature, rather then accessing them from the phone-side. This feature allows you to access the redirected keys and voicemail option from the server using the IP Phone's Services Menu. By default, the server-side keys function the same as the phone-side key features.

The following table identifies the phone states that apply to each key redirection.

Hard Keys/Options	Redirects in
Conference (Conf)	the connected state
Transfer (Xfer)	the connected and dialing states
Redial	all states
Intercom (Icom)	all states
Voicemail	all states

Notes:
1. Key remapping takes precedence over redirecting.
2. Disabling the redial, conference, or transfer features on the phone also disables the redirection of these keys.

The following URI configuration parameters control the redirection of the keys and the voicemail option:

- **redial script**
- **xfer script**
- **conf script**
- **icom script**
- **voicemail script**

An Administrator can configure the XML key, redirection URI parameters using the configuration files only.

Configuring XML Redirection of the Redial, Xfer, Conf, and Icom Keys, and the Voicemail Option

Use the following procedure to configure XML redirection of the Redial, Xfer, Conf, and Icom keys, and the Voicemail option.

Configuration Files

For the specific parameters you can set in the configuration files, see Appendix A, the section, "XML Key Redirection Settings (for Redial, Xfer, Conf, Icom, Voicemail)" on page A-242.

Options Key Redirection

The IP phones allow the redirecting of the Options Key to an XML script. This allows the server to provide the phone with available options, rather then accessing them from the phone-side. You access the Options Key XML script by pressing the Options Key. You can still access the Options Menu from the phone-side by pressing and holding the Options key to display the phone-side Options Menu.

The following URI configuration parameter controls the redirection of the Options Key:

- **options script**

IMPORTANT NOTES

- If no Options URI script is configured, the local Options Menu on the phone displays as normal.
- If you configure password access to the Options Menu, this password is required when accessing the local Option Menu, but is not required for the Options Key redirection feature.
- Pressing the Options Menu for redirection from the server does not interfere with normal operations of the phone (for example, pressing the options menu when on a call does not affect the call).
- If the phone is locked, you must unlock the phone before accessing the Options Menu redirect feature. After pressing the Options Key, the phone displays a screen that allows you to unlock the phone before continuing.

An Administrator can configure the XML Options Key, redirection URI parameter using the configuration files only.

Configuring XML Redirection of the Options Key

Use the following procedure to configure XML redirection of the Options key.

Configuration Files

For the specific parameter you can set in the configuration files, see Appendix A, the section, "Options Key Redirection Setting" on page A-244.

XML Applications and Off-Hook Interaction

A feature on the IP phone allows you to specify whether the phone is prevented from going into the off-hook/dialing state when the handset is off-hook and the call ends.

By default, the phone behaves as follows:

You are in a call using the handset and the phone displays an XML application. The far-end terminates the call, and a new XML application gets pushed/pulled onto the display. Since the handset is off-hook and in idle mode, the "offhook idle timer" starts. When this timer expires, the phone applies dial tone and moves to the off-hook/dialing state, which then destroys the XML application that was being displayed.

With the "off-hook interaction" feature you can set an "**auto offhook**" parameter that determines whether or not the phone is prevented from entering the off-hook/dialing state, if the handset is off-hook and the call ends.

An Administrator can enable (allow phone to enter the off-hook dialing state) or disable (prevent the phone from entering the off-hook dialing state) using the "**auto offhook**" parameter in the configuration files only.

Configuring the Off-Hook Interaction Feature

Use the following procedure to configure the XML application and off-hook interaction feature.

Configuration Files
For the specific parameter you can set in the configuration files, see Appendix A, the section, "Off-Hook and XML Application Interaction Setting" on page A-244.

XML URI for Key Press Simulation

The Phones provide a feature that allow an XML Developer or Administrator to define XML Key URIs that can send key press events to the phone, just as if the physical hard key, softkey, or programmable key were pressed on the phone.

For more information about this feature, see Chapter 6, the section, "XML URI for Key Press Simulation" on page 6-51.

XML Override for a Locked Phone

The IP phones have a feature that allows a locked phone to be overridden when an XML application is sent to the phone. This feature also allows you to still use any softkeys/ programmable key/Extension Module Keys applicable to the XML application even though the phone is locked. However, any keys NOT associated with the XML application cannot be used when the phone is locked.

Also, XML Get Requests override the locked feature on the phone so that any softkey pressed by the user that initiates a Get Request continues to get sent.

To allow the overriding of the locked phone for XML applications, the System Administrator must enter the following parameter in the configuration files:

- **xml lock override**

Configuring XML Override for a Locked Phone using the Configuration Files

Configuration Files
For the specific parameter you can set in the configuration files, see Appendix A, the section, "XML Override for a Locked Phone Setting" on page A-245.

Note: A System Administrator can also lock and unlock a remote phone using the "**lock**" and "**unlock**" commands with the **AastraIPPhoneExecute** object in an XML application. For more information about this feature, contact Aastra Customer Support regarding the "Aastra XML Development Guide."

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-192.

Configurable Indication of Terminated Calls

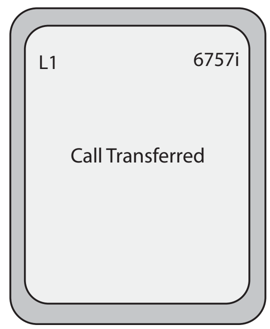
An Administrator can configure a parameter called, “**far-end disconnect timer**” which allows you to enable or disable whether or not the near-end phone displays a disconnected screen (“Call Terminated”) when the far-end hangs up. An audible busy signal is also heard on the phone. If enabled with a maximum time interval value, this parameter also specifies the interval of time that the busy signal is audible.

You can enable/disable this new parameter using the configuration files only.

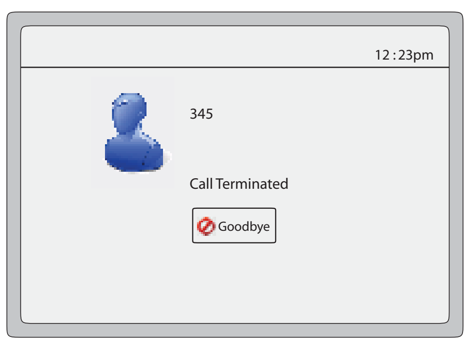
The following are call terminated screens that display for each type of Aastra phone.

Call Terminated Screens

3-Line Display Screens



5-Line Display Screens



6739i Display Screens

The following table identifies when a call terminated screen displays on the phone for different scenarios.

IF	THEN
1 line active and far-end disconnects,	the line in focus: • displays disconnected screen. • plays busy tone. • displays "Call Terminated" message on the screen. the line not in focus: • plays busy tone.
2 or more lines active, and far-end disconnects,	the line in focus: • displays disconnected screen. • plays busy tone. • displays "Call Terminated" message on the screen for 5 seconds. When 5 second times out: - the busy tone stops - the disconnected screen disappears.
2 or more lines active, and a line NOT in focus is disconnected by the far-end,	• no busy tone plays • no disconnected screen displays • no "Call Terminated" message displays
An incoming call comes in on the line in focus that has a disconnected screen displaying,	• the line in focus with no calls on hold: - displays a ringing screen • the line in focus WITH calls on hold: - flashes its' Line LED
An incoming call comes in on another line (NOT in focus), and the disconnected screen is displaying on the line in focus,	• the disconnected screen no longer displays on the line in focus.
A phone application is NOT in focus,	• busy tone plays • no disconnected screen displays • When the phone application in focus on screen stops: - busy tone plays - disconnected screen displays

Note: This "indication of terminated calls" feature does not affect parked calls on the phone or the conference call feature.

Configuring Indication of Terminated Calls

You can enable or disable whether or not the phone displays an indication of a terminated call using the parameter, "**far-end disconnect timer**." This parameter also specifies the maximum time interval that the busy tone is audible on the phone. You can configure the indication of terminated calls 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, "Disable User Login to Aastra Web UI" on page A-195.

Handling Call Termination Screens on the Phone UI

The following procedure describes how to handle the call terminated screens on your phone.

***For all phones*:**

	IP Phone UI
	• Press the Goodbye key. The busy tone stops and the call terminated screen no longer displays. or Select any Line key. The busy tone stops and the call terminated screen no longer displays. A dial screen displays.

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`

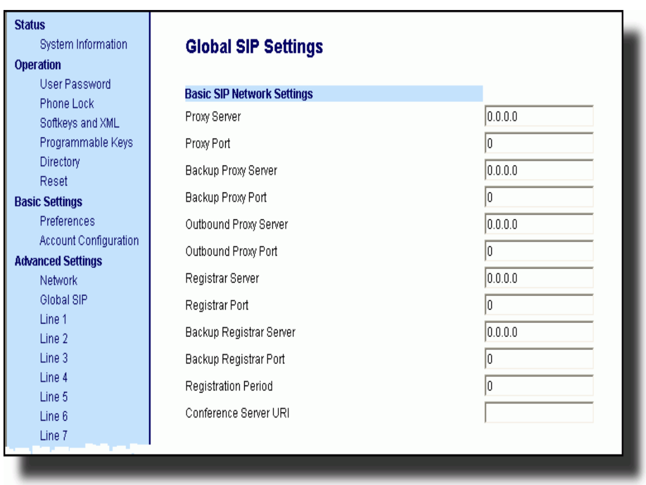
Configuration Files

For specific parameters you can set in the configuration files, see Appendix A, the section, "Centralized Conferencing Settings" on page A-99.

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.** 

Aastra Web UI

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)

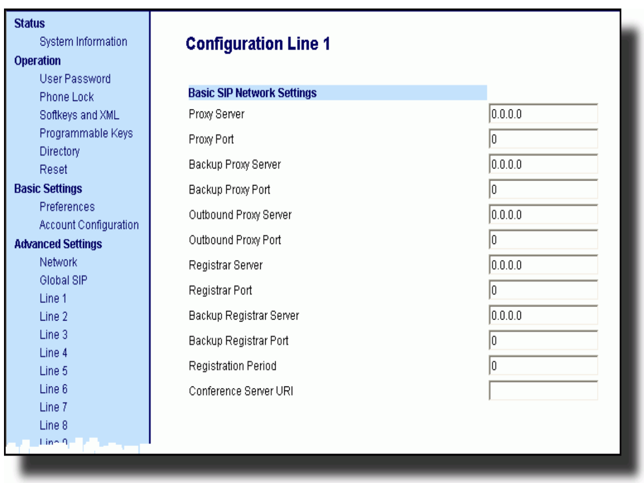
    By setting this field to **conf** or **Conference**, you specify conf@<proxy_server _address>:<proxy_port>. For example, if the proxy server address is 206.229.26.60 and the port used is 10060, then by setting this parameter to **conf**, you are specifying the following: conf@206.229.26.60:10060.
  - — 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>

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

Per-Line Configuration

1 Click on **Advanced Settings->Line <#>->Basic SIP Network Settings**



	Aastra Web UI
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.

“SIP Join” Feature for 3-Way Conference

The IP Phones support RFC 3911 which allows an additional caller to join an active call between two parties if the caller knows the dialog information. This feature begins a conference using a join header as described in RFC 3911.

The “SIP Join“ feature provides the following:

- Security via the whitelist (which is a feature that already exists on the phone).
- Initiates an offhook action uri when it is answered.
- Initiates an onhook action uri at call termination.
- Creates a caller list entry.

This feature is disabled by default. You can enable the “SIP Join” feature by setting the “sip join support” parameter in the configuration files.

Limitations of the “SIP Join” Feature

The following are limitations of the “SIP Join” feature:

- Not applicable to a conference call already in progress.
- Not applicable to a CT handset that has two active calls.
- Not applicable to a phone mixing RTP.
- Allows secondary parties to join calls if they can determine the dialog parameters. In order to provide security, it is recommended that the Administrator configure the SIP whitelist.
- Not applicable while the active call between two parties is in the early dialog state.

Configuring the "SIP Join" Feature Using the Configuration Files

You use the following parameters to configure the "SIP Join feature in the configuration files:

sip join support

Configuration Files

For specific parameters you can set in the configuration files, see Appendix A, the section, "SIP Join Feature for 3-Way Conference" on page A-101.

Conference Ends and Transfers Remaining Parties

On the IP Phones, a Host can drop from a 3-way conference and let the other two parties remain connected by pressing a **Xfer** key.

Notes:

1. The 3-line LCD display phones (6730i, 6731i, 6753i, and 9143i) and cordless handsets (CT handsets) do not support scrolling between lines.

2. The phones MUST have a **Xfer** key (hardkey, or configured programmable or softkey) on the phone.

Reference

For more information about using the 3-way conference and the Xfer key, see the IP Phone Model-specific User Guide.

Limitations

The following are limitations of this feature:

- The phone MUST have a Xfer key configured on the phone (programmable key or softkey)
- Asterisk SLA does not support this feature.
- A "xfer script" can override this feature if a custom application is created and control is given to the server-side.
- The Caller ID of the dropped Host still displays on the remaining parties' phones.

Transferring Two Existing Calls

The IP Phones allow Users to perform a transfer of a call when there are currently two active calls on the phone. For example, on the User's phone, there are two active calls - Call A and Call B. Call B is on hold. The User can perform the following to transfer Call A to Call B:

1. Press **Xfer** key. A new line opens.
2. Press or scroll to the **Line** where Call B is on hold.
3. Press **Xfer** key.

Call A is transferred to Call B.

Reference

For more information about using the Xfer key to transfer two existing calls, see the IP Phone Model-specific User Guide.

Limitations

The following are limitations of this feature:

- If a user presses the **Xfer** key but dosen't continue to transfer the call by pressing a **Line** key, there is no indication on the phone that the transfer didn't take place.
- Once the transfer takes place, the phone which has been transferred to, still displays the original call.
- For this feature, 3-line LCD display phones (6730i, 6731i, 6753i, 9143i) do not support scrolling between lines.

Authentication Support for HTTP/HTTPS Download Methods, used with Broadsoft Client Management System (CMS)

The IP phones support HTTP/HTTPS digest authentication as defined in RFC 2617. (The HTTP client supports digest authentication; the HTTP server does not; the HTTP server supports basic authentication). This feature allows the phones to interoperate with Broadsoft's CMS phone configuration tool.

Using the configuration files, you can enable/disable the following parameter to display a LOG IN softkey which allows the HTTP/HTTPS server to perform digest authentication:

- **http digest force login -** specifies whether or not to display the LOG IN softkey on the IP Phone UI screen. Valid values are **0** (disabled) or **1** (enabled). Default is **0** (disabled).

If the "**http digest force login**" parameter is set to **1** (enabled), after the phone boots, the LOG IN softkey displays on the phone's LCD. If the user presses this softkey, a username/password screen displays, allowing the user to enter the configured username and password that is sent to the HTTP/HTTPS server for digest authentication by the server. By default, username is "**aastra**" and password is "**aastra**". You can enter the username and password in two ways:

- Using the configuration files, you can change the default values for the following parameters:
  - **http digest username -** specifies the username to use for HTTP/HTTPS digest authentication.
  - **http digest password -** specifies the password to use for HTTP/HTTPS digest authentication.
- By enabling the "**http digest force login**" parameter (setting to 1) **-** the phone displays the LOG IN key so the user can enter the default username/password via the IP Phone UI.

Configuring Broadsoft CMS Support via the Configuration Files

Configure Broadsoft CMS support on the IP Phone using the following parameters in the configuration files:

- **http digest force login**
- **http digest username**
- **http digest password**

Configuration Files

For specific parameters you can set in the configuration files, see Appendix A, the section, "HTTP/HTTPS Authentication Support for Broadsoft CMS" on page A-102.

Using the IP Phone when Broadsoft CMS is Enabled

If you enable the HTTP/HTTPS digest authentication feature, the phone behaves as follows with the BroadSoft CMS tool:

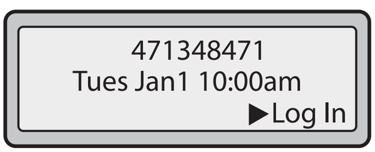
1. Factory default the phone.
2. Configure the HTTP or HTTPS server (specify the HTTP or HTTPS server, path, and port).
3. Restart the phone.

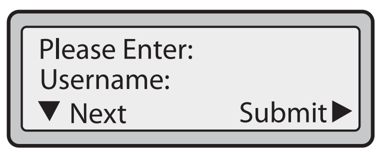
   The first time the phone reboots, the phone is challenged by the server. The phone sends the default username of “**aastra**” and the default password of “**aastra**” to the server.

   The server sends the default profile to the phone. This profile includes the information “**http digest force login: 1**”.When the phone receives the profile, it displays the “**Log In**” key on the IP Phone UI’s idle screen.
4. Press the “**Log In**" key to displays the username/password screen.

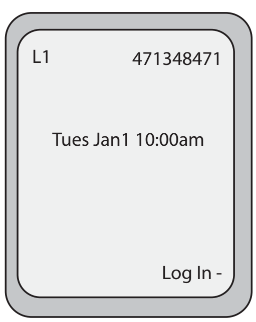
Note: On the 8 and 11-Line LCD phones, you use the **Log In** softkey to log in. On 3-Line LCD phones, you press the **right arrow** key to log in.

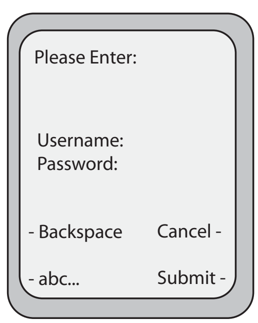
3-Line LCD Phones Login Screens





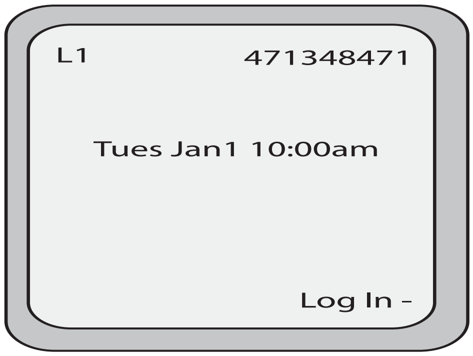
8 and 11-Line LCD Phones Login Screens



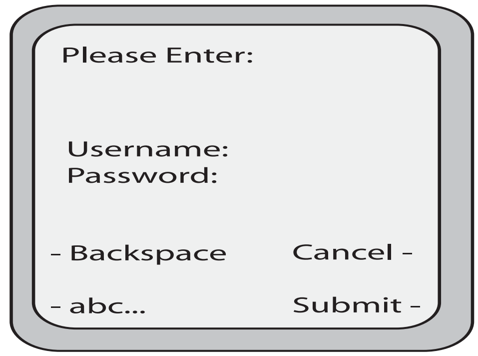


6739i Phone Login Screen

Login Screen



Username/Password Screen



5. Enter a username in the “**Username**” field (up to 40 characters) and a password in the “**Password**” field (up to 20 characters).

Note: The “**Username”** and “**Password”** fields accept special characters, such as, @, #, %, =, _, etc. You can also specify domain names in the **Username** field (i.e., user@domain).

---

6. After entering the username and password, press **Submit**.
The phone attempts to authenticate with the server. If successful, the phone reboots and loads the user configuration. If unsuccessful, the phone displays “*Authentication Failed*”.

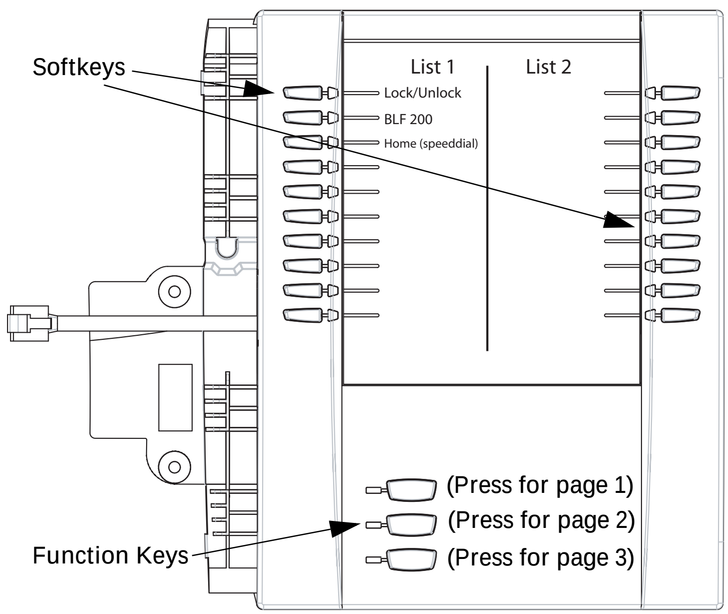
Customizing the Display Columns on the M675i Expansion Module

The M675i 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:

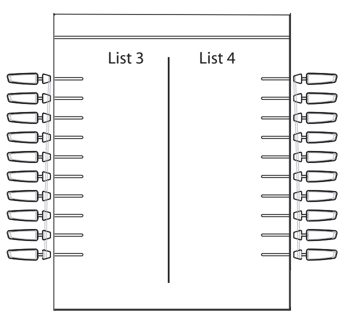
Page 1 "List 1" and "List 2"

Page 2 "List 3" and "List 4"

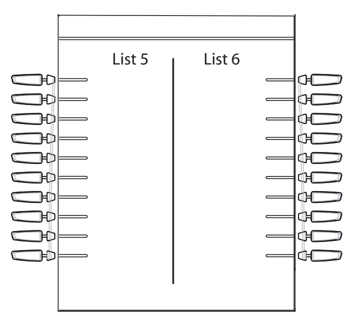
Page 3 "List 5" and "List 6"



Page 1 Screen



Page 2 Screen



Page 3 Screen

To use the M675i, 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 M675i 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)

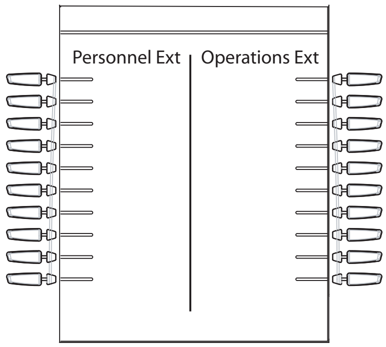
- expmod3page1left
- expmod3page1right
- expmod3page2left
- expmod3page2right
- expmod3page3left
- expmod3page3right

Example

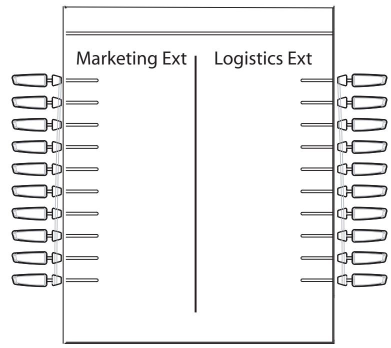
The following is an example of configuring Expansion Module 1 column headings.

```
expmod1page1left: Personnel Ext
expmod1page1right: Operations Ext
expmod1page2left: Marketing Ext
expmod1page2right: Logistics Ext
expmod1page3left: Engineering Ext
expmod1page3right: Shipping Ext
```

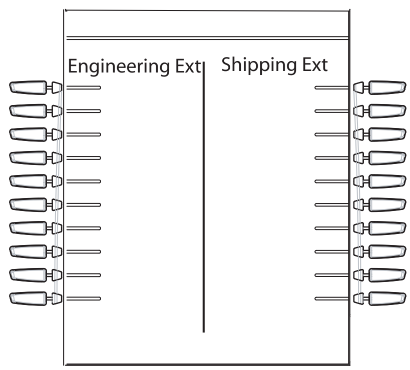
These settings display to the Expansion Module as shown in the following illustrations.



Page 1 Screen



Page 2 Screen



Page 3 Screen

Customizing the M675i Expansion Module Column Display.

Configuration Files

For specific parameters you can set in the configuration files, see Appendix A, the section, "Customizing M675i Expansion Module Column Display" on page A-236.

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
TR-069 Support	The IP Phones support the Technical Report (TR) 069 Protocol, a Protocol that provides the communication between Customer-Premises Equipment (CPE) (like the IP Phones) and Auto Configuration Servers (ACS) over DSL/broadband connections.
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 (6757i CT and 9480i CT only)	Allows you to enable or disable a single call restriction between the 6757i 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.
As-Feature-Event Subscription	Allows you to enable or disable a specific line on the phone with the BroadSoft's server-side DND, CFWD, or ACD features.
Blacklist Duration	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.
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.

Feature	Description
802.1x Support	Allows you to enable or disable the 802.1x Protocol support on the IP Phones.
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.
GRUU and sip.instance Support	The IP phones provide GRUU support using draft-ietf-sip-gruu-15. A sip.instance is added to all non-GRUU contacts.
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.
Configurable DNS Queries	Allows you to specify the Domain Name Service (DNS) query method to use when the phone performs a DNS lookup.
Ignore Out of Sequence Errors	Allows you to configure the phone to ignore CSeq number errors on all SIP dialogs on the phone.
“Early-Only” Parameter in Replaces Header RFC3891	The phones support the “early-only” parameter in the “Replaces” header according to RFC3891.
Switching Between Early Media and Local Ringing	The phones support switching between early media and local ring tone.
Enable Microphone During Early Media	Allows you to enable or disable the microphone during early media.
“Call-Info” Header to 200ok Responses for Shared Call Appearance (SCA) Lines	In Release 2.6 and up, a “Call-Info” header is included in the 200ok response to an INVITE, RE-INVITE, and UPDATE messages for SCA lines.
Reason Header Field in SIP Message	The IP Phones support the receiving of the Reason Header Field in a SIP CANCEL message, as described in RFC3326.
Configurable “Allow” and “Allow-Event” Optional Headers	On the IP Phones, an Administrator can enable or disable whether or not the optional “Allow” and “Allow-Events” headers are included in the NOTIFY message from the phone.
Configurable SIP P-Asserted Identity (PAI)	The IP Phones support a private extension to SIP for Asserted Identity within trusted networks (as defined in RFC 3325).
Configurable Route Header in SIP Packet	The IP Phones support a parameter that enables or disables the addition of the Route header in a SIP packet.
Configurable Compact SIP Header	The phones provide a feature that allows an Administrator to shorten the length of a SIP packet by using the compact form. This feature is in accordance with Compact SIP Headers defined in RFC 3261.
Reject INV or BYE when Unsupported Value in REQUIRE Header	The IP Phones support a parameter that allows you to enable or disable the rejection of an INV or BYE with a “420 Bad Extension” if the INV or BYE contains an unsupported value in the REQUIRE header.
XML URI for Key Press Simulation	The phones provide a feature that allow an XML Developer or Administrator to define XML Key URIs that can send key press events to the phone, just as if the physical hard key, softkey, or programmable key were pressed on the phone.

TR-069 Support

The IP Phones support the Technical Report (TR)-069 Protocol. This Protocol is a bi-directional HTTP based protocol that provides the communication between Customer-Premises Equipment (CPE) (like the IP Phones) and Auto Configuration Servers (ACS) over DSL/broadband connections. It includes both a safe auto configuration and the control of other CPE management functions within an integrated framework.

Service providers can, through TR-069, use one common platform to manage (through the internet) all of their customer premise devices, regardless of the device or manufacturer. If TR-069 is enabled on the phones, when the remote ACS boots the phones, they contact the ACS and establish the configuration automatically.

In addition to configuring the phone with TR-069, you can also do the following:

- Reboot the phone
- Reset to factory defaults
- Update the firmware of the device
- Backup/restore configuration
- Upload the log file

Reference

For more information about TR-069, see the *Aastra TR-069 Configuration Guide*.

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:

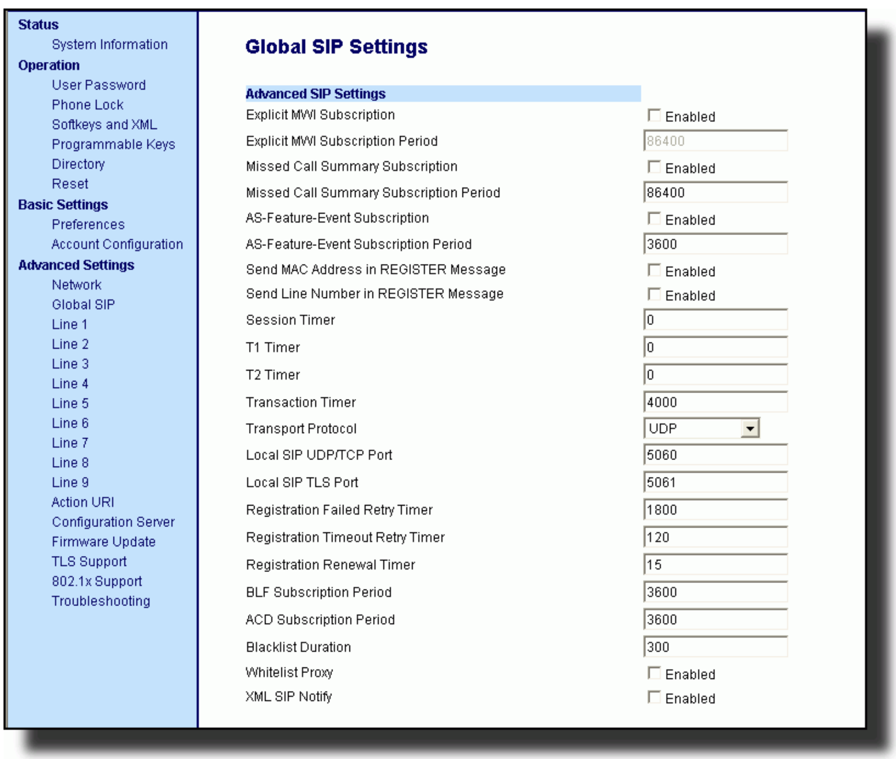
- **sip send mac**
- **sip send line**

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-238.

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-238.

SIP Message Sequence for Semi-Attended Transfer

The SIP message sequence for a Semi-Attended Transfer allows the transferor to start the transfer while the target phone is still ringing.

The following parameter supports different behaviors of a semi-attended transfer:

sip refer-to with replaces

This parameter is configurable via the configuration files only.

The combination of this new parameter (**"sip refer-to with replaces"**) and the existing parameter (**"sip cancel after blind transfer"**) determines how the semi-attended transfer is completed. The following table shows how the old and new parameters work together:

IF	THEN
The "sip cancel after blind transfer" parameter is set to 0 and the "sip refer-to with replaces" parameter is set to 0	The phone sends CANCEL before REFER for semi-attended transfer.
The "sip cancel after blind transfer" parameter is set to 1 and the "sip refer-to with replaces" parameter is set to 0	The phone sends CANCEL after REFER for semi-attended transfer.
The "sip cancel after blind transfer" parameter is set to 0 and the "sip refer-to with replaces" parameter is set to 1	The phone sends REFER with Replaces for semi-attended transfer and NO CANCEL.
The "sip cancel after blind transfer" parameter is set to 1 and the "sip refer-to with replaces" parameter is set to 1	The phone sends REFER with Replaces for semi-attended transfer and NO CANCEL.

On the transferor phone, the REFER request will always be sent to the transferee.

Configuring SIP Message Sequence for Semi-Attended Transfer

Configuration Files

For the specific parameter you can set in the configuration files for customizing the semi-attended transfer method, see Appendix A, the section, "Semi-Attended Transfer Settings" on page A-238.

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 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-238.

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 Settings." on page A-239.

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 (6757i CT and 9480i CT only)

On the 6757i CT and 9480i CT, an administrator can enable or disable a single call restriction between the 6757i CT and 9480i CT base unit and a call server.

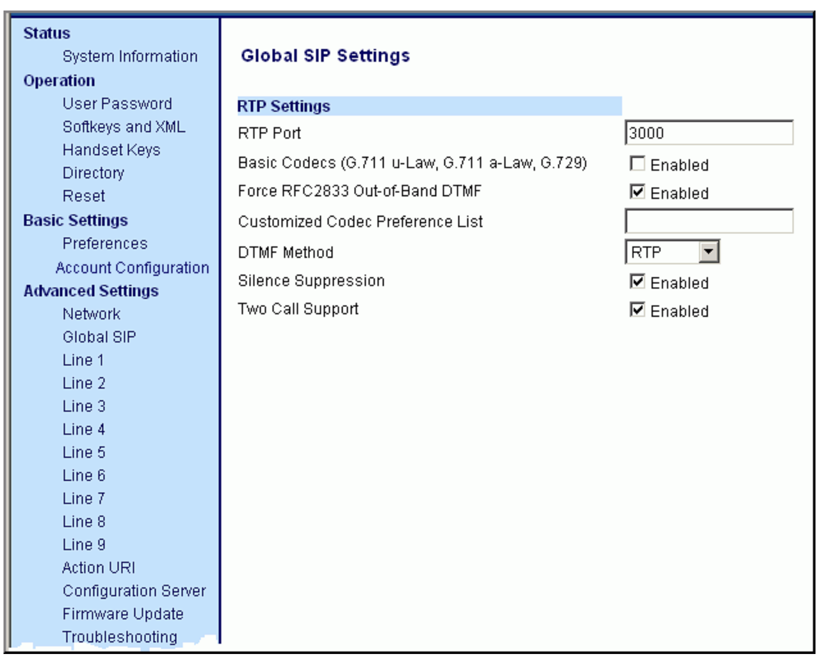
When this feature is enabled (set to 1), you can make separate active calls from the 6757i CT and 9480i CT base unit and from the cordless handset. If this feature is disabled (set to 0), only one call can be active at a time either from the base unit or from the handset. When this feature is disabled, 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 6757i CT and 9480i CT, see Appendix A, the section, "Single Call Restriction Setting" on page A-240.

<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 Advanced Settings->Global SIP->RTP Settings.



</td></tr>
<tr><td>2</td><td>Enable the "Two Call Support" field by checking the check box.
(Disable this field by unchecking the box).</td></tr>
</table>

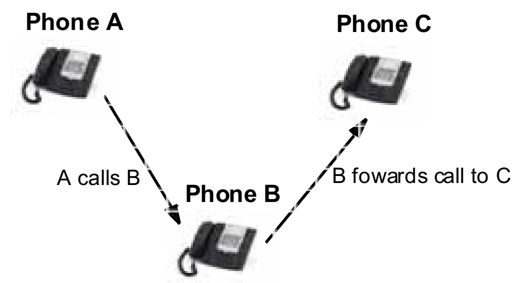
Aastra Web UI

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**.

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 this feature, 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

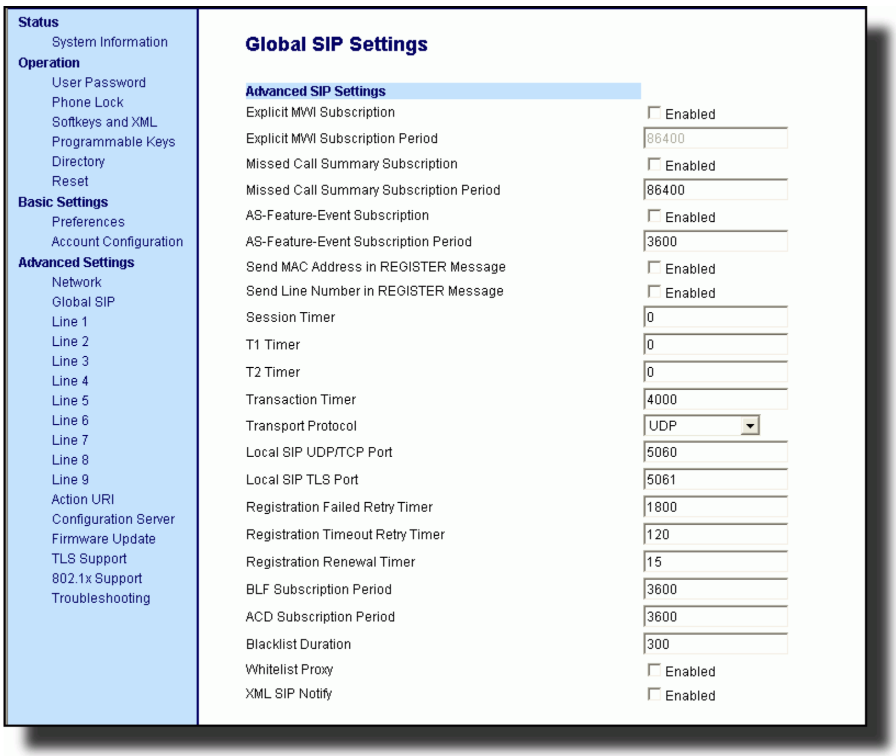
- **`sip lineN missed call summary subscription`**

Configuration Files

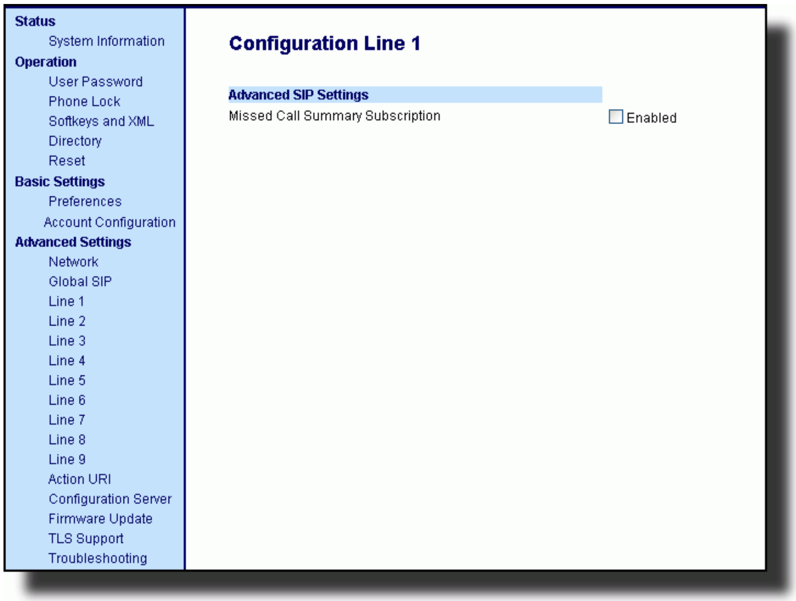
For the specific parameters you can set in the configuration files for Missed Call Summary Subscription, see Appendix A, the section, "Missed Call Summary Subscription Settings" on page A-110.

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.** 
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.

<table>
<tr><th colspan="2">Aastra Web UI</th></tr>
<tr><td colspan="2">Per-Line Configuration</td></tr>
<tr><td>1</td><td>Click on Advanced Settings->Line <N>->Advanced SIP Settings.



</td></tr>
<tr><td>2</td><td>Click [Save Settings] to save your changes.</td></tr>
</table>

As-Feature-Event Subscription

The IP phones support server-side Do Not Disturb (DND), Call Forward (CFWD), and Automatic Call Distribution (ACD) feature events.This feature is called “as-feature-event” and works with the DND, CFWD, and ACD keys.

Notes:
1.The DND, CFWD, and ACD server-side feature is not applicable to the CT handset.

This feature is configurable using the configuration files or the Aastra Web UI.

How it Works on the Phone UI

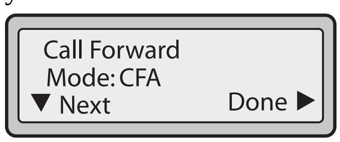
When you enable the "as-feature-event" on the phone, AND you activate a DND, CFWD, and/or ACD key, pressing the key performs as follows:

- If the key is configured for an account on the phone, the server applies DND, CFWD or ACD to that account. (For information about CFWD and DND account configuration, see Chapter 3, the section, "Account Configuration" on page 3-33).
- If the key is "custom" configured, a screen displays on the phone allowing the user to choose the account to apply DND or CFWD. (For information about CFWD and DND custom configuration, see Chapter 3, the section, "Account Configuration" on page 3-33).
- A solid "Message Waiting Indicator" (MWI) indicates if one line/account has DND or CFWD enabled, and the LED next to the DND/CFWD key is ON. A status displays on the LCD that indicates the status of the line in focus (for example, the status of CFWD could be "Call Forward Busy" (CFWDB) or "Call Forward No Answer" (CFWDNA).

Note: If the ACD key is configured on the phone, and the "as-feature-event" is not enabled, the phones uses the ACD icons and LED behavior from a Sylantro server instead.

When you press the DND, CFWD, or ACD key, only one attempt is made to enable/disable the "as-feature-event" feature on the server. The message "*Trying*" displays on the phone's LCD after pressing the key. If the attempt is successful, the idle screen displays. If the attempt is unsuccessful, the message "*Failed*" displays. The user can press the softkey again to re-attempt the feature if required.

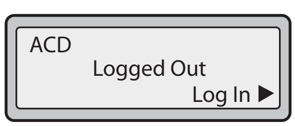
The following screen displays on the IP Phone UI for server-side call forwarding:

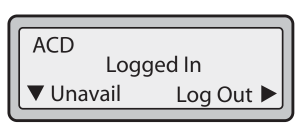


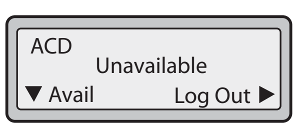
For server-side ACD, when you press the ACD softkey, the screen that displays is dependant on the state of the ACD subscription. Possible state for ACD are::

- **Logged Out -** User has the option of logging in.
- **Logged In -** User has the option of logging out or making the phone unavailable.
- **Unavailable -** User has the option of logging out or making the phone available.

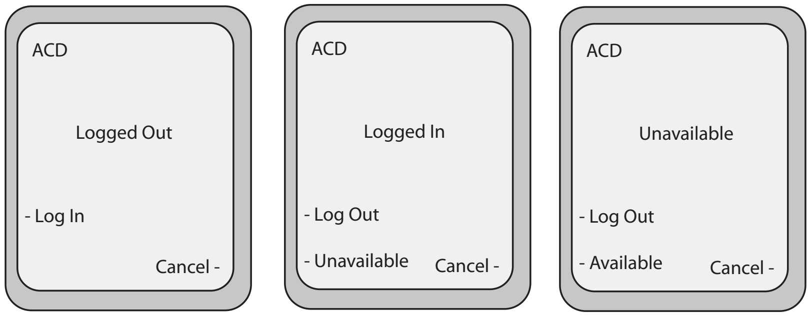
ACD Screen For 3-Line LCD Phones:



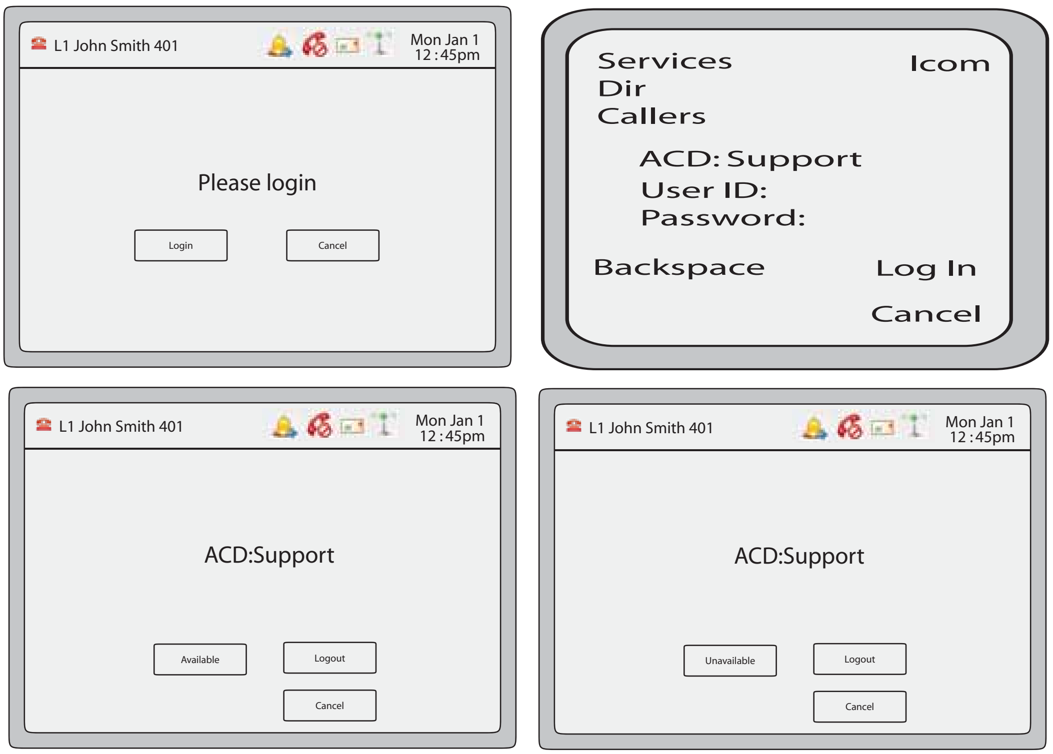




ACD Screen for 8 and 11-Line LCD Phones



ACD Screen for 6739i Phone



Note: IMPORTANT!

If DND and CFWD are configured to use "**Account**" mode on the IP Phone, pressing the DND and CFWD keys apply to the account in focus as described in Chapter 3, the section, "Account Configuration" on page 3-33.

If ACD is configured on the phone, the ACD softkey applies to the line for which the key is configured. The ACD softkey must be configured for the first line of an account. For example, if account 2 has line 3 and line 4 you must configure the ACD softkey for line 3.

Configuring As-Feature-Event Subscription Using the Configuration Files

If the phone-side features of the DND, CFWD, and ACD keys are enabled, the phone uses the existing parameter values for these keys. If the server-side features are enabled, the phone saves the state of the features from the server on the phone.

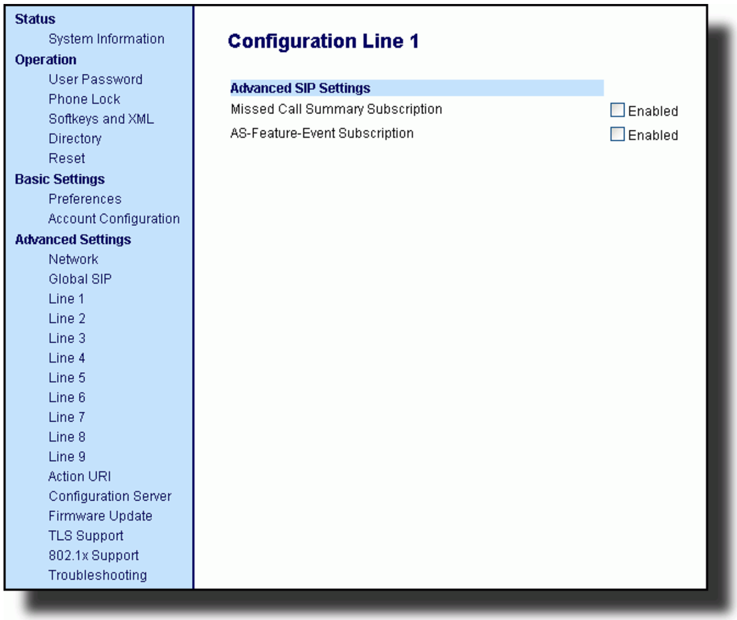
Use the following parameters to enable/disable the server-side "as-feature-event" on the IP Phone:

- **sip lineN as-feature-event subscription**
- **sip as-feature-event subscription period**

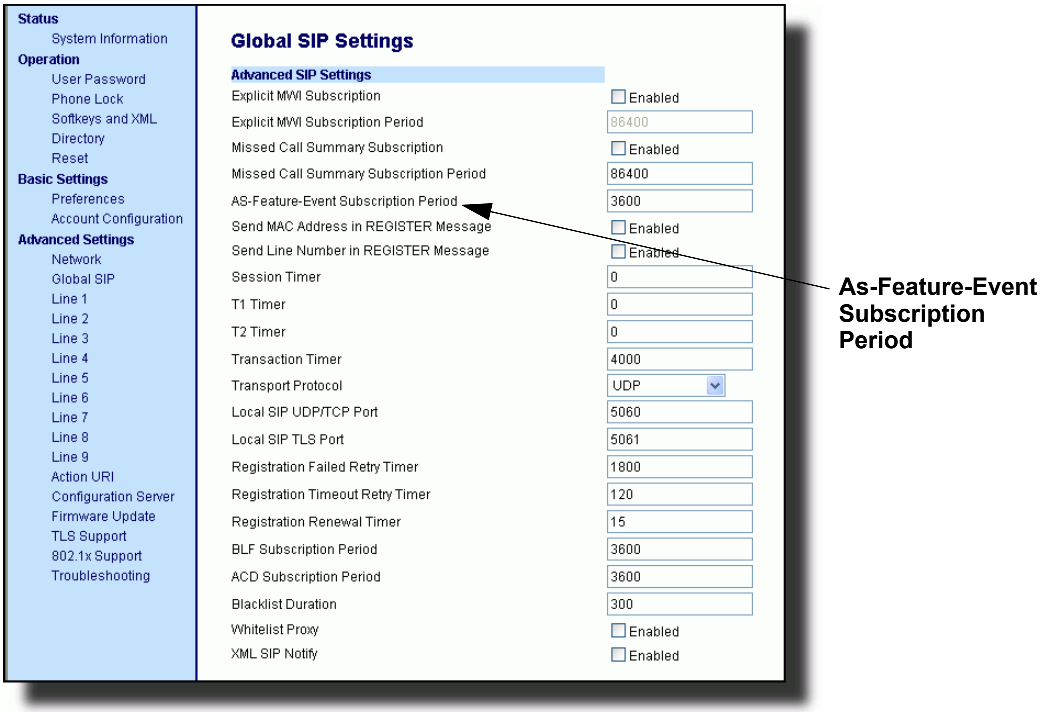
Configuration Files
For the specific parameters you can set in the configuration files, see Appendix A, the section, "As-Feature-Event Subscription Settings" on page A-113.

Configuring As-Feature-Event Subscription Using the Aastra Web UI

Use the following procedure to enable/disable the server-side “as-feature-event” on the IP Phone using the Aastra Web UI.

	Aastra Web UI
1	Click on **Advanced Settings->LineN->Advanced SIP Settings**. 
2	In the “Advanced SIP Settings” section, enable the "**As--Feature-Event Subscription**" field, by checking the check box. (Disable this field by unchecking the box).
3	Click Save Settings to save your changes.

<table>
<tr><th colspan="2">Aastra Web UI</th></tr>
<tr><td>4</td><td>Click on Advanced Settings->Global SIP->Advanced SIP Settings.



</td></tr>
<tr><td>5</td><td>In the “Advanced SIP Settings” section, in the “As-Feature-Event Subscription Period” field, enter the amount of time, in seconds, that the phone waits after sending a SUBSCRIBE, to receive a NOTIFY response from the server side.
Default is 3600. Valid values are</td></tr>
<tr><td>6</td><td>Click Save Settings to save your changes.</td></tr>
</table>

Blacklist Duration

The Blacklist Duration feature 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 (5 minutes). If you set the duration to 0 seconds, then you disable the blacklist feature.

Configuring Blacklist Duration Using the Configuration Files

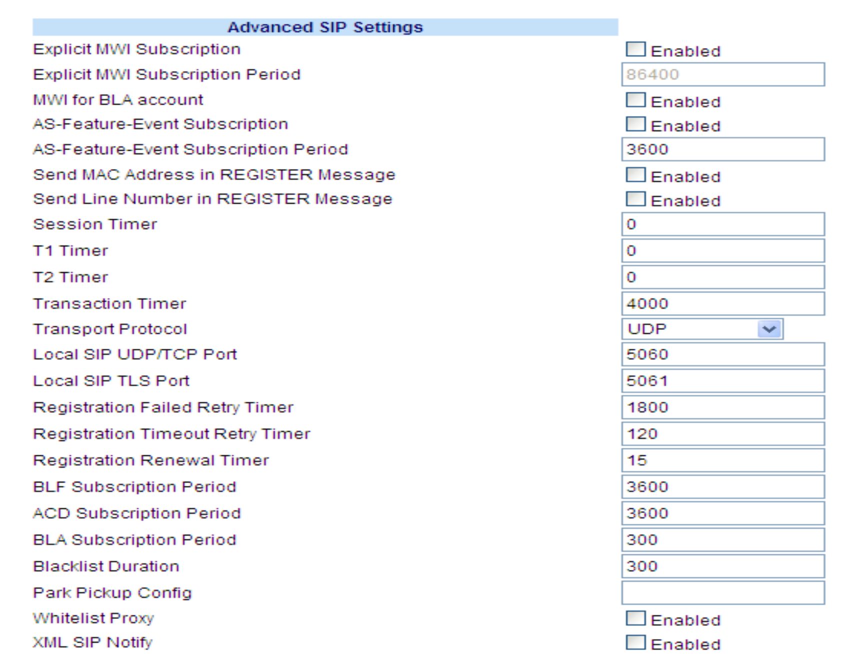
Use the following parameter to configure the Blacklist Duration in the configuration files:

- **`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, "Blacklist Duration Setting" on page A-241.

Configuring a Server Blacklist Using the Aastra Web UI

You use the following procedure to configure Blacklist Duration 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>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.</td></tr>
<tr><td>3</td><td>Click Save Settings to save your changes.</td></tr>
</table>

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:

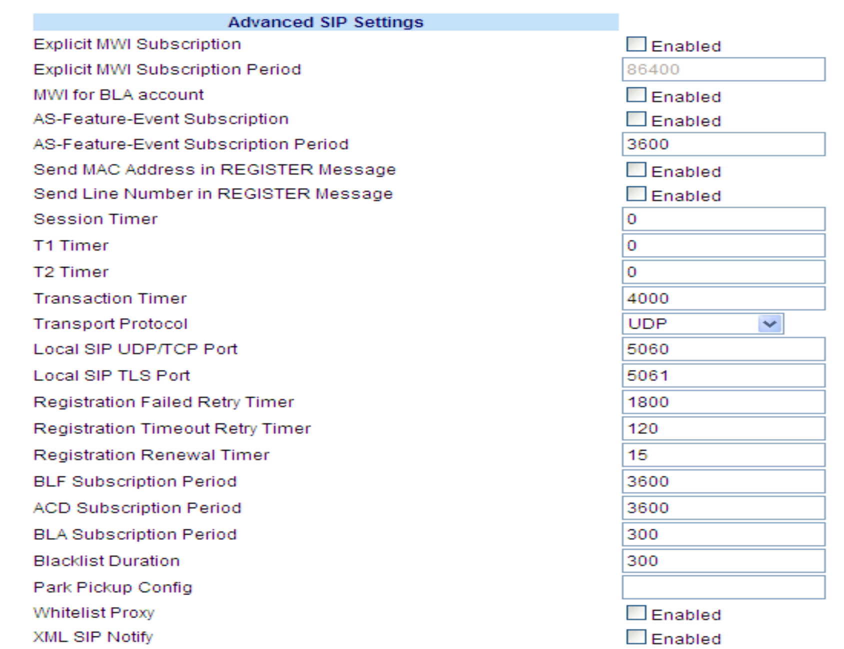
- **`sip whitelist`**

Configuration Files
For the specific parameters you can set in the configuration files for setting Whitelist Proxy, see Appendix A, the section, "Whitelist Proxy Setting" on page A-241.

Configuring Whitelist Proxy Using the Aastra Web UI

Use the following procedure to configure the Whitelist Proxy feature using the Aastra Web UI.

	Aastra Web UI
1	Click on **Advanced Settings->Global SIP->Advanced SIP Settings**



2	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.
3	Click Save Settings to save your changes.

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 the following 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 SIP for Asserted Identity within trusted networks (as defined in RFC 3325).

Note: The phones support PAI header in the UPDATE message, according to draft-ietf-sipping-update-pai-00. This feature is always enabled.

If an UPDATE is received with a PAI header from a trusted source, the phone updates the display with this information. The phone ignores any PAI received from untrusted entities

---

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`**

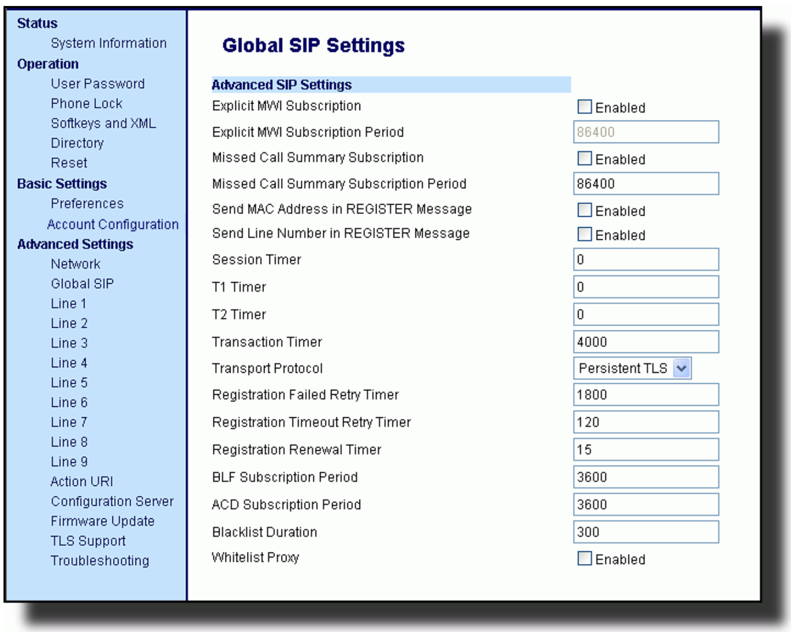
Configuration Files

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-114.

Configuring TLS Using the Aastra Web UI

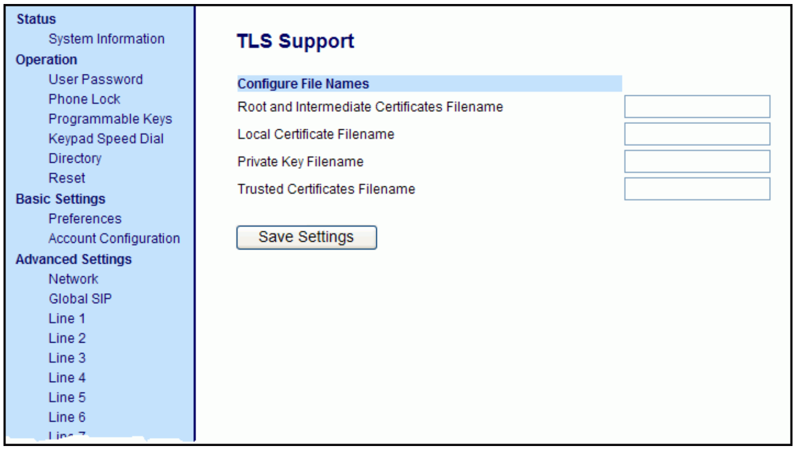
To configure TLS using the Aastra Web UI, you must enable TLS or Persistent TLS first. Then you must define the TLS certificate file names that you want the phone to use. Use the following procedure to configure TLS using the Aastra Web UI.

<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>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.</td></tr>
<tr><td>3</td><td>Click [Save Settings] to save your changes.</td></tr>
</table>

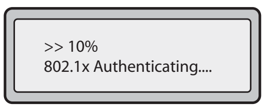
	Aastra Web UI
4	Click on **Advanced Settings->TLS Support.** 
5	Enter the certificate file names and the private key file name in the appropriate fields. The Root and Intermediate Certificate files contain one root certificate and zero or more intermediate certificates which must be placed in order of 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  to save your changes.

802.1x Support

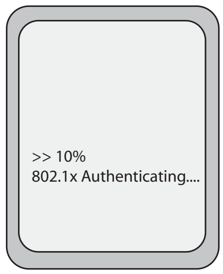
The IP phones support the IEEE 802.1x protocol. The 802.1x protocol is a standard for passing Extensible Authentication Protocol (EAP) over a wired or wireless Local Area Network (LAN).

The 802.1x protocol on the IP phone facilitates media-level access control, and offers the capability to permit or deny network connectivity, control LAN access, and apply traffic policy, based on user or endpoint identity. This feature supports both the EAP-MD5 and EAP-TLS protocols.

If 802.1x on the phone is enabled, the following screen displays during startup of the phone.



3-Line LCD Displays



8 and 11-Line LCD Displays

6739i Display

If the 802.1x failed to authenticate with the server, the phone continues it's normal startup process using DHCP. However, the network port on the phone may or may not be disabled, depending on the switch configuration.

Certificates and Private Key Information

- If the certificates and private key are NOT stored in the phone:
  - — the phone connects to an open unauthenticated VLAN and the certificates are downloaded.
    or
  - — the phone connects using EAP-MD5 to a restricted VLAN and the certificates are downloaded.
- If the certificates and private key ARE stored in the phone, the phone uses them during the authentication process.
- If the phone uses EAP-TLS for successful authentication, after the phone reboots, it downloads the latest certificates and private key files to the phone.
- The private key uses AES-128 to encrypt the private key file.

- Switch Supplicant Mode - The switch supports the following 2 modes:
  - **Single supplicant -** This mode enables the port once any machine connected to this port is authenticated. For security reasons, the IP phone has the option to disable the pass-through port.
  - **Multiple supplicants -** Using this mode, the switch can support multiple clients connected to same port. The switch distinguishes between the clients based on their MAC address.
- Factory default and recovery mode deletes all certificates and private keys, and sets the EAP type to **disabled**.

You can configure the 802.1x feature on the IP phone using the configuration files, the IP Phone UI, or the Aastra Web UI.

Note: If configuring 802.1x using the IP Phone UI, the certificates and private keys must already be configured and stored on the phone. Use the configuration files or the Aastra Web UI to load certificates and private keys.

Configuring the 802.1x Protocol Using the Configuration Files

You use the following parameters to configure the 802.1x Protocol on your phone using the configuration files.

For EAP-MD5 use:

- **eap type**
- **identity**
- **md5 password**
- **pc port passthrough enabled**

For EAP-TLS use:

- **eap type**
- **identity**
- **802.1x root and intermediate certificates**
  (use 1 root and 0 or 1 intermediate certificates)
- **802.1x local certificate**
  (use 1 local certificate)
- **802.1x private key**
  (1 private key that corresponds to local certificate)
- **802.1x trusted certificates**
  (0 or more trusted certificates (a maximum of 2

)

Configuration Files

For the specific parameters you can set in the configuration files for setting 802.1x support, see Appendix A, the section, "802.1x Support Settings" on page A-121.

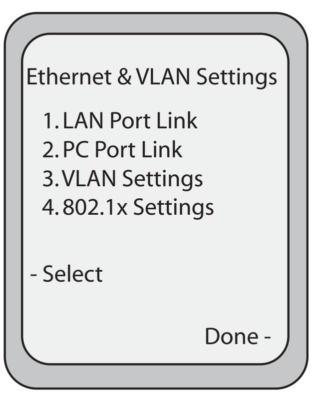
Configuring the 802.1x Protocol Using the IP Phone UI

Use the following procedure to configure the 802.1x Protocol on your phone using the IP Phone UI.

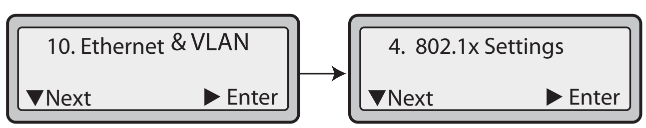
Note: If configuring 802.1x using the IP Phone UI, the certificates and private keys must already be configured and stored on the phone. Use the configuration files or the Aastra Web UI to load certificates and private keys.

Aastra IP Phone UI

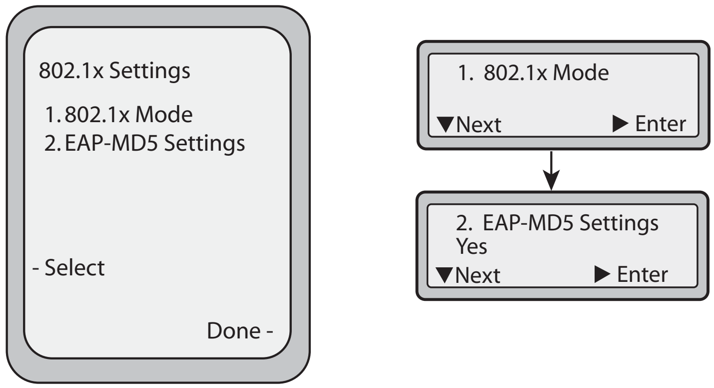
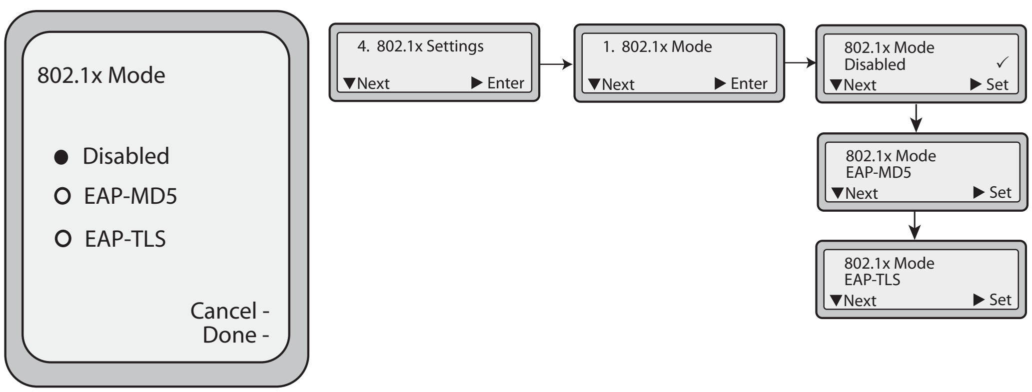
Step	Action
1	Press [Options key] to enter the Options List.
2	Select **Administrator Menu** and enter your Administrator password.
3	**On 11-Line LCDs:** Select **Network Settings->Ethernet & VLAN->802.1x Settings.** **On 3-Line LCDs:** Select **Network Settings->Ethernet & VLAN->802.1x Settings.**



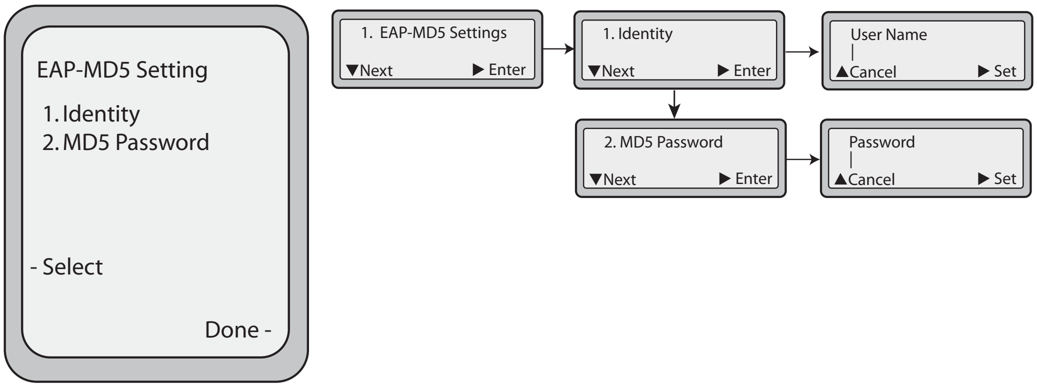
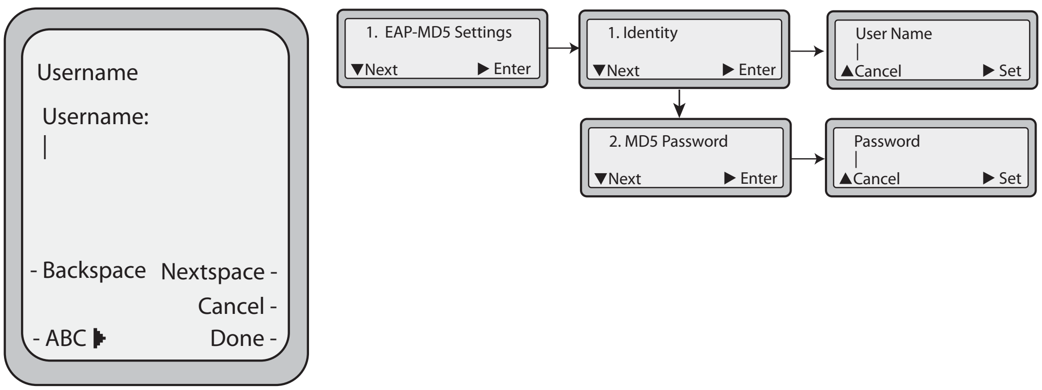
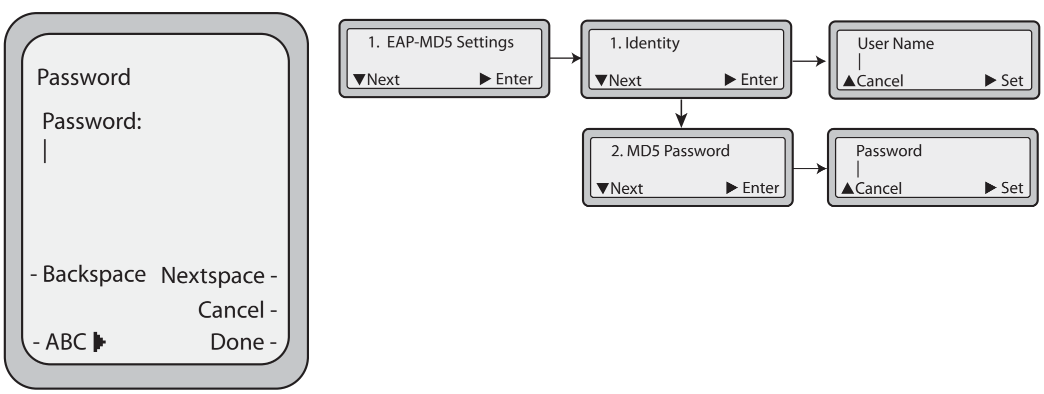
11-Line LCDs



3-Line LCDs

Aastra IP Phone UI	
Step	**Action**
4	Select **802.1x Settings.**
5	Select **802.1x Mode.** 
6	Select **EAP-MD5** to configure the phone to use MD5 authentication; or Select **EAP-TLS** to configure the phone to use TLS authentication.  **Note:** The 802.1x Protocol is disabled by default. If you select EAP-TLS authentication, you must use the configuration files or the Aastra Web UI to configure the certificates and private key for the phone.
7	Press **Done** (on 8 and 11-Line LCD phones) or Press **Set** (on 3-Line LCD phones).
8	To configure MD5, select **EAP-MD5 Settings**.

Aastra IP Phone UI

Step	Action
9	Select **Identity**. 
10	Enter the identity or **username** used for authenticating the phone. 
11	Use the keys as applicable while entering the username; then press **Done** (8 and 11-Line LCD phones). or Press **Set** (3-Line LCD phones).
12	Enter the password used for authenticating the phone. 

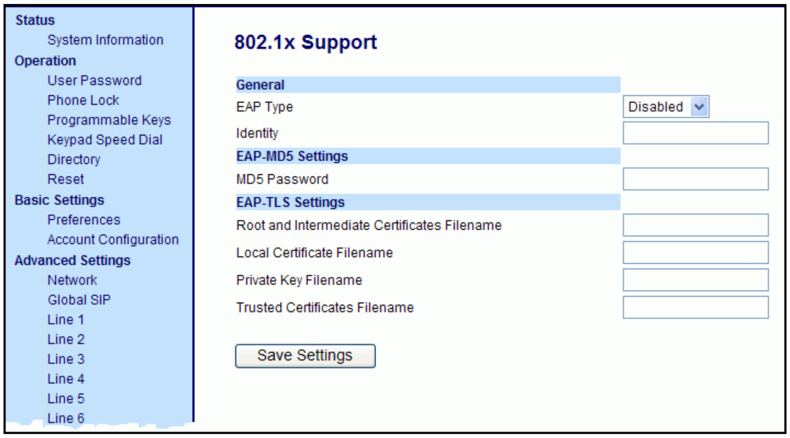
Aastra IP Phone UI	
Step	**Action**
13	Use the keys as applicable while entering the password; then press **Done** (8 and 11-Line LCD phones). or Press **Set** (3-Line LCD phones.
14	Press the key to save your settings and exit from the IP Phone UI.

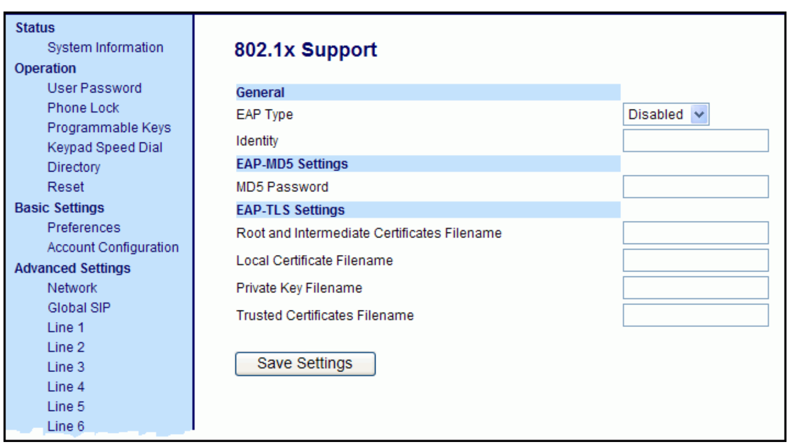
For the 6739i

Aastra IP Phone UI	
Step	**Action**
1	Press to enter the Options List.
2	Press **Advanced** and enter your Administrator password using the pop-up keyboard.
3	Press **Network.**
4	Press the button to scroll to the next page.
5	Press **Ethernet & VLAN.**
6	Press **802.1x Support.**
7	Press **EAP Type** and select a value to set. Valid values are: • **Disable** (Default) • **EAP-MD5** (phone uses MD5 authentication) • **EAP-TLS** (phone uses TLS authentication) • **Note:** The 802.1x Protocol is disabled by default. If you select EAP-TLS authentication, you must use the configuration files or the Aastra Web UI to configure the certificates and private key for the phone.
8	Press **EAP-TLS Settings**.
9	Press **Identity,** and then press the text box. A pop-up keyboard displays allowing you to enter a username used for authenticating the phone.
10	Press **MD5 Password**, and then press the text box to enter the password used for authenticating the phone. **Note:** You must restart the phone for the 802.1x authentication parameters to take affect.
11	Navigate to the **Options** screen and press **Restart** to restart the phone.

Configuring the 802.1x Protocol Using the Aastra Web UI

Use the following procedure to configure the 802.1x Protocol on your phone using the Aastra Web UI.

<table>
<tr><th colspan="2">Aastra Web UI</th></tr>
<tr><td>1</td><td>Click on Advanced Settings->802.1x Support.


</td></tr>
<tr><td colspan="2">To configure EAP-MD5:</td></tr>
<tr><td>2</td><td>In the “EAP Type” field, select EAP-MD5.
Valid values are: Disabled (Default), EAP-MD5, and EAP-TLS.</td></tr>
<tr><td>3</td><td>In the “Identify” field, enter an Identity for the IP phone for which you are configuring 802.1x.
For example, phone1.</td></tr>
<tr><td>4</td><td>In the “MD5 Password” field, enter a password used for the MD5 authentication of the phone.
For example, password1.</td></tr>
<tr><td>5</td><td>Click Save Settings to save your changes.</td></tr>
<tr><td colspan="2">To configure EAP-TLS:</td></tr>
<tr><td>1</td><td>Click on Advanced Settings->802.1x Support.


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

Aastra Web UI	
1	In the **“EAP Type**” field, select **EAP-TLS**. Valid values are: **Disabled** (Default), **EAP-MD5**, and **EAP-TLS**.
2	In the “**Identity**” field, enter an Identity for the IP phone for which you are configuring 802.1x. For example, **phone1**.
3	In the “**Root and Intermediate Certificates Filename**” field, enter the filename that contains the root and intermediate certificates related to the local certificate. For example: **root_Intermed_certifi.pem**.
4	In the “**Local Certificate Filename**” field, enter the filename that contains the local certificate. For example: **localcertificate.pem**.
5	In the “**Private Key Filename**” field, enter the filename that contains the private key. For example: **privatekey.pem**.
6	In the “**Trusted Certificates Filename**” field, enter the filename that contains the trusted certificates. For example: **trusted_certificates.pem**.
7	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-128.

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-246.

GRUU and sip.instance Support

Globally Routable User-Agent URIs (GRUUs) provide a way for anyone on the Internet to route a call to a specific instance of a User-Agent.

The IP phones provide GRUU support using draft-ietf-sip-gruu-15. A sip.instance is added to all non-GRUU contacts. By default, this feature is enabled. You can enable or disable this support using the configuration files.

Limitations of the GRUU Feature

The following are limitations of the GRUU feature on the phones:

- GRUU-Draft-15 is not compatible with versions prior to GRUU-Draft-10.
- Phones do not support temporary or phone-created GRUUs.

Enabling/Disabling GRUU and sip.instance Support

Use the following procedure to enable/disable GRUU and sip.instance support.

Configuration Files
For the specific parameter you can set in the configuration files, see Appendix A, the section, "GRUU and sip.instance Support" on page A-246.

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 Speed Dial 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 Speed Dial 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 Speed Dial 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

Configurable DNS Queries

The domain name system (DNS) is the way that Internet domain names are located and translated into Internet Protocol addresses. A domain name is a meaningful and easy-to-remember identifier for an Internet address.

The lists of domain names and IP addresses are distributed throughout the Internet in a hierarchy of authority within a database of records. There is usually a DNS server within close proximity to your geographic location that maps the domain names in your Internet requests or forwards them to other servers in the Internet.

The IP Phones support three methods of DNS lookups allowing it to adapt to various deployment environments: **eNone**, **eSRV**, and **eNAPTR&SRV**. (See the following table for a description of each method). When the IP Phone accesses the Internet, it sends out a DNS query to the proxy to lookup the IP address and the port, and then waits for a response from the proxy.

You can configure the phone to use any one of these methods by entering the applicable value in the configuration files:.

Configuration File Value	DNS Server Method Used	Description
0	**A only**	The phone sends "**A**" (Host IP Address) lookup for the IP address and uses the default port number of 5060.
1	**SRV & A**	The phone sends "**SRV"** (Service Location Record) lookup to get the port number. Most often, the IP address is included in the response from the DNS server to avoid extra queries. If there is no IP address returned in the response, the phone sends out the "**A**" DNS lookup to find the IP address.
2	**NAPTR & SRV & A**	First, the phone sends "**NAPTR**" (Naming Authority Pointer) lookup to get the "**SRV**" pointer and service type (such as "aastra.com SIP+2DT .... _sip.tcp.aastra.net", which means the service prefers to use TCP and "_sip.tcp.aastra.net" for the SRV query instead of the default "_sip._tcp.aastra.com"). If the NAPTR record is returned empty then the default value is used, so in the same case, the phone will use "_sip._udp.aastra.com" for the next step lookup. Next, the phone does SRV lookup to get the IP address and port number. If there is no IP address in the SRV response then it sends out and "**A**" lookup to get it.

Note: On the phone side, if you configure the phone with a Fully-Qualified Domain Name (FQDN) proxy and specified port, the phone always sends "**A only**" lookups to find the Host IP Address of the proxy.

Configuring the DNS Query Method

You can configure the DNS query method for the phone to use for performing DNS lookups using the following parameter in the configuration files:

- **sip dns query type**

Configuration Files

For the specific parameter you can set in the configuration files, see Appendix A, the section, "DNS Query Setting" on page A-247.

Ignore Out of Sequence Errors

An Administrator can configure the phone via the "**sip accept out of order requests**" parameter to ignore CSeq number errors on all SIP dialogs on the phone. When this parameter is enabled, the phone no longer verifies that the sequence numbers increase for each message within a dialog, and does not report a "CSeq Out of Order" error if they do not increase.

Note: As the default Asterisk configuration does not fully track dialogs through a reboot, it is recommended that this parameter be enabled when using the BLF feature with an Asterisk server. If you do not enable this feature, then rebooting the Asterisk server may cause BLF to stop working. With this parameter enabled, the BLF key starts working again when the phone re-subscribes, which by default, are one hour apart.

An Administrator can enable/disable this feature using the configuration files only.

Enabling/Disabling "Out of Order SIP Requests"

Use the following procedure to enable/disable "out of order SIP requests".

Configuration Files
For the specific parameter you can set in the configuration files, see Appendix A, the section, "Ignore Out of Order SIP Requests" on page A-248.

“Early-Only” Parameter in Replaces Header RFC3891

The phones support the “early-only” parameter in the “Replaces” header according to RFC3891. When the phone receives a Replaces header with the early-only parameter, it replaces the existing dialog if the call is still in the early state. If the call has been answered, then the Replaces request is rejected.

Note: This feature is not supported in outgoing requests.

Switching Between Early Media and Local Ringing

Previous to release 2.6, the phone generated a local ring tone when receiving a 180 response unless early media was being played, regardless of SDP. The local ring tone stopped when the first RTP packet was received in the early session. The local ring tone did not resume when it received subsequent 180 responses, even though early media was not being received.

In Release 2.6 and up, the phone supports switching between early media and local ring tone. Upon receiving a 180 response, the phone now generates a local ring tone unless it is receiving an early media flow. If the phone receives any subsequent 180 responses, it regenerates the local ring tone unless it is receiving early media flow.

Enable Microphone During Early Media

The phones now allow Administrators to enable or disable the microphone during early media by configuring the “**sip early media mute mic**” parameter. Early media indicates the period when a call has not fully established (i.e. the far end has not answered the call). By enabling this parameter, Administrators can mute the microphone during early media to prevent the far end from listening into the call prior to answering it.

Enabling/Disabling “Microphone During Early Media”

Use the following procedure to enable/disable “microphone during early media:

Configuration Files

For the specific parameter you can set in the configuration files, see Appendix A, the section, “Enable Microphone During Early Media” on page A-190.

“Call-Info” Header to 200ok Responses for Shared Call Appearance (SCA) Lines

In earlier versions of the IP phone software releases, a "Call-Info" header was only added to the requests generated by the phone for Shared Call Appearance (SCA) lines. A “Call-Info” header ia now also included in the 200ok response to an INVITE, RE-INVITE, and UPDATE messages for SCA lines. No configuration is required for this feature.

Reason Header Field in SIP Message

The IP Phones support the receiving of the Reason Header Field in a SIP CANCEL message, as described in RFC3326. This allows a call that is answered from somewhere else to still display in the Callers List. Also, the missed calls indicator and counter do not change.

Limitation

If the call is answered somewhere else, the duration of the call does not display in the Callers List.

Configurable “Allow” and “Allow-Event” Optional Headers

On the IP Phones, an Administrator can enable or disable whether or not the optional “Allow” and “Allow-Events” headers are included in the NOTIFY message from the phone.

SIP NOTIFY messages from the phone may contain optional headers called “Allow” and “Allow-events”. If the NOTIFY message contains these headers, the UDP packet returned by the server may be too large and may fragment the packet. To prevent the fragmenting of the UDP packet, the “Allow” and “Allow-events” headers may be removed using the parameter, “**sip notify opt headers**”. If this parameter is set to “**0**” (disabled), the optional headers are not included in the SIP NOTIFY neassage which reduces the size of the packet returned by the server, and prevents fragmentation of the packet.

The value set for this parameter specifies whether or not to include the optional headers in the SIP NOTIFY message from the phone.

An Administrator can enable/disable the optional “Allow” and “Allow-Event” headers using the following parameter in the configuration files:

- **sip notify opt headers**

Enablling/Disabling Optional “Allow” and “Allow-Event” Headers

Use the following procedure to enable/disable “Allow” and “Allow-Event” headers.

Configuration Files
For the specific parameter you can set in the configuration files, see Appendix A, the section, “Optional “Allow” and “Allow-Event” Headers” on page A-248.

Configurable SIP P-Asserted Identity (PAI)

The IP Phones support a private extension to SIP for Asserted Identity within trusted networks (as defined in RFC 3325). 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 Trusted Network, the IP phone reads the contents of the P-Asserted-Identity (PAI) header field and displays it on the phone UI.

The phones provide the ability for the Administrator to enable or disable the display of P-Asserted Identity (PAI) information on the phone using the following parameter in the configuration files:

- **sip pai**

Enabling/Disabling P-Asserted Identity (PAI)

Use the following procedure to enable/disable PAI.

Configuration Files

For the specific parameter you can set in the configuration files, see Appendix A, the section, "P-Asserted Identity (PAI)" on page A-249.

Configurable Route Header in SIP Packet

The IP Phones support the following parameter:

- **sip remove route**

This parameter enables or disables the addition of the Route header in a SIP packet. Enable this parameter for outbound proxies that do not support Route headers.

Note: When enabled this will break all support for SIP routing, so if some other device in the network attempts to add itself to the route it will fail.

Enabling/Disabling the Route Header in the SIP Packet

Use the following procedure to enable/disable the addition of the Route header in the SIP packet.

Configuration Files

For the specific parameter you can set in the configuration files, see Appendix A, the section, "Route Header in SIP Packet" on page A-249.

Configurable Compact SIP Header

The phones provide a feature that allows an Administrator to shorten the length of a SIP packet by using the compact form. This feature is in accordance with Compact SIP Headers defined in RFC 3261.

For example, the following SIP header is the long format:

```
Via: SIP/2.0/UDP
10.50.91.2:5060;branch=z9hG4bK571ebe0c;rport=5060;received=10.50.91.2
From: "Unknown" <sip:Unknown@10.50.91.2>;tag=as19d00fc8
To: <sip:1106@10.50.110.54:5060;transport=udp>;tag=916699998
Call-Id: 73cad5456806f3a7768d17e8617279d7@10.50.91.2
CSeq: 102 OPTIONS
```

The following SIP header is equivalent to the above SIP header, but uses the short (compact) format instead:

```
v: SIP/2.0/UDP
10.50.91.2:5060;branch=z9hG4bK571ebe0c;rport=5060;received=10.50.91.2
f: "Unknown" <sip:Unknown@10.50.91.2>;tag=as19d00fc8
t: <sip:1106@10.50.110.54:5060;transport=udp>;tag=916699998
i: 73cad5456806f3a7768d17e8617279d7@10.50.91.2
CSeq: 102 OPTIONS
```

By default, the IP Phones use the long format. However, an Administrator can provision the short (compact) format using the configuration files. The Aastra Web UI does not support this configuration feature.

Enabling/Disabling the Compact SIP Headers Feature

Use the following procedure to enable/disable the Compact SIP Header in the SIP packet.

Configuration Files
For the specific parameter you can set in the configuration files, see Appendix A, the section, "Compact SIP Header" on page A-250.

Reject INV or BYE when Unsupported Value in REQUIRE Header

The IP Phones support the following parameter:

- **sip enforce require hdr**

This parameter allows you to enable (1) or disable (0) the rejection of an INV or BYE with a "420 Bad Extension" if the INV or BYE contains an unsupported value in the REQUIRE header.

Enabling/Disabling a Rejection of the INV or BYE

Use the following procedure to enable/disable the athe rejection of the INV or BYE if the INV or BYE contains an unsupported value in the REQUIRE header..

Configuration Files

For the specific parameter you can set in the configuration files, see Appendix A, the section, "Rejection of INV or BYE" on page A-250.

XML URI for Key Press Simulation

The phones provide a feature that allow an XML Developer or Administrator to define XML Key URIs that can send key press events to the phone, just as if the physical hard key, softkey, or programmable key were pressed on the phone.

When the Key URI event is sent from the server to the phone, the phone initiates the event as if the key was physically pressed. If the key is not present on the phone (hard key) or not available (softkey or programmable key), when the phone receives the URI, the event is discarded. If you are in the process of changing the softkey or programmable key setting, or the key is disabled while the event is being processed, the request is discarded. The phone maps key events to it's physical keys and not to it's mapped logical keys.

The following table identifies the XML URIs for pressing buttons on the phone..

XML Key URI	Description
Line Keys	
Key:Line1 to Key:Line4	Line 1 to 4 Keys **Note:** The phone ignores URI line keys 5 to 9 since it does not have Line 5 to 9 physical keys.
Keypad Keys	
Key:KeyPad0 to Key:KeyPad9	Numeric Keypad Keys 0-9
Key:KeyPadStar	* - Star Key
Key:KeyPadPound	# Hash Key
Softkeys	
Key:SoftKey1 to Key:SoftKey<n>	Softkey 1 to <n> (valid softkeys depend on the number of physical softkeys on the phone)
Key:TopSoftKey1 to Key:TopSoftKey<n> top	Top softkeys 1 to <n> ((valid top softkeys depend on the number of physical top softkeys on the phone)
Programmable Keys	
Key:PrgKey1 to Key:PrgKey<n>	Programmable keys 1 to <n> (valid programmable keys depend on the number of physical programmable keys on the phone)
Expansion Module Keys	
Key:ExpMod1SoftKey1 to Key:ExpMod1SoftKey60	Expansion module 1 softkeys 1 to 60 **Note:** The phone ignores URI expansion module key events if the keys are not physically present on the expansion module.
Key:ExpMod2SoftKey1 to Key:ExpMod2SoftKey60	Expansion module 2 soft keys 1 to 60
Key:ExpMod3SoftKey1 to Key:ExpMod3SoftKey60	Expansion module 3 soft keys 1 to 60
Volume Key	
Key:VolDwn	Volume Decrease Key
Key:VolUp	Volume Increase Key

XML Key URI	Description
Feature Keys	
Key:Xfer	Transfer Key
Key:Conf	Conference Key
Key:Services	Services Key
Key:Intercom	Intercom Key
Key:Headset	Headset Key **Note:** For Headset URI key, the behavior will be as if the "speaker/headset" key is pressed; and does not switch to headset for headset key event or to speaker for speaker key event.
Key:Speaker	Speaker Key **Note:** For Speaker URI key, the behavior will be as if the "speaker/headset" key is pressed; and does not switch to headset for headset key event or to speaker for speaker key event.
Key:Mute	Mute Key
Key:Hold	Hold Key
Key:Redial	Redial Key
Key:Callers	Callers Key
Key:Directory	Directory Key
Key:Options	Options Key
Key:Save	Save Key
Key:Delete	Delete Key
Key:Swap	Swap Key
Key:Goodbye	GoodBye Key
Navigation Keys	
Key:NavUp	Navigation Up Key
Key:NavDwn	Navigation Down Key
Key:NavLeft	Navigation Left Key
Key:NavRight	Navigation Right Key
Function Keys (only if physically configured on the phone or expansion module)	
KeyPark	Park Softkey
KeyPickup	Pickup Softkey

Notes:

1. If the URI key is a valid key, the phone executes the key regardless of the current state on the phone.

2. Park and Pickup XML URI softkeys are available **ONLY** if these features are physically configured on the phone or expansion module.

Examples

There are two ways to format the XML key URI:

For XML Post Messages

<ExecuteItem URI="<XML Key URI>" />

Example:

```
<ExecuteItem URI="Key: Line1" />
```

For XML Key Scripts

<URI><XML Key URI></URI>

Example:

```
<URI>Key: Line1</URI>
```

```
<SoftKey index="1">
<Label>Keypad1</Label>
<URI>Key: Line1</URI>
</SoftKey>
```

Domain Name Server (DNS) Pre-caching Support

The IP phones now support the use of a local DNS host file to resolve DNS queries, and supports pre-provisioning of DNS SRV records. This feature allows administrators to configure the phone to download a text file which contains persistent DNS “A record” hostname to IP address mappings. In addition, support for persistent DNS “SRV records” has been added to permit SRV based high availability of services.

There are two methods used to configure DNS pre-caching on the IP phone:

- Configure a unix style “host” file used instead of a DNS “A query” to resolve hostnames to IP addresses. The host file is downloaded and cached on the IP phone.
- Configure DNS “SRV queries” for geographic redundancy and failover. The configured SRV entries are used to pre-load the DNS cache on the IP phone.

Both these methods are configurable using the configuration files only, and are primarily intended for use when a third party hosting provider delivers SIP services but does not have local access or control of the LAN side DNS infrastructure.

➡ **Note:** Time-to-Live (TTL) used in this feature is hard-coded for each server and not configurable.

Configuring DNS “Host File” Pre-caching from the Configuration Server

The DNS host file must reside on the same server as the configuration files (*aastra.cfg/.tuz*, etc.) and the filename to download is specified within the configuration.

Use the following parameter to configure the phone to use the host file for host IP address lookups.

Configuration Files
For the specific parameter you can set in the configuration files, see Appendix A, the section, “DNS Host File” on page A-251.

The following procedure is an example of how to use the “sip dns host file” parameter to configure DNS lookup pre-caching from the configuration server.

Configuring “DNS Host” Pre-caching	
1	Using a text-based editing application, create a blank text file
2	Enter the IP addresses of the DNS servers in your local network. For example: **1.2.3.4 server1** **5.6.7.8 server2** **9.0.1.2 server3** **Note:** Ensure each line uses a Carriage Return (CR) or Carriage Return + Line Feed (CRLF) to terminate the line.
3	Save the file as “*<filename>.txt*”. For example, “***hostfile.txt***”.
4	Using a text-based editing application, open the *aastra.cfg* file for the phone(s) for which you want to apply the DNS hostfile.
5	Enter the following parameter in the *aastra.cfg* file followed by the host file name as the value: **sip dns host file: <filename>.txt** For example: **sip dns host file: hostfile.txt** **Note:** If using a text file on a PC to enter this value, you must enter a carriage return (CR) after entering the host file name.
6	Save the file. Make sure the *aastra.cfg* and the *hostfile.txt* files are on the configuration server in your network before downloading to the phone(s).
7	Restart the phone(s) in your network. The phone(s) downloads the specified host text file and stores it locally on the phone’s flash memory. Upon each subsequent boot of the phones, if the host text file is available on the configuration server, it is downloaded to replace the locally cached copy; otherwise, the previously cached copy is retained and used unchanged. The configuration of the phone(s) can now use server1, server2, or server3 for SIP or other services instead of using the IP addresses. The phones will continue resolving the host names even if DNS on the network has conflicting or missing entries for server1, server2, or server3, or if the local LAN DNS server fails to respond.

Configuring DNS “Service (SRV) Records” pre-caching

In addition to using a host file to resolve host names to IP addresses, an Administrator can also configure DNS “SRV records” (Service Records) for geographic redundancy and failover between application servers in the network.

The SIP registration and SIP proxy features on the phones previously allowed the use of server queries only to live DNS servers. Using the host file and specific DNS SVR parameters extends this mechanism to allow pre-configuration of server values in the *aastra.cfg* file. The following new parameters are used for this feature:

- **sip dns srvX name**
- **sip dns srvX priority**
- **sip dns srvX weight (supported in a future release)**
- **sip dns srvX port**
- **sip dns srvX target**
-

Note: The “X” indicates a DNS SRV with a value from 1 to 4.

You can configure up to 4 DNS SRV records, with each server having a **priority** which tells the phone which server to use, and a host name or **target**. The IP phone will use the DNS SRV record with the lowest-numbered priority value first, and will only failover to other records if the connection with this record's host fails. Thus a service may have a designated failover server, which is only used if the primary server fails.

If a service has multiple SRV records with the same priority value, the IP phone(s) use the weight field to determine which host to use. The weight value is a ratio compared to the weight of other records with the same name and priority value.

Note: The “sip dns srvX weight” parameter must be configured but the phones will support this feature in a future release.

In the following example, both the priority and weight fields are used to provide a combination of load balancing and backup service.

Example

sip dns srv1 name: _sip._udp.example.com
sip dns srv1 priority: 10
sip dns srv1 weight: 60
sip dns srv1 port: 5060
sip dns srv1 target: bigbox.example.com

sip dns srv2 name: _sip._udp.example.com
sip dns srv2 priority: 10
sip dns srv2 weight: 20
sip dns srv2 port: 5060
sip dns srv2 target: smallbox1.example.com

sip dns srv3 name: _sip._udp.example.com
sip dns srv3 priority: 10
sip dns srv3 weight: 20
sip dns srv3 port: 5060
sip dns srv3 target: smallbox2.example.com

sip dns srv4 name: _sip._udp.example.com
sip dns srv4 priority: 20
sip dns srv4 weight: 10
sip dns srv4 port: 5060
sip dns srv4 target: backupbox.example.com

The first three records (SRV 1, 2, and 3) share a priority of 10, so the weight field's value is used by the phones to load balance across the three target host names.
Bigbox will get 60% of the load, and smallbox1 and smallbox2 will each get 20% load.

If all three servers with priority 10 are unavailable, the next highest priority record is selected, in this case backupbox. This could be a server in another physical location.

The server entries in the *aastra.cfg* file can use DNS hostnames or can use IP addresses. If hostnames are used, any pre-cached DNS A records via the host file mechanism are used before resorting to live DNS query if there is no local match.

For example, the following hostfile.text uses IP addresses that are used in the DNS server queries:

```
hostfile.txt
192.168.2.3 bigbox.example.com
192.168.3.4 smallbox1.example.com
192.168.8.1 smallbox2.example.com
47.28.05.69 backupbox.example.com
```

Use the following procedure in the configuration files to configure DNS server query support for the phones.

Configuration Files

For the specific parameter you can set in the configuration files, see Appendix A, the section, "DNS Server Query" on page A-251.

Use the following procedure to configure DNS SRV record pre-caching..

	Configuring DNS SRV Record Pre-caching
1	Using a text-based editing application, open the *aastra.cfg* file.
2	Enter the parameter, "**sip dns srvX name**", where "X" is a value from 1 to 4. Enter a value for the DNS SRV service URI. For example: **sip dns srv1 name: _sip._udp.example.com**
3	Enter the parameter, "**sip dnx srvX priority**, where "X" is a value from 1 to 4. Enter a value for the DNS server priority. Valid values are 0 to 65535. Default is 0. For example: **sip dns srv1 priority: 10** After this parameter is downloaded from the configuration server to the phone, the phone uses the DNS server with the lowest numbered priority first to perform DNS lookups.
4	Enter the parameter, "**sip dnx srvX weight**, where "X" is a value from 1 to 4. Enter a value for the DNS server weight. Valid values are 0 to 65535. Default is 0. For example: **sip dns srv1 weight: 60** **Note:** The "sip dns srv1 weight" parameter must be configured but will be supported in a future release.
5	Enter the parameter, "**sip dnx srvX port**, where "X" is a value from 1 to 4. Enter a value for the port number on the target host. Valid values are 0 to 65535. Default is 0. For example: **sip dns srv1 port: 5060**
6	Enter the parameter, "**sip dnx srvX target**, where "X" is a value from 1 to 4. Enter a value for the DNS server target. Valid values are the host name or a fully qualified domain name. For example: **sip dns srv1 target: bigbox.example.com**
7	Save and close the file.
8	Place the *aastra.cfg* file on the configuration server and download to the phones.

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 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*" to encrypt the *<MAC>.tuz* file.

➡ **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.

To make changes to the configuration files, the System Administrator must save the original files.

➡ **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.

The security feature described above prevents unauthorized parties from **reading** or **writing** the contents of the *<MAC>.tuz* file. It also provides the following:

- Prevents users from using the *<MAC>.tuz* file that does not match the user's phone MAC address.
- Renders the *<MAC>.tuz* file invalid if the user renames the file.
- Works with IP phone releases prior to Release 2.2.
- Provides compatibility between the previous encryption routine and the new decryption routine.

Procedure to Encrypt 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 of the configuration files used for the family of Aastra IP phones, using
56bit triple-DES and site-specific keys.

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Usage:
anacrypt {infile.cfg|-d <dir>} [-p password] [-m] [-i] [-v] [-h]
```

Anacrypt Switch	Description
{infile.cfg | -d <dir>}	Specifies that all .cfg files in <dir> should be encrypted
[-p password]	Specifies password used to generate keys
-m	Generate MAC.tuz files that are phone specific. This switch generates files that are only usable for phones with firmware version 2.2.0 and above.
-i	Generate security.tuz file
-v	Specifies the version of encryption that the anacrypt tool uses. Use version 1 encryption (i.e., -v1) to generate firmware that is readable by all model phones. Without the -v1 switch, the anacrypt tool generates files that are only readable by phones with firmware 2.2.0 and above.
-h	Display program help text

Note: Incorrect password produces garbage. For site-specific keyfile security.cfg the plaintext must match password.

Examples

The following examples illustrate the use of the anacrypt.exe file.

Example 1

Generating a security.tuz file with password 1234abcd

For firmware version 2.2.0 and up:

C:\>**anacrypt -i -p 1234abcd**

For any firmware version:

C:\>**anacrypt -i -p 1234abcd -v1**

Example 2

Encrypting a single aastra.cfg file with password 1234abcd

C:\>**anacrypt aastra.cfg -p 1234abcd**

Example 3

Encrypting a <mac>.cfg file with password 1234abcd

C:\>**anacrypt 00085d000000.cfg -p 1234abcd**

Example 4

Encrypting a <mac>.cfg file with password 1234abcd using the new MAC encryption (-m is only supported for firmware version 2.2.0 and up)

C:\>**anacrypt 00085d000000.cfg -m -p 1234abcd**

Example 5

Encrypting all cfg files in C:\data with password 1234abcd using the new MAC encryption and generating a security.tuz file at the same time.(2.2.0 and up)

C:\>**anacrypt -d C:\data -p 1234abcd -m -i**

Example 6

Encrypting all cfg files in C:\data with password 1234abcd using the and generating a security.tuz file at the same time for all firmware versions.(any version)

C:\>**anacrypt -d C:\data -p 1234abcd -i -v1**

Vendor Configuration File Encryption

Some vendors can have specific methods to encrypt files on their configuration servers. For each phone, the configuration server can generate a random hex string (encryption key) that is used to encrypt the phone's MAC-specific configuration file.

The encryption key is placed in a plain text MAC-specific configuration file that the server downloads to the phone. After the phone receives the file, it updates the encryption key.

This method of encryption does not affect the implementation of the Aastra method of file encryption.

Note: The *aastra.cfg* file is not encrypted with this feature.

You can set the phone-specific encryption key using the configuration files only.

For more information about configuration file encryption, contact Aastra Technical Support.

Configuring Vendor Configuration File Encryption

Use the following procedure to configure vendor configuration file encryption on the IP Phones.

Configuration Files
For specific parameters you can set in the configuration files for automatic update, see Appendix A, the section, "Configuration Encryption Setting" on page A-250.

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:

Upgrading the Firmware

The IP phones support the protocols, TFTP, FTP, HTTP or HTTPS to download configuration files and upgrade firmware to the phones from a configuration server.

The configuration server should be ready and able to accept connections. For information on configuration server requirements, see Chapter 1, the section, "Configuration Server Requirement" on page 1-35.

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

- Using the "**Firmware Update**" page in the Aastra Web UI at the location ***Advanced Settings->Firmware Update***.
- Using the IP Phone UI or the Aastra Web UI to **restart** the phone. The phone automatically looks for firmware updates and configuration files during the boot process.
- Setting an **Auto-Resync** feature to automatically update the firmware, configuration files, or both at a specific time in a 24-hour period). (Feature can be enabled using the configuration files or the Aastra Web UI).

Using the "Firmware Update" Page in the Aastra Web UI

You can use the Aastra Web UI to manually force a firmware update from the configuration server to a phone in your network by selecting ***Advanced Settings->Firmware Update***. You can configure the phone to perform the update using any of the protocols that the phone supports: TFTP, FTP, HTTP, or HTTPS.

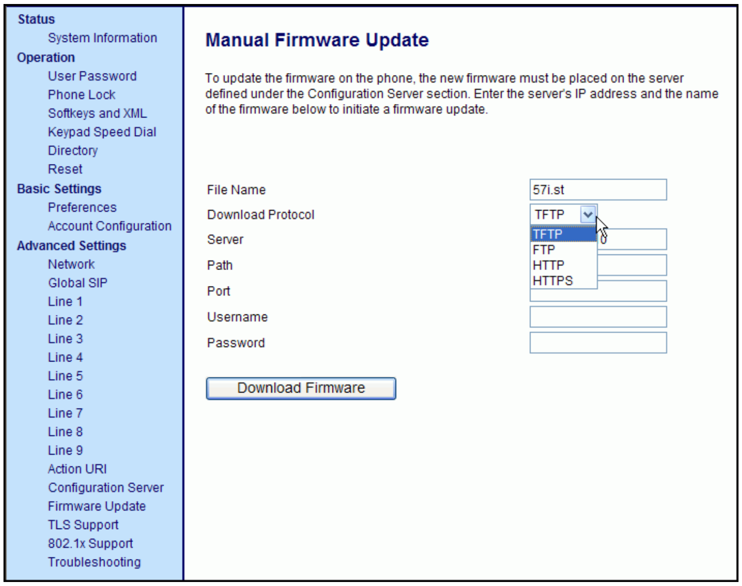
Warning: Do not reset or turn off the phone until the download is complete.

Note: This procedure downloads an updated <*phone model.st*> file as well as any other firmware files that were updated, to your phone.

Use the following procedure to manually update the firmware on your phone from the specified configuration server.

Aastra Web UI

Step	Action
1	Click on **Advanced Settings->Firmware Update.**



Step	Action
2	In the "**File Name**" field, enter the firmware file name (*<phone model>.st*) that you want to download to your IP phone. For example, *53i.st (for a 6753i phone)*. **Note:** This file name must match the actual name of the firmware file residing on your configuration server. **IMPORTANT**: The phones use the following naming convention for the ***.st*** files. See the following table.

IP Phone Model	Associated Firmware
9143i	9143i.st
9480i	9480i.st
9480i CT	9480iCT.st
6730i	6730i.st
6731i	6731i.st
6739i	6739i.st
6753i	53i.st
6755i	55i.st
6757i	57i.st
6757i CT	57iCT.st

Aastra Web UI

Step	Action
3	In the "**Download Protocol**" field, select the protocol from the list to use for downloading the new firmware. Valid values are: • **TFTP** • **FTP** • **HTTP** • **HTTPS**
4	In the "**Server**" field, enter the IP address in dotted decimal format, of the TFTP configuration server, or the domain name of the FTP, HTTP, or HTTPS configuration servers (dependant on the protocol you selected in step 3.) For example: 432.221.45.6.
5	In the "**Path**" field, enter the path location on the protocol server for where the new firmware resides. For example, C:\aastra\configserver\firmwareupgrade.
6	In the "**Port**" field, enter the port number of the protocol server. For example, 80 (for HTTP) or 443 (for HTTPS). **Note:** This field is not applicable to the TFTP and FTP protocols.
7	(FTP only) In the "**Username** " field, enter the username that is used for authentication when the FTP server is accessed.
8	(FTP only) In the "**Password** " field, enter the password that is used for authentication when the FTP server is accessed.
9	Click Download Firmware . This starts the upgrade process. If the upgrade is successful the following message displays on the screen: "Firmware Upgrade Successful".

Using the Restart Feature

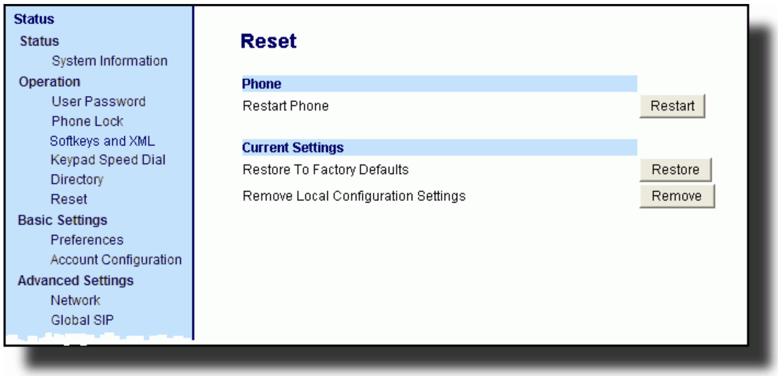
Restarting the phone forces the phone to check for both firmware and configuration files stored on the configuration server.

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 3-Line LCD Displays::*** Press **#** to confirm. **Note:** To cancel the Restart, press the 3 key. ***For 8 and 11-Line LCD Displays::*** Press **Restart**. **Note:** To cancel the Restart, press **Cancel**.
For the 6739i:	
1	Press on the phone to enter the Options List.
2	Press **Restart**. The following prompt displays: *"Restart the phone?"*
3	Press **<Yes>** to restart the phone or Press **<No>** to go back to the Optons Screen.

Restarting the Phone Using the Aastra Web UI

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

Using the Auto-Resync Feature

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, HTTP, and HTTPS 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.

An Administrator can enable Auto-Resync using the configuration files or the Aastra Web UI. In the configuration files you set the following parameters:

- **auto resync mode -** Determines whether the configuration server automatically updates the phone's configuration files only, the firmware only, both the firmware and configuration files, or disables automatic updates. This parameter works with TFTP, FTP, HTTP, and HTTPS servers.
- **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, HTTP and HTTPS servers.
- **auto resync max delay -** Specifies the maximum time, in minutes, the phone waits past the scheduled time before starting a checksync.
- **auto resync days -** Specifies the amount of days that the phone waits between checksync operations.

In the Aastra Web UI, you can set the following parameters at the path ***Advanced Settings->Configuration Server->Auto-Resync***:

- **Mode**
- **Time (24 hours)**
- **Maximum Delay**
- **Days**

Setting the "**auto resync max delay**" (Maximum Delay) and "auto resync days" (Days) parameters can greatly reduce the load placed on the configuration server when downloading configurations.

Enabling Auto-Resync Using the Configuration Files

Use the following procedure to configure automatic updates of the IP phone firmware, configuration files, or both.

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-18.

.

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.

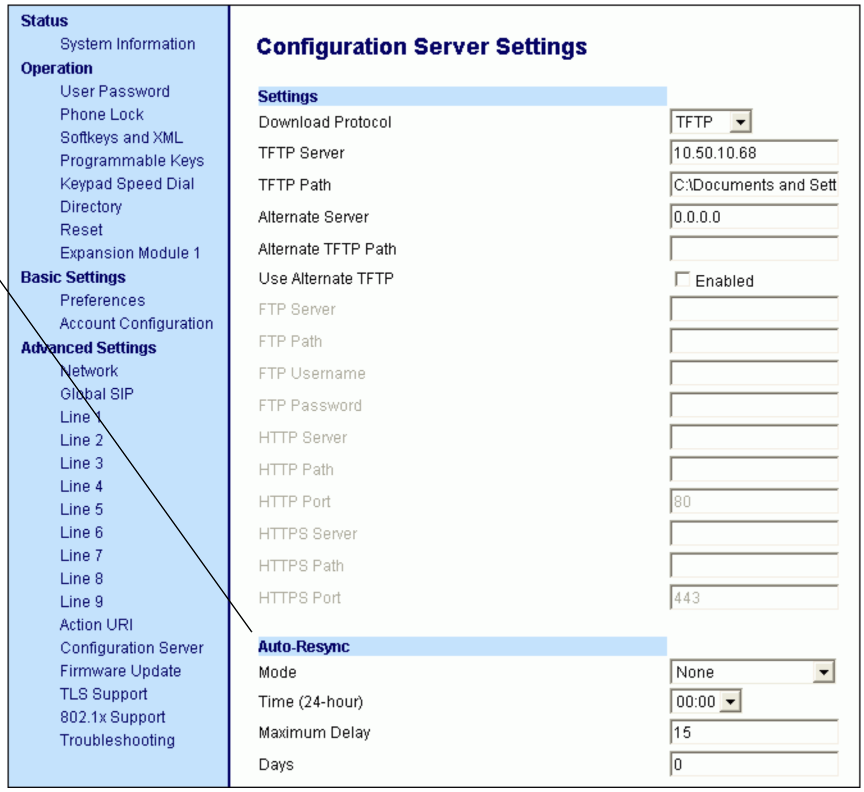
5. The automatic update feature works with both encrypted and plain text configuration files.

Enabling Auto-Resync Using the Aastra Web UI

Use the following procedure to enable auto-resync for the phones in your network.

Aastra Web UI

<table>
<tr><th>Step</th><th>Action</th></tr>
<tr><td>1</td><td>Click on Advanced Settings->Configuration Server->Auto-Resync.

Auto-Sync Parameters

</td></tr>
<tr><td>2</td><td>In the “Mode” field, select the auto-resync mode you want to use to automatically update the phone.

Valid values are:

None Disable auto-resync
Configuration Files Updates the configuration files on the IP phone automatically at the specified time if the files on the server have changed.
Firmware Updates the firmware on the IP phone automatically at the specified time if the files on the server have changed.
Both 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.</td></tr>
</table>

Aastra Web UI

Step	Action
3	In the “**Time (24-hour)**” field, select the time that you want the update to take place. Valid values are 00:00 to 23:30 (in 30 minute increments). **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	In the “**Maximum Delay**” field, specify the maximum time, in minutes, the phone waits past the scheduled time before starting a checksync. The range is 0 to 1439. Default is 15.
5	In the “**Days**” field, specify the amount of days that the phone waits between checksync operations. The range is 0 to 364. Default is 0. **Note:** A value of 0 causes the phone to checksync every time the clock reads the proper time. A value of 1 forces the phone to wait 24 hours prior to doing the first checksync.
6	Click Save Settings to save your settings. These changes are not dynamic. You must restart your IP phone for the changes to take affect.
7	Click on **Operation->Reset**.
8	In the "**Restart Phone**" field click Restart to restart the IP phone and apply the update. The update performs automatically at the time you designated.

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 (including “Free Memory”and “Maximum Memory Block Size”)

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.

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.

Reference

For more information about the log setting configuration parameters, see Appendix A, the section, “Troubleshooting Parameters” on page A-255.

For information about configuring the log settings using the Aastra Web UI, see “Performing Troubleshooting Tasks” on page 9-6.

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.

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
PROF (Profiler module)	log module prof
XML (Extension Markup Lanaguage)	log module xml
STUN (Simple Traversal of User Datagram Protocol (UDP) through Network Address Translation (NAT)	log module stun

Setting Values for the Module/Debug Levels

There are 6 debug levels for the modules. Each debug level has a value you can use to turn individual levels ON and OFF. The following table identifies these debug levels and their values. The value of “**1**” (fatal errors) is the default setting for all modules.

Debug Level	Value
Fatal Errors	1 (default)
Errors	2
Warnings	4
Init	8
Functions	16
Info	32
All debug levels OFF	0
All Debug Levels ON	65535

Example 1

To turn two or more debug levels on at the same time, you add the value associated with each level. For example,

Fatal Errors + Errors + Warnings = 1 + 2 + 4 = **7**

```
log module linemgr: 7
log module user interface: 7
log module sip: 7
```

In the above example, fatal errors, general errors, and warnings are logged for the line manager, user interface, and SIP call control modules.

Example 2

Functions and Info = 16 + 32 = **48**

```
log module dis: 48
log module net: 48
log module snd: 48
```

In the above example, functions and general information are logged for the display drivers, network, and sound modules.

Example 3

```
log module rtpt: 0
log module ind: 65535
```

In the above example, all debug levels are OFF for the Real Time Transport module. All debug levels are ON for the indicator module.

You can set the Module/Debug Levels using the configuration files or the Aastra Web UI.

Reference

For more information about the debug level configuration parameters, see Appendix A, the section, “Troubleshooting Parameters” on page A-255.

For information about configuring the log settings using the Aastra Web UI, see “Performing Troubleshooting Tasks” on page 9-6.

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.

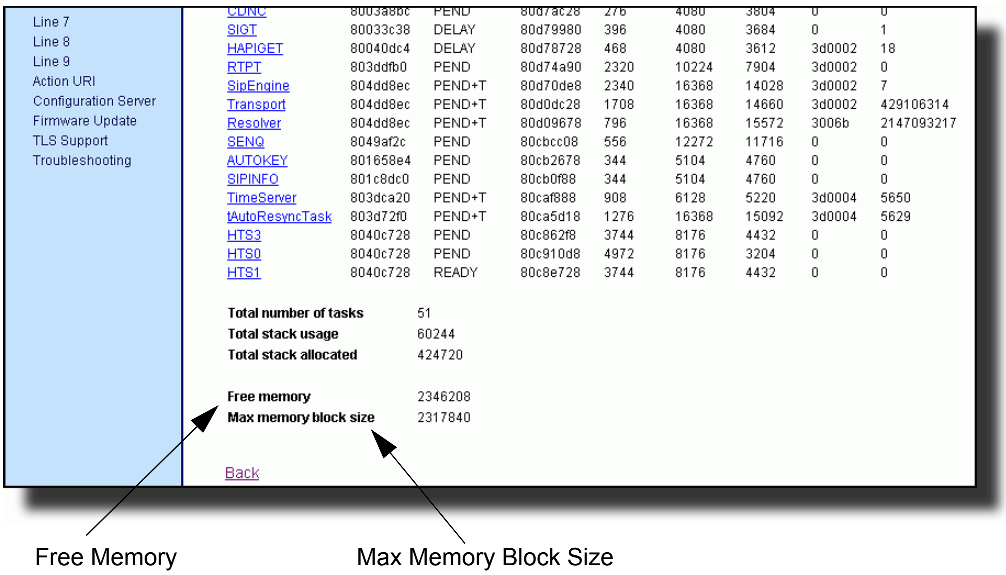
In the “**Support Information**” section, you can:

- **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.

Using the Aastra Web UI and selecting “**Show Task and Stack Status**” displays the tasks and stack status on the IP phone. This screen also displays the **Free Memory** and the **Max Block Free Memory** currently on the phone as shown in the following illustration. This information is for troubleshooting purposes only.

.



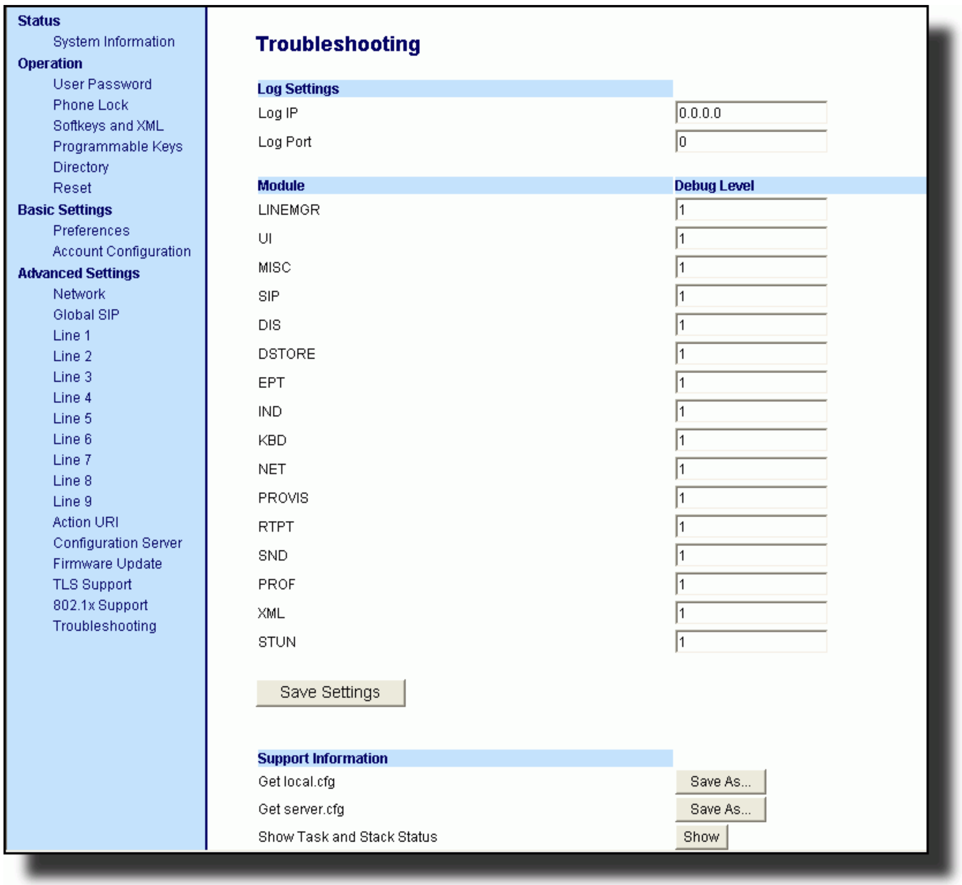
Performing Troubleshooting Tasks

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, "Log Settings" on page A-255.

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.
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.

Aastra Web UI

<table>
<tr><th>Step</th><th>Action</th></tr>
<tr><td>8</td><td>Enter a debug level value in the "Debug Level" field for a module. Valid values are:
<table>
<tr><th>Debug Level</th><th>Value</th></tr>
<tr><td>Fatal Errors</td><td>1 (default)</td></tr>
<tr><td>Errors</td><td>2</td></tr>
<tr><td>Warnings</td><td>4</td></tr>
<tr><td>Init</td><td>8</td></tr>
<tr><td>Functions</td><td>16</td></tr>
<tr><td>Info</td><td>32</td></tr>
<tr><td>All debug levels OFF</td><td>0</td></tr>
<tr><td>All Debug Levels ON</td><td>65535</td></tr>
</table>
The value of "0" turns all debug levels OFF for a module. The value of "65535" turns all debug levels ON for a module.

To turn two or more debug levels on at the same time, you add the value associated with each level. For example, Fatal Errors + Errors + Warnings = 1 + 2 + 4 = 7

<code>log module linemgr: 7</code>
<code>log module user interface: 7</code>
<code>log module sip: 7</code>

In the above example, fatal errors, general errors, and warnings are logged for the line manager, user interface, and SIP call control modules.</td></tr>
<tr><td>9</td><td>Click [Save Settings] to save your settings.</td></tr>
<tr><td>10</td><td>Click on Operation->Reset.</td></tr>
<tr><td>11</td><td>In the "Restart Phone" field click [Restart] to restart the IP phone.</td></tr>
<tr><td colspan="2">To perform support tasks:</td></tr>
<tr><td>12</td><td>To store the local configuration file to the specified location, click on [Save As...] in the "Get local.cfg" field.</td></tr>
<tr><td>13</td><td>To store the server configuration file to the specified location, click on [Save As...] in the "Get server.cfg" field.</td></tr>
<tr><td>14</td><td>To display task and stack status information, as well as Free Memory and Maximum Block Free Memory on the phone, 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.</td></tr>
</table>

Reference

For information that describes solutions to most common problems using the IP phones, see the next section, "Troubleshooting Solutions" on page 9-19.

WatchDog Task Feature

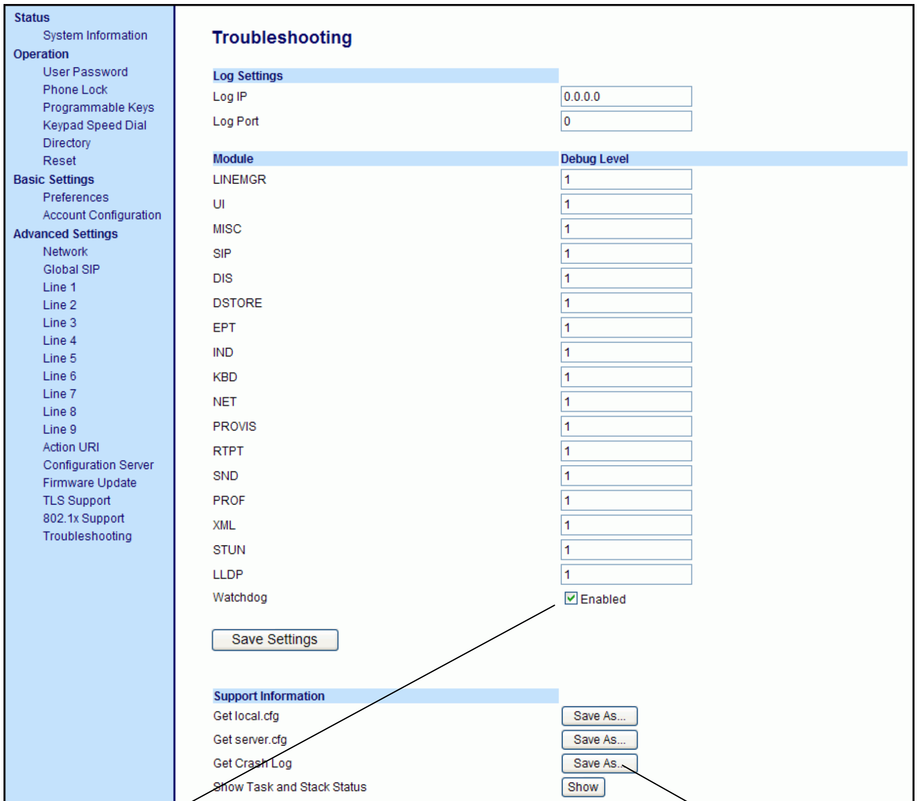
The IP Phones include a troubleshooting feature called the "**WatchDog**" task that monitors the status of the phones and provides the ability to get stack traces from the last time the phone failed. When the phone detects a failure (i.e., a crash), it automatically reboots. You can view a WatchDog crash file using the Aastra Web UI at the path, *Advanced Settings->Troubleshooting*. You can enable/disable the WatchDog task using the configuration files or the Aastra Web UI.

Enabling/Disabling WatchDog

Use the following procedure to enable/disable the WatchDog.

Configuration Files
For specific parameters you can set in the configuration files, see Appendix A, the section, "WatchDog Settings" on page A-257.

Use the following procedure to enable/disable the WatchDog task for the IP Phones using the Aastra Web UI. You can also view the “Crash Log” generated by the WatchDog task using the Aastra Web UI.

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


Enable/Disable WatchDog View Crash Log</td></tr>
<tr><td colspan="2">Enable/Disable WatchDog Task</td></tr>
<tr><td>2</td><td>The “WatchDog” field is enabled by default. To disable the WatchDog task, click in the “Enabled” box to clear the check mark.</td></tr>
<tr><td>3</td><td>Click Save Settings to save your changes.</td></tr>
<tr><td colspan="2">View the Crash Log</td></tr>
<tr><td>4</td><td>To view a crash log, in the “Get a Crash Log” field, click the SAVE AS button. You can open the file immediately, or you can save the Crash Log to your PC.</td></tr>
</table>

Error Messages Display

An Administrator can view generated error messages that may have occurred during startup or reboot of the IP Phones. The IP Phone UI has a selection on the Phone Status page called, "**Error Messages**" at the location, ***Options->Phone Status->Error Messages***. The Aastra Web UI also allows you to view these error messages at the location ***Advanced Settings->Troubleshooting->Error Messages***. These options allow you to view error messages generated by modules during startup only (not after registration has completed). You can use this information for troubleshooting purposes or for reporting the errors to the Administrator.

The IP Phone stores and displays up to 10 error messages (any extra error messages beyond 10 are discarded). The time and date of each error message also displays. After a reboot, the previous error messages are discarded and, if applicable, new error messages display. If there are no error messages during startup or after a reboot, the message, "*No Error Messages*" displays on the screen. Error messages display in the language currently set on the phone.

The following table identifies the possible error messages that may display.

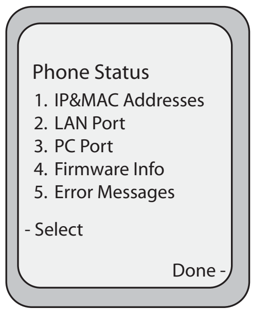
Possible Error Message	Description
Bad Certificate	A Transport Layer Security (TLS) certificate is not valid. The invalid certificate can be any of the following: • root and intermediate certificate • local certificate • private key filename • trusted certificate
802.1x Startup Failed	The Extensible Authentication Protocol TLS (EAP-TLS) certificates and/or the EAP-MD5 information has failed on the phone.
LLDP Startup Failed	Link Layer Discovery Protocol (LLDP) failed during startup of the phone.
HTTP Connection Manager Init Failed	The Hypertext Transfer Protocol (HTTP) connection manager initialization failed while updating the configuration on the phone.
Failed to config Line Manager	The configuration of the Line Manager module on the phone has failed.

Viewing the Error Messages Using the IP Phone UI

Use the following procedure to view the error messages, if any, that generated during startup.

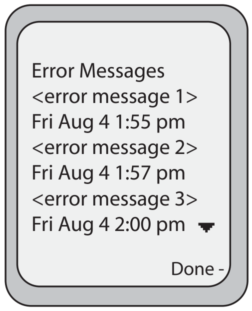
IP Phone UI	
1	Press the **Options** key on the phone to enter the Options List.
2	Select **Phone Status.**
3	Select **Error Messages**.

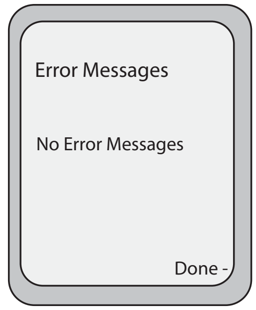
8 and 11-Line LCD Phones



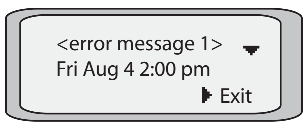
3-Line LCD Phones

<table>
<tr><th colspan="2">IP Phone UI</th></tr>
<tr><td>4</td><td>If error messages display, use the UP and DOWN navigation keys to view the messages. If no error message exist, the message, “No Error Messages” displays on the screen.

8 and 11-Line LCD Phones


Error Messages


No Error Messages

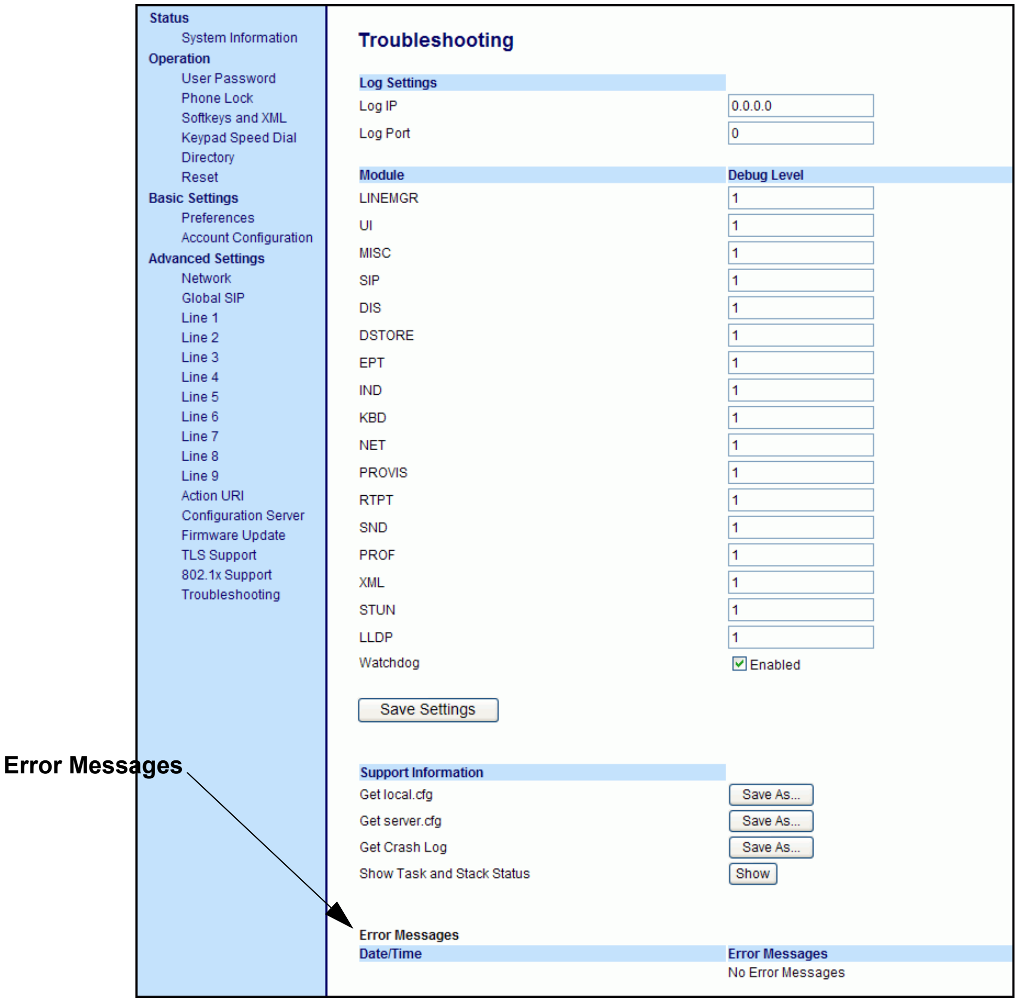
3-Line LCD Phones


Error Messages


No Error Messages</td></tr>
<tr><td>5</td><td>When done viewing, press Done or Exit to exit the Error Messages screen.</td></tr>
</table>

Viewing the Error Messages Using the Aastra Web UI

Use the following procedure to view the error messages, if any, that generated during startup.

<table>
<tr><th colspan="2">Aastra Web UI</th></tr>
<tr><td>1</td><td>Click on Advanced Settings->Troubleshooting->Error Messages.


</td></tr>
<tr><td>2</td><td>Scroll down to the “Error Messages” section to view the error messages that may have generated during startup or reboot of the IP Phone.</td></tr>
</table>

Configuration and Crash File Retreival

In addition to using the Troubleshooting page in the Aastra Web UI, an Administrator can also use three new configuration parameters in the configuration files to enable/disable the uploading of support information to a pre-defined server. These parameters are:

- **upload system info server -** Specifies the server for which the phone sends the system and crash files.
- **upload system info manual option -** Enables and disables the ability to manually upload support information from the IP Phone UI and Aastra Web UI.
- **upload system info on crash -** Enables and disables the watchdog to automatically reboot the phone and send a crash file to the pre-defined server.

When this feature is enabled (configuration files only), support files can be automatically or manually generated and uploaded when the server detects a phone failure. An Administrator or User can manually send the files when required using the IP Phone UI or the Aastra Web UI. Each time the files are generated and uploaded, a new timestamp on the file name is created so that existing files are not overwritten on the server. File names are generated in the format
MAC ID_Date_Time_server.cfg, MAC ID_Date_Time_local.cfg, and
MAC ID_Date_Time_crash.cfg

Notes:
1. The phone performs the generation and sending of Support Information in the background. This feature does not affect the use or operation of the phone.
2. During a startup or reboot of the phone, an upload of Support Information is automatically generated and sent to the pre-defined server.
3. This feature supports the TFTP, FTP, HTTP, and HTTPS Protocols.

The following table identifies the methods you can use to retrieve support information from the phone to the pre-defined server when the above configuration parameters are enabled.

Method for Retrieving Support Info	Description
Configuration Files (automatic retrieval)	1 Enter "**upload system info server**" parameter in the configuration files and specify the server for which the phone sends the system crash information 2 Enter "**upload system info on crash: 1**" to enable the phone to automatically send system crash information to the pre-defined server each time the watchdog reboots.
IP Phone UI (manual retrieval)	1 On the phone, navigate to ***Options->Phone Status->Upload System Info***. 2 Press "**Select**" or "**Enter**". The system information is immediately sent to the pre-defined server and the message "*Files Sent*" displays.
Aastra Web UI (manual retrieval)	1 On the Aastra Web UI, navigate to ***Status->System Information->Support Information***. 2 Press <**Upload**>. The system information is immediately sent to the pre-defined server and the message "*Files Sent*" displays.

When this feature is enabled, the phone sends the following support information files to the server:

- **Server.cfg -** File in the format ***MAC ID_Date_Time_server.cfg*** that contains configuration information from the aastra.cfg and and the <mac>.cfg files. The MAC address, date, and time are specified in the file name to identify the phone sending the information, and the date and time the file was generated and sent to the server.
(for example, **00093D435522_2010-02-25_1141am_server.cfg**)
- **Local.cfg -** File in the format ***MAC ID_Date_Time_local.cfg*** that contains information of locally modified values made using the Aastra Web UI and/or the IP Phone UI. The MAC address, date, and time are specified in the file name to identify the phone sending the information, and the date and time the file was generated and sent to the server.
(for example, **00043D199345_2010-02-26_1030am_local.cfg**)
- **Crash.cfg -** (only generated if an error or crash occurs on the phone) File in the format ***MAC ID_Date_Time_crash.cfg*** that contains information about a current phone error/crash causing a reboot of the phone . The MAC address, date, and time are specified in the file name to identify the phone sending the information, and the date and time the file was generated and sent to the server.
(for example, **00033D000111_2010-02-27_0204pm_crash.cfg**)

Configuring Crash File Retrieval Using the Configuration Files

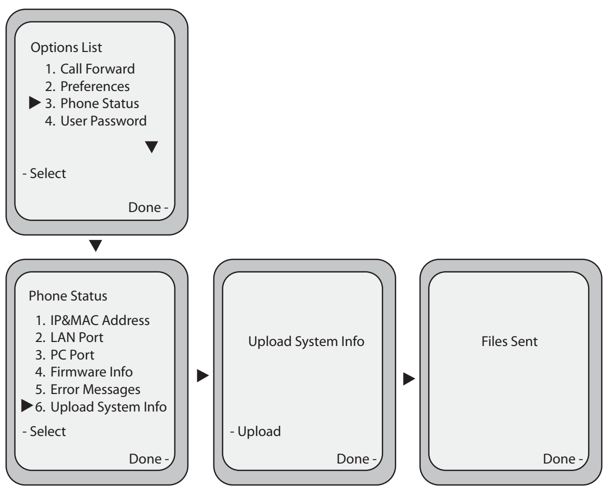
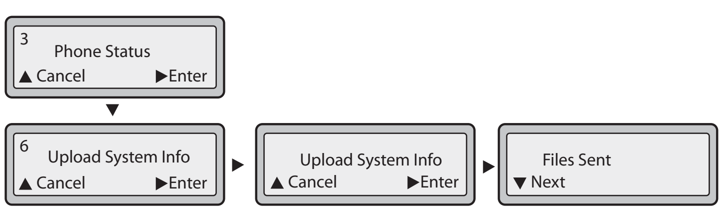
Use the following procedure to configure crash file retrieval from the phone to a server.

Configuration Files

For specific parameters you can set in the configuration files, see Appendix A, the section, "Crash File Retrieval" on page A-258.

Generating and Sending Support Information from the IP Phone UI

Use the following procedure to generate and send Support Information files to the server.

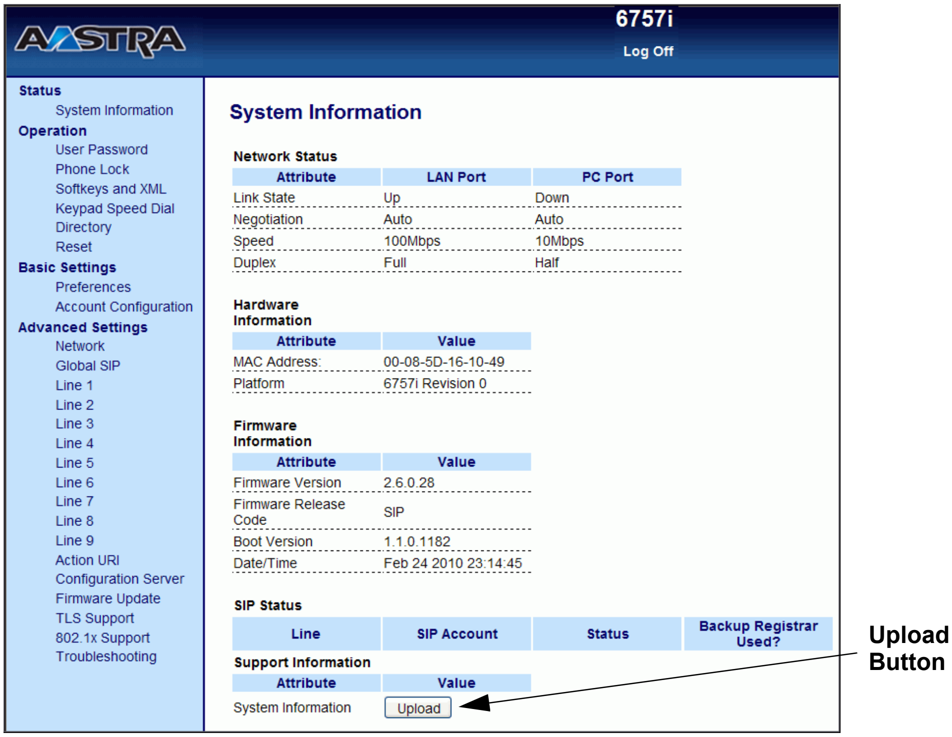
IP Phone UI	
Step	**Action**
For 5-Line Phones:	
1	Press [Options key] on the phone to enter the Options List.
2	Select **Phone Status**.
3	Select **Upload System Info** and press **<Select>.** The phone immediately generates the applicable Support Information files (server.cfg, local. cfg, and/or crash.cfg) and sends the files to the pre-defined server. The message "*Files Sent*" displays. 
For 3-Line Phones:	
1	Press [Options key] on the phone to enter the Options List.
2	Select **Phone Status**.
3	Select **Upload System Info** and press **<Enter>.** The phone immediately generates the applicable Support Information files (server.cfg, local. cfg, and/or crash.cfg) and sends the files to the pre-defined server. The message "*Files Sent*" displays. 

Generating and Sending Support Information from the Aastra Web UI

Use the following procedure to generate and send Support Information files to the server.

Aastra Web UI

1 Click on **Status->System Information-> Support Information.**



2 **Note:** Press <**Upload**>. The phone immediately generates the applicable Support Information files (server.cfg, local. cfg, and/or crash.cfg) and sends the files to the pre-defined server. The message "*Files Sent*" displays.

Limitations

- If sending the Support Information files to a folder on the server, then writing privileges must be allowed for that folder.
- If the Administrator password and username are configured in the Server.cfg file, a User can retrieve that information after the Server.cfg file is loaded to the server.
- TFTP does not report transmission failure if the destination server is down. In this case, the Support Information files are not sent.

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 4-104.

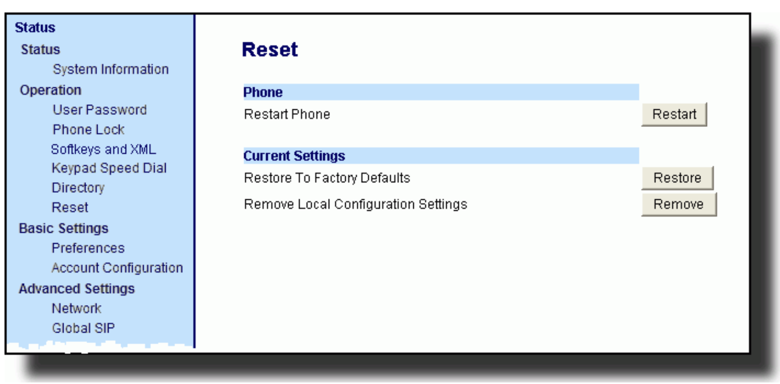
For specific protocol parameters you can set in the configuration files, see Appendix A, the section, "Configuration Server Settings" on page A-18.

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 3-Line LCD Displays::*** Press **#** to confirm. **Note:** To cancel the Restart, press the 3 key. ***For 8 and 11-Line LCD Displays::*** Press **Restart**. **Note:** To cancel the Restart, press **Cancel**.
For the 6739i:	
1	Press on the phone to enter the Options List.
2	Press **Restart**. The following prompt displays: *"Restart the phone?"*
3	Press **<Yes>** to restart the phone or Press **<No>** to go back to the Optons Screen.

Aastra Web UI

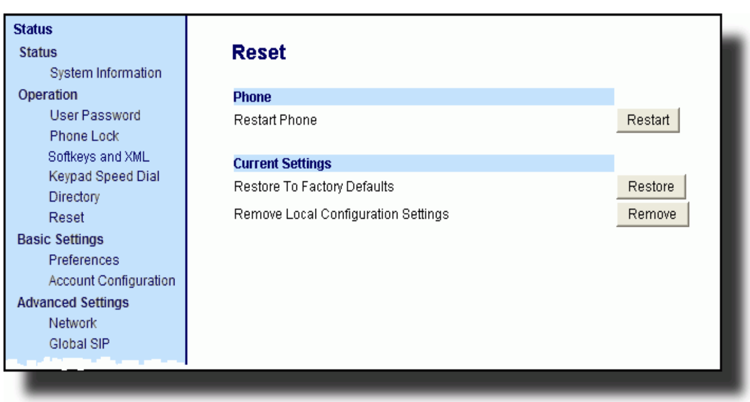
1	Click on **Operation->Reset->Phone.** 
2	Click Restart to restart the phone.

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 3-Line LCD Displays::*** The "*Restore Defaults?*" prompt displays. Press **#** to confirm. ***For 8 and 11-Line LCD Phones:*** The "*Reset phone to factory defaults?*" prompt displays. Press **Default** to confirm.
For the 6739i:	
1	Press on the phone to enter the Options List.
2	Press **Advanced** and enter your Administrator password using the pop-up keyboard that displays. (Default is **22222**).
3	Press **Reset**. The "*Reset Configuration?*" prompt displays.
4	Press **Factory Default**. The phone immediately sets the phone to factory defaults and automatically restarts the phone.

Aastra Web UI

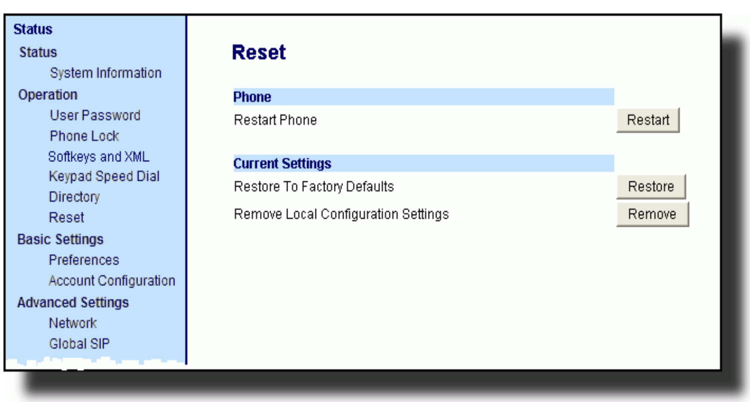
Step	Action
1	Click on **Operation->Reset->Current Settings.** 
2	In the "**Restore to Factory Defaults**" field, click Restore . This restores all factory defaults, and removes any saved configuration and directory list files.

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 3-Line LCD Displays::*** The “*Erase local config?*” prompt displays. Press **#** to confirm. ***For 8 and 11-Line LCD Displays::*** The “*Erase local config?*” prompt displays. Press **Erase** to confirm.
For the 6739i:	
1	Press on the phone to enter the Options List.
2	Press **Advanced** and enter your Administrator password using the pop-up keyboard that displays. (Default is **22222**).
3	Press **Reset**. The “*Reset Configuration?*” prompt displays.
4	Press **Erase Local Cfg.**. The phone immediately erases the local configuration and automatically restarts the phone.

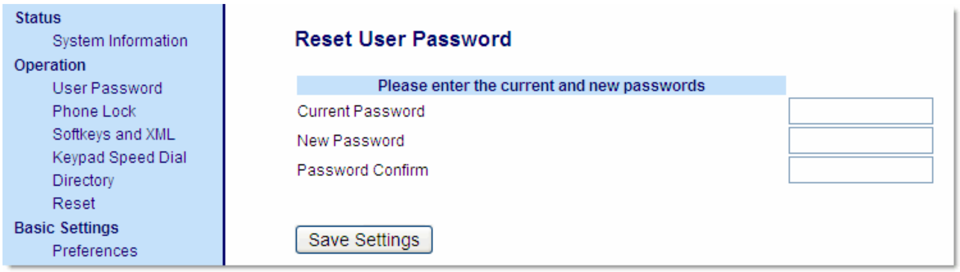
Aastra Web UI

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.
For the 6739i:	
1	Press on the phone to enter the Options List.
2	Press **Password**.
3	Press the **<Current Password>** field. A keyboard displays on the screen.
4	Enter the current user password in the text box and press the **<Enter>** key on the keyboard.
5	Press the **<New Password>** field.
6	Enter the new user password in the text box and press the **<Enter>** key on the keyboard.
7	Press the **<Re-enter Password>** field.
8	Re-enter the new user password in the text box and press the <**Enter>** key on the keyboard.
9	Press <**Save>**. 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?

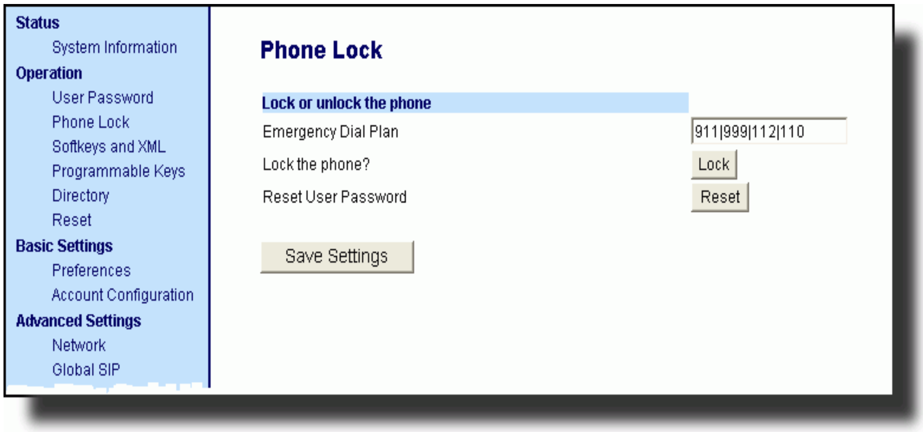
IP Phone UI

Step	Action
Lock the phone:	
1	Press [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 [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.

For 6739i:

IP Phone UI

Step	Action
Lock the phone:	
1	Press the **Options** key on the phone to enter the Options List.
2	Select **Lock.** The prompt, "*Lock the phone*?" displays.
3	Press **Yes** to lock the phone. The phone locks.
Unlock the phone:	
1	Press the **Options** key on the phone to enter the Options List. A "Phone is Locked" screen displays allowing you to press an "Unlock the Phone" button.
2	Press **Unlock the Phone**. A prompt, "Enter Unlock Password" displays as well as a keyboard.
3	Enter the user or administrator password and press **Enter**. Default is "**22222**". A prompt "Unlock the Phone?" displays.
4	Press **Yes** to unlock the phone. 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 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*".

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:

Setting Parameters in Configuration Files

You can set specific configuration parameters in the configuration files for configuring you IP phone. The *aastra.cfg* and *<mac>.cfg* files are stored on the server. The *aastra.cfg* file stores global IP phone configuration settings. The *<mac>.cfg* file stores configuration settings specific to the IP phone with that MAC address. When you restart the IP phone, these files are downloaded to the phone.

If you make changes to the phone configuration, the changes are stored in a local configuration on the phone (not on the server).

Configuration changes made to the *<mac>.cfg* file override the configuration settings in the *aastra.cfg* file.

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 -37.

This section includes the following types of configurable parameters:

- Operational, Basic, and Advanced Parameters on page A-6
- Mapping Key Settings on page A-199
- Softkey/Programmable Key/Feature Key/Expansion Module Key Parameters on page A-202
- Advanced Operational Parameters on page A-238
- Troubleshooting Parameters on page A-255

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** **Services (3-Line LCD phones)** **Tones** **Contrast** **Set Audio (8 and 11-Line LCD phones only)** **Handset Pairing (9480i CT and 6757i CT)**
Status	**Status**
Password	Removed
Administrator Menu	Removed
Restart Phone	Removed
Phone Lock	**Phone Lock**

For 6739i

Full Options Menu	**Simplified Options Menu**
Audio	**Audio**
Display	Display
Set Time	**Removed**
Language	Removed
Bluetooth	Removed
Softkeys	**Removed**
Status	**Status**
Advanced	**Removed**
Password	**Removed**
Restart	**Removed**
Lock	**Lock**

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

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,, Domain Name Servers (DNS), TFTP, HTTP, HTTP Port, HTTPS, HTTPS Port, 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.	
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. **Note:** For DHCP to automatically populate this parameter, your DHCP server must support Option 66.	
Format	IP address	
Default Value	0.0.0.0	
Range	Not Applicable	
Example	ip: 192.168.0.25	

Parameter – *subnet mask* *Subnet Mask* (in Web UI)	**IP phone UI** Options->Administrator Menu-> Network Settings **Aastra Web UI** Advanced Settings->Network-> Basic Network Settings **Configuration Files** aastra.cfg, <mac>.cfg
Description	Subnet mask defines the IP address range local to the IP phone. **Note:** For DHCP to automatically populate this parameter, your DHCP server must support Option 66.
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** Options->Administrator Menu-> Network Settings **Aastra Web UI** Advanced Settings->Network-> Basic Network Settings **Configuration Files** aastra.cfg, <mac>.cfg
Description	The IP address of the network's gateway or default router IP address. **Note:** For DHCP to automatically populate this parameter, your DHCP server must support Option 66.
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. **Note:** For DHCP to automatically populate this parameter, your DHCP server must support Option 66.	
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. **Note:** For DHCP to automatically populate this parameter, your DHCP server must support Option 66.	
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	ethernet port 0: 3	

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. **Note:** PC Port parameters are not applicable to the 6730i IP Phone.	
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	

Parameter – *pc port passthru enabled*	**IP phone UI**	Options->Administrator Menu-> Network Settings->Ethernet Link
	Aastra Web UI	Advanced Settings->Network-> Basic Network Settings
	Configuration Files	aastra.cfg, <mac>.cfg
Description	Enables or disables the PC port. **Note:** PC Port parameters are not applicable to the 6730i IP Phone.	
Format	Integer	
Default Value	1 (enable)	
Range	0 (disable) 1 (enable)	
Example	pc port passthru enabled: 1	

DHCP Option Settings

Option 12

Parameter– *hostname* *Hostname* (in Web UI)	**Configuration Files** **IP Phone UI** **Aastra Web UI**	aastra.cfg, <mac>.cfg Options->Administrator Menu-> Network Settings->Hostname Advanced Settings->Network-> Basic Network Settings
Description	Specifies the hostname DHCP Option 12 that the phone sends with the DHCP Request packet. **Note:** If you change this parameter, you must restart your phone for the change to take affect.	
Format	String	
Default Value	[<model><MAC IP Address>]	
Range	Up to 64 alpha-numeric characters **Note:** The value for this parameter can also be a fully qualified domain name.	
Example	hostname: aastra4	

Option 77

Parameter– *dhcp userclass* *DHCP User Class* (in Web UI)	**Configuration Files** **IP Phone UI** **Aastra Web UI**	aastra.cfg, <mac>.cfg Options->Administrator Menu-> Network Settings->DHCP Settings-> DHCP User Class Advanced Settings->Network-> Advanced Network Settings
Description	Specifies the User Class DHCP Option 77 that the phone sends to the configuration server with the DHCP Request packet. **Note:** If you specify a value for this parameter, you must restart your phone for the change to take affect. Any change in its value during start-up results in an automatic reboot.	
Format	String	
Default Value	“”	
Range	Up to 64 alpha-numeric characters	
Example	dhcp userclass: admin	

Options 159 and 160 - DHCP Option Override

Parameter– *dhcp config option override* *DHCP Download Options* (in Web UI)	**Configuration Files** **IP Phone UI** **Aastra Web UI**	aastra.cfg, <mac>.cfg Options->Administrator Menu-> Network Settings->DHCP Settings-> Download Options Advanced Settings->Network-> Advanced Network Settings
Description	The value specified for this parameter overrides the precedence order for determining a configuration server. **Note:** You must restart the IP Phone for this parameter to take affect.	
Format	Integer	
Default Value	0 (Any - no override - uses normal precedence order of 43, 160, 159, 66)	
Range	-1 (Disabled - ignores all DHCP configuration options (43, 66, 159, 160) 0 (Any) 43 66 159 160	
Example	dhcp config option override: 66	

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	0-4294967295	
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	0-4294967295	
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. **Dial plan(characters)** **Length (bytes)** 911 14 4xx 18 x+#|xx+* 35 911|999|112|110|450 54 911|112|011XX+#|101XX+#|1[2-3]XXXXXXXXX|[4-5]XXXXXXXXX|[6-7]XXXXXXXXXX,3|[8-9]XXXXXXXXXXX,2|XX+*|XX+#|*XXX+#|#XX+#|4xx,2 325 **Note:** Contact your local phone service provider for available emergency numbers in your area.
Format	Integer
Default Value	x+#|xx+*
Range	Up to 512 characters
Example	emergency dial plan: 911|999

User Dial Plan Setting

Parameter – *sip user parameter dial plan*	**Configuration Files**aastra.cfg, <mac>.cfg
Description	The dial plan that the network uses to distinguish the phone's username that looks like a PSTN number, from the real PSTN number. Note: You can configure the "sip user parameter dial plan" parameter on a global basis only. If it is misconfigured, then the parameter is ignored. Entering no value disables this feature.
Format	Alpha-numeric characters
Default Value	Blank
Range	Up to 512 characters; More than 512 characters disables this parameter.
Example	sip user parameter dial plan: 6xx | 8xxxx |9xxxxxxx

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 intercae for a single IP phone when this parameter is entered in the *<mac>.cfg* file. Enables or disables the Aastra Web UI interface for all phones when this parameter is placed in the *aastra.cfg* file.
Format	Integer
Default Value	1 (admin/user enabled)
Range	0 (admin/user disabled) 1 (admin/user enabled) 2 (only admin enabled)
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. 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. **Note:** For DHCP to automatically populate this parameter, your DHCP server must support Option 66.	
Format	IP address or qualified domain name	
Default Value	0.0.0.0	
Range	Not Applicable	
Example	tftp server: 192.168.0.130	

Parameter – *tftp path*	**Aastra Web UI** **IP Phone UI** **Configuration Files**	Advanced Settings-> Configuration Server->Settings Options->Administrator Menu-> Configuration Server->TFTP Settings aastra.cfg, <mac>.cfg
Description	Specifies the path name for which the configuration files reside on the TFTP server for downloading to the IP Phone. **Note:** Enter the path name in the form *folderX\folderX\folderX*. For example, ***ipphone\6757i\configfiles***.	
Format	String	
Default Value	Not Applicable	
Range	Up to 64 alphanumeric characters	
Example	tftp path: configs\tftp	

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 – *alternate tftp path*	**Aastra Web UI**	Advanced Settings-> Configuration Server->Settings
	IP Phone UI	Options->Administrator Menu-> Configuration Server->TFTP Settings
	Configuration Files	aastra.cfg, <mac>.cfg
Description	Specifies a path name for which the configuration files reside on an alternate TFTP server for downloading to the IP Phone. **Note:** Enter the path name in the form *folderX\folderX\folderX*. For example, ***ipphone\6757i\configfiles***.	
Format	String	
Default Value	Not Applicable	
Range	Up to 64 alphanumeric characters	
Example	alternate tftp path: configs\alternate	

Parameter – *use alternate tftp* *Use Alternate TFTP* (in Web UI)	**IP phone UI**	Options->Administrator Menu-> Network Settings->TFTP Server->Select TFTP
	Aastra Web UI	Advanced Settings->Configuration Server
	Configuration Files	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** Options->Administrator Menu-> Network Settings->FTP Server **Aastra Web UI** Advanced Settings->Configuration Server **Configuration Files** 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. **Note:** For DHCP to automatically populate this parameter, your DHCP server must support Option 66.
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 path*	**Aastra Web UI** Advanced Settings-> Configuration Server->Settings **IP Phone UI** Options->Administrator Menu-> Configuration Server->FTP Settings **Configuration Files** aastra.cfg, <mac>.cfg
Description	Specifies a path name for which the configuration files reside on an FTP server for downloading to the IP Phone. **Note:** Enter the path name in the form *folderX\folderX\folderX*. For example, ***ipphone\6757i\configfiles***.
Format	String
Default Value	Not Applicable
Range	Up to 64 alphanumeric characters
Example	ftp path: configs\ftp

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: 6757iaastra	

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	

<table>
<tr><td>Parameter –
http server

HTTP Server
(in Web UI)</td><td>IP phone UI

Aastra Web UI
Configuration Files</td><td>Options->Administrator Menu->
Network Settings->HTTP Server
Advanced Settings->Configuration Server
aastra.cfg, <mac>.cfg</td></tr>
<tr><td>Description</td><td colspan="2">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).

Note: For DHCP to automatically populate this parameter, your DHCP server must support Option 66.</td></tr>
<tr><td>Format</td><td colspan="2">IP address or fully qualified Domain Name</td></tr>
<tr><td>Default Value</td><td colspan="2">0.0.0.0</td></tr>
<tr><td>Range</td><td colspan="2">Not Applicable</td></tr>
<tr><td>Example</td><td colspan="2">http server: 192.168.0.132</td></tr>
</table>

<table>
<tr><td>Parameter –
http path

HTTP Path
(in Web UI)</td><td>IP phone UI

Aastra Web UI
Configuration Files</td><td>Options->Administrator Menu->
Network Settings->HTTP Server
Advanced Settings->Configuration Server
aastra.cfg, <mac>.cfg</td></tr>
<tr><td>Description</td><td colspan="2">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.</td></tr>
<tr><td>Format</td><td colspan="2">dir/dir/dir</td></tr>
<tr><td>Default Value</td><td colspan="2">Not Applicable</td></tr>
<tr><td>Range</td><td colspan="2">Up to 63 alphanumeric characters</td></tr>
<tr><td>Example</td><td colspan="2">http path: ipphones/6757i</td></tr>
</table>

Parameter – *http port* *HTTP Port* (in Web UI)	**IP Phone UI:** **Aastra Web UI:** **Configuration Files**	Options->Administrator Menu-> Configuration Server->HTTP Settings Advanced Settings->Configuration Server-> Settings aastra.cfg, <mac>.cfg
Description	Specifies the HTTP port that the server uses to load the configuration to the phone over HTTP. **Note:** For DHCP to automatically populate this parameter, your DHCP server must support Option 66.	
Format	Integer	
Default Value	80	
Range	1 through 65535	
Example	http port: 1025	

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**). **Note:** For DHCP to automatically populate this parameter, your DHCP server must support Option 66.	
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/6755i	

Parameter – *https port* *HTTPS Port* (in Web UI)	**IP Phone UI:** **Aastra Web UI:** **Configuration Files**	Options->Administrator Menu-> Configuration Server->HTTPS Settings Advanced Settings->Configuration Server-> Settings aastra.cfg, <mac>.cfg
Description	Specifies the HTTPS port that the server uses to load the configuration to the phone over HTTPS. **Note:** For DHCP to automatically populate this parameter, your DHCP server must support Option 66.	
Format	Integer	
Default Value	443	
Range	1 through 65535	
Example	https port: 1025	

<table>
<tr><td>Parameter –
auto resync mode

Mode
(in Web UI)</td><td>Aastra Web UI

Configuration Files</td><td>Advanced Settings->Configuration Server->
Auto-Resync
aastra.cfg, <mac>.cfg</td></tr>
<tr><td>Description</td><td colspan="2">Determines whether the configuration server automatically updates the configuration files only, the firmware only, both the firmware and configuration files, or disables automatic updates.
This parameter works with TFTP, FTP, HTTP and HTTPS 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.</td></tr>
<tr><td>Format</td><td colspan="2">Integer</td></tr>
<tr><td>Default Value</td><td colspan="2">Aastra Web UI
None

Configuration Files
0</td></tr>
<tr><td>Range</td><td colspan="2">Aastra Web UI
None
Configuration Files
Firmware
Both

Configuration Files
0 (none)
1 (configuration files only)
2 (firmware only)
3 (configuration files and firmware)</td></tr>
<tr><td>Example</td><td colspan="2">auto resync mode: 1</td></tr>
</table>

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, HTTP and HTTPS 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	

Parameter– *auto resync max delay* *Maximum Delay* (in Web UI)	**Configuration Files** **Aastra Web UI**	aastra.cfg, <mac>.cfg Advanced Settings->Configuration Server-> Auto-Resync
Description	Specifies the maximum time, in minutes, the phone waits past the scheduled time before starting a checksync.	
Format	Integer	
Default Value	15	
Range	0-1439	
Example	auto resync max delay: 20	

Parameter– *auto resync days* *Days* (in Web UI)	**Configuration Files** **Aastra Web UI**	aastra.cfg, <mac>.cfg Advanced Settings->Configuration Server-> Auto-Resync
Description	Specifies the amount of days that the phone waits between checksync operations. **Note:** A value of 0 causes the phone to checksync every time the clock reads the proper time. A value of 1 forces the phone to wait 24 hours prior to doing the first checksync.	
Format	Integer	
Default Value	0	
Range	0-364	
Example	auto resync days: 1	

Multiple Configuration Server Settings

Parameter – *firmware server*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	Specifies the URL of a server (other than the original configuration server) from which the phones in the network get their firmware. **Note:** The default method for the update of firmware to the phones is from the original configuration server. The Administrator must specify a correct server URL for the phones to get their firmware information from that server. If the URL is incorrect, no firmware download occurs to the phones from the specified server.
Format	String (up to 256 characters) **FTP** "ftp://username:password@0.0.0.0:port/path" **TFTP** "tftp://0.0.0.0:port/path" **HTTP** "http://0.0.0.0:port/path" **HTTPS** "https://0.0.0.0:port/path"
Default Value	Blank
Range	Not Applicable
Example	**firmware server:** Leaving this parameter blank downloads all configuration and firmware files from the original configuration server. **firmware server: tftp://10.30.102.158/test1** The above example uses TFTP to download all firmware files that exist in the "test1" directory on the specified server, to the phones.

Additional Information

The directory files, language packs, TLS certificate files, 802.1x certificate files, and HTTPS files can also be downloaded to the phone from a server other than the configuration server. For each of these types of files, you can specify a URL (server IP address) from which the phone gets these files. You can use existing parameters on the phone to specify the URL. For more information about this feature, refer to Chapter 1, the section, "Directory Files, Language Packs, TLS Certificates, 802.1x Certificates, HTTPS Files and Multiple Configuration Servers" on page1-39.

For information on configuring the directory, language pack, TLS certificates, 802.1x certificates, and HTTPS parameters, see the applicable parameters in this Appendix.

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 of 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. **Note:** The phones support decoding and playing out DTMF tones sent in SIP INFO requests. The following DTMF tones are supported: • Support signals 0-9, #, * • Support durations up to 5 seconds	
Format	Integer	
Default Value	51720	
Range	Not Applicable	
Example	sip nat rtp port: 51730	

Rport Setting

Parameter – *sip rport*	**Aastra Web UI:** **Configuration Files**	Advanced Settings->Network-> Advanced Network Settings aastra.cfg, <mac>.cfg
Description	Allows you to enable (1) or disable (0) the use of Rport on the IP phone. “Rport” in RFC 3581, allows a client to request that the server send the response back to the source IP address **and** the port from which the request came. **Note:** Configuring the Rport parameter is recommened for clients behind a Network Address Translation (NAT) or firewall.	
Format	Boolean	
Default Value	0	
Range	0 (disable) 1 (enable)	
Example	sip rport: 1	

Local SIP UDP/TCP Port Setting

Parameter – *sip local port*	**Aastra Web UI:** Advanced Settings->Global SIP->Advanced SIP Settings **Configuration Files** aastra.cfg, <mac>.cfg
Description	Specifies the local source port (UDP/TCP) from which the phone sends SIP messages.
Format	Numeric
Default Value	5060
Range	Greater than 1024 and less than 65535 **Notes**: **1.** It is recommended that you avoid the conflict RTP port range in case of a UDP transport. **2.** 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.If symmetric UDP signaling is disabled, the phone sends from random ports but it listens on the configured SIP local port.
Example	sip local port: 5060

Local SIP TLS Port

Parameter – *sip local tls port*	**Aastra Web UI:** Advanced Settings->Global SIP->Advanced SIP Settings **Configuration Files** aastra.cfg, <mac>.cfg
Description	Specifies the local source port (SIPS/TLS) from which the phone sends SIP messages.
Format	Numeric
Default Value	5061
Range	Greater than 1024 and less than 65535 **Note:** It is recommended that you avoid the conflict with any TCP ports being used. For example: WebUI HTTP server on 80/tcp and HTTPS on 443/tcp.
Example	sip local tls port: 5061

SIP STUN Parameters

Parameter – *sip stun ip*	**Aastra Web UI:** Advanced Settings->Network-> Advanced Network Settings **Configuration Files** aastra.cfg, <mac>.cfg
Description	IP address and/or qualified domain name of the **STUN** server (also know as Simple Traversal of UDP through NAT). **STUN** is a protocol that governs the exchange of data over a UDP connection by communication devices operating behind a NAT or firewall. Since the behaviors of NATs can be different, the STUN protocol allows the exchange of data to be consistent over any type of NAT. The STUN/TURN configuration is only used for media (RTP traffic) that goes through the server - not for signaling. (For signaling, you need to enable Rport if the NAT device does not recognized SIP). The STUN/TURN configuration applies globally to each phone. If you configure both STUN and TURN on the phone, it discovers what type of NAT device is between the phone and the public network. If the NAT device is full cone, restricted cone, or port restricted cone, the phone uses STUN. If the NAT device is symmetric, the phone uses TURN. If you configure STUN only, the phone uses STUN without the NAT discovery process. **Notes**: **1.** The NAT IP configuration parameter takes precedence over the STUN and TURN parameters. **2.** STUN does not work if the NAT device is symmetric.
Format	IP Address or Fully Qualified Domain Name (FQDN)
Default Value	0.0.0.0
Range	Two (2) IP addresses or domain names separated by a comma **Note:** The first is the primary and the second is the backup.
Example	sip stun ip: 10.50.103.12, stunbackup.aastra.com

Parameter – *sip stun port*	**Aastra Web UI:** Advanced Settings->Network-> Advanced Network Settings **Configuration Files** aastra.cfg, <mac>.cfg
Description	Port number of the **STUN** server (also know as Simple Traversal of UDP through NAT) **Notes**: **1.** The NAT IP configuration parameter takes precedence over the STUN and TURN parameters. **2.** STUN does not work if the NAT device is symmetric.
Format	Integer
Default Value	3478
Range	0 to 65535 (One (1) or two (2) port numbers separated by a comma) **Note:** The first is the primary and the second is the backup.
Example	sip stun port: 3478,3479

SIP TURN Parameters

Parameter – *sip turn ip*	**Aastra Web UI:** Advanced Settings->Network-> Advanced Network Settings **Configuration Files** aastra.cfg, <mac>.cfg
Description	IP address and/or qualified domain name of the **TURN** server (also known as Traversal Using Relay NAT). TURN is a protocol that governs the reception of data over a Transmission Control Protocol (TCP) or a UDP connection by a single communications device operating behind a NAT or firewall. A TURN server relays packets from an external IP address towards an internal device only if that internal device has previously sent a packet through the same TURN server to that particular external IP address. The STUN/TURN configuration is only used for media (RTP traffic) that goes through the server - not for signaling. (For signaling, you need to enable Rport if the NAT device does not recognized SIP). The STUN/TURN configuration applies globally to each phone. If you configure both STUN and TURN on the phone, it discovers what type of NAT device is between the phone and the public network. If the NAT device is full cone, restricted cone, or port restricted cone, the phone uses STUN. If the NAT device is symmetric, the phone uses TURN. If you configure TURN only, the phone uses TURN with the NAT discovery process. TURN is compatible with all types of NAT devices but can be costly since all traffic goes through a media relay (which can be slow, can exchange more messages, and requires the TURN server to allocate bandwidth for calls). **Note:** The NAT IP configuration parameter takes precedence over the STUN and TURN parameters.
Format	IP Address or Fully Qualified Domain Name (FQDN)
Default Value	0.0.0.0
Range	Two (2) IP addresses or domain names separated by a comma **Note:** The first is the primary and the second is the backup.
Example	sip turn ip: 10.50.103.12, turnbackup.aastra.com

Parameter – *sip turn port*	**Aastra Web UI:** Advanced Settings->Network-> Advanced Network Settings **Configuration Files** aastra.cfg, <mac>.cfg
Description	Port number of the **TURN** server (also known as Traversal Using Relay NAT). **Note:** The NAT IP configuration parameter takes precedence over the STUN and TURN parameters.
Format	Integer
Default Value	3479
Range	0-65535 (One (1) or two (2) port numbers separated by a comma) **Note:** The first is the primary and the second is the backup.
Example	sip turn port: 3479,3480

Parameter – *sip turn user*	**Aastra Web UI:** Advanced Settings->Network-> Advanced Network Settings **Configuration Files** aastra.cfg, <mac>.cfg
Description	Username that a user must enter when accessing an account on the TURN server. **Note:** The NAT IP configuration parameter takes precedence over the STUN and TURN parameters.
Format	Alphanumeric characters
Default Value	Not Applicable
Range	Up to 63 alphanumeric characters
Example	sip turn user: 0412919146

Parameter – *sip turn pasx*	**Aastra Web UI:** Advanced Settings->Network-> Advanced Network Settings **Configuration Files** aastra.cfg, <mac>.cfg
Description	Password that a user must enter when accessing an account on the TURN server. **Note:** The NAT IP configuration parameter takes precedence over the STUN and TURN parameters.
Format	Alphanumeric characters
Default Value	Not Applicable
Range	Up to 63 alphanumeric characters
Example	sip turn pass: 42447208233b8b8b8a234

SIP Keep Alive Support

Parameter – *sip keepalive timer*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	This value is how many seconds to wait before sending a SIP UDP keep alive packet to the configured SIP servers. A zero value disables this feature. **Note:** This is only for UDP transport protocol.
Format	Integer
Default Value	0
Range	Any positive integer
Example	sip keepalive timer: 6

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	

HTTPS Server Certificate Validation Settings

Parameter– *https validate certificates* *Validate Certificates* (in Web UI)	**Configuration Files** **IP Phone UI** **Aastra Web UI**	aastra.cfg, <mac>.cfg Options->Administrator Menu-> Configuration Server->HTTPS Settings-> Cert Validation Advanced Settings->Network->HTTPS Settings
Description	Enables or disables the HTTPS validation of certificates on the phone. When this parameter is set to 1, the HTTPS client performs validation on SSL certificates before accepting them. **Note: If you are using HTTPS as a configuration method, and use a self signed certificate, you must set this parameter to "0" (disabled) before upgrading to Release 2.3 of the IP Phones.**	
Format	Boolean	
Default Value	1 (enabled)	
Range	0 (disabled) 1 (enabled)	
Example	https validate certificates: 0	

Parameter– *https validate expires* *Check Certificate Expiration* (in Web UI)	**Configuration Files** aastra.cfg, <mac>.cfg **IP Phone UI** Options->Administrator Menu->Configuration Server->HTTPS Settings->Cert Validation->Check Expires **Aastra Web UI** Advanced Settings->Network->HTTPS Settings
Description	Enables or disables the HTTPS validation of the expiration of the certificates. When this parameter is set to 1, the HTTPS client verifies whether or not a certificate has expired prior to accepting the certificate. **Note:** If the "**https validate expires**" parameter is set to **enable**, the clock on the phone must be set for the phone to accept the certificates.
Format	Boolean
Default Value	1 (enabled)
Range	0 (disabled) 1 (enabled)
Example	https validate expires: 0

Parameter– *https validate hostname* *Check Certificate Hostnames* (in Web UI)	**Configuration Files** aastra.cfg, <mac>.cfg **IP Phone UI** Options->Administrator Menu->Configuration Server->HTTPS Settings->Cert Validation->Check Hostnames **Aastra Web UI** Advanced Settings->Network->HTTPS Settings
Description	Enables or disables the HTTPS validation of hostnames on the phone.
Format	Boolean
Default Value	1 (enabled)
Range	0 (disabled) 1 (enabled)
Example	https validate hostname: 0

Parameter– *https user certificates* *Trusted Certificates Filename* (in Web UI)	**Configuration Files** **Aastra Web UI**	aastra.cfg, <mac>.cfg Advanced Settings->Network->HTTPS Settings
Description	Specifies a file name for a .PEM file located on the configuration server. This file contains the User-provided certificates in PEM format. These certificates are used to validate peer certificates. **Note:** You must disable the "**https validate certificates**" parameter in order for the phone to accept the User-provided certificates. You can use this parameter in three ways: • To download no certificates • To download a certificate from the original configuration server • To download a certificate from another specified server To download a specific file, the string value MUST HAVE A FILENAME at the end of the string. For example: https user certificates: ftp://admin:admin!@1.2.3.4:50/path/phonesHTTPSUserCert.pem where "path" is the directory and "phonesHTTPSUserCert.pem" is the filename. If you do not specify a filename, the download fails. See examples for each below.	
Format	Alphanumeric string in the format <filename.pem>	
Default Value	Not Applicable	
Range	Not Applicable	
Example	The following example downloads no HTTPS user certificate file: https user certificates: The following example downloads the HTTPS user certificate file from the original configuration server. https user certificates: phonesHTTPSUserCert.pem The following example uses FTP to download the firmware file "phonesHTTPSUserCert.pem" (HTTPS user certificate file) from the "path" directory on server 1.2.3.4 using port 50: https user certificates:ftp://admin:admin!@1.2.3.4:50/path/phonesHTTPSUserCert.pem	

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 **Configuration Files**	Advanced Settings->Network->VLAN->Port 0 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`
	
	Note: PC Port parameters are not applicable to the 6730i IP Phone.	

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	

RTCP Summary Reports

Global Parameters

Parameter – *sip rtcp summary reports*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	Specifies whether or not to send of RTCP summary reports. **Note:** You must restart the phone after setting a value for this parameter.
Format	Boolean
Default Value	0
Range	0 (disabled - OFF) 1 (enabled - ON)
Example	sip rtcp summary reports: 1

Parameter – *sip rtcp summary report collector*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	Specifies the hostname server for which to send (collect) the RTCP summary reports. **Note:** You must restart the phone after setting a value for this parameter.
Format	<username>@<server> **Note:** Hostname/server string must not exceed 128 characters in length.
Default Value	Not Applicable
Range	Not Applicable
Example	sip rtcp summary report collector: collector@example.org

Parameter – *sip rtcp summary report collector port*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	Specifies the port address of the hostname server receiving the RTCP summary reports. **Note:** You must restart the phone after setting a value for this parameter.
Format	Integer
Default Value	0
Range	0 to 65536
Example	sip rtcp summary report collector port: 5060

Line Parameters

Parameter – *sip lineN rtcp summary reports* (N is a line number from 1 to 9)	**Configuration Files** aastra.cfg, <mac>.cfg
Description	Enables/disables the specified line number on the phone for which to send the RTCP summary reports. **Note:** You must restart the phone after setting a value for this parameter.
Format	Boolean
Default Value	0 (disabled)
Range	0 (disabled) 1 (enabled)
Example	sip line1 rtcp summary reports: 1

Parameter – *sip lineN rtcp summary report collector* (N is a line number from 1 to 9)	**Configuration Files** aastra.cfg, <mac>.cfg
Description	Per-line parameter specifying the hostname of the server receiving the RTCP summary reports. **Note:** You must restart the phone after setting a value for this parameter.
Format	<username>@<server> **Note:** Hostname/server string must not exceed 128 characters in length.
Default Value	Not Applicable
Range	Not Applicable
Example	sip line1 rtcp summary report collector: collector@example.org

Parameter – *sip lineN rtcp summary report collector port* (N is a line number from 1 to 9)	**Configuration Files** aastra.cfg, <mac>.cfg
Description	Per-line parameter specifying the port address of the server receiving the RTCP summary reports. **Note:** You must restart the phone after setting a value for this parameter.
Format	Integer
Default Value	0
Range	0 to 65536
Example	sip line1 rtcp summary report collector port: 5060

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	0-63	
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	0-63	
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	0-63	
Example	tos rtcp: 3	

Time and Date Settings

Parameter – *time format* *Time Format* (in Phone UI)	**IP phone UI** Options->Time and Date->Time Format **Configuration Files** aastra.cfg, <mac>.cfg
Description	This parameter changes the time to 12 hour or 24 hour format. Use “0” for the 12 hour format and “1” for the 24 hour format.
Format	Integer
Default Value	0
Range	0 (12 hr format)) 1 (24 hr format)
Example	time format: 0

Parameter – *date format* *Date Format* (in Phone UI)	**IP phone UI** Options->Time and Date->Date Format **Configuration Files** aastra.cfg, <mac>.cfg
Description	This parameter allows the user to change the date to various formats.
Format	Integer
Default Value	0
Range	0 (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) 8 (DD MMM YYYY) 9 (WWW DD MMM) 10 (DD MMM) 11 (DD.MM.YYYY)
Example	date format: 7

Parameter – *dst config* *Daylight Savings* (in Phone UI)	**IP phone UI** Options->Time and Date->Daylight Savings **Configuration Files** 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

Time Zone Name

Parameter – *time zone name* *Time Zone* (in Phone UI)	**IP Phone UI** Options->Time and Date->Time Zone->Others **Configuration Files** aastra.cfg, <mac>.cfg
Description	Assigns a time zone name to the time server. The **Custom** option allows you to customize additional time zone parameters. The **DP-Dhcp** option allows you to enable and disable a DHCP Option 2 value for the phone to use as an offset from Coordinated Universal Time (UTC). If this parameter is enabled, the phone derives the time and date from UTC and the time offset offered by the DHCP server. **Notes:** **1.** Assigning the name "**Custom**" (with initial cap) in the configuration files, allows you to create a custom time zone using the additional parameters in the section "Custom Time Zone and DST Settings" on page A-57. **2.** When DHCP Option 2 is enabled on the phone, the phone still uses the custom timezone configuration settings to control daylight savings time. **3.** The default behavior for the phone is to use the NTP server from Option 42 (or current configuration setting) and the current timezone settings.
Format	Text
Default Value	US-Eastern **Note:** If the time zone name parameter is set to a value other than Dhcp, then DHCP Option 2 is disabled.
Range	See "Time Zone Name/Time Zone Code Table" on page A-51 for specific time zone names. **Custom** (allows you to create a customized time zone). **DP-Dhcp** (allows you to enable and disable a DHCP Option 2 value for the phone to use as an offset from Coordinated Universal Time (UTC))
Example	time zone name: US-Central time zone name: Custom time zone name: DP-Dhcp

Time Zone Name/Time Zone Code Table

Time Zone Name	Time Zone Code
AD-Andorra	CET
AE-Dubai	GST
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
AZ-Baku	AZT
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 CA-Atlantic CA-Eastern CA-Saskatchewan CA-Central CA-Mountain CA-Pacific CA-Yukon CH-Zurich CK-Rarotonga CL-Santiago CL-Easter CN-China CO-Bogota CR-Costa Rica CU-Havana CY-Nicosia CZ-Prague	NST AST EST EST CST MST PST PST CET CKS CLS EAS CST COS CST CST EES CET
DE-Berlin DK-Copenhagen DM-Dominica DO-Santo Domingo Dhcp	CET CET AST AST DP
EE-Tallinn ES-Madrid ES-Canary	EET CET WET
FI-Helsinki FJ-Fiji FK-Stanley FO-Faeroe FR-Paris	EET NZT FKS WET CET
GB-London GB-Belfast GD-Grenada GE-Tbilisi GF-Cayenne GI-Gibraltar GP-Guadeloupe GR-Athens GS-South Georgia GT-Guatemala GU-Guam GY-Guyana	GMT GMT AST GET GFT CET AST EET GST CST CST GYT
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

Time Zone Name	Time Zone Code
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 MU-Mauritius MX-Mexico City MX-Cancun MX-Merida MX-Monterrey MX-Mazatlan MX-Chihuahua MX-Hermosillo MX-Tijuana	CET EET CET AST AST CET MUT 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
OM-Muscat	GST
PA-Panama PE-Lima PL-Warsaw PR-Puerto Rico PT-Lisbon PT-Madeira PT-Azores PY-Asuncion	EST PES CET AST WET WET AZO PYS
RO-Bucharest RU-Kaliningrad RU-Moscow RU-Samara RU-Yekaterinburg RU-Omsk RU-Novosibirsk RU-Krasnoyarsk RU-Irkutsk RU-Yakutsk RU-Vladivostok RU-Sakhalin RU-Magadan RU-Kamchatka RU-Anadyr	EET EET MSK SAM YEK OMS NOV KRA IRK YAK VLA SAK MAG PET ANA

Time Zone Name	Time Zone Code
SE-Stockholm SG-Singapore SI-Ljubljana SK-Bratislava SM-San Marino SR-Paramaribo SV-El Salvador	CET SGT CET CET CET SRT CST
TR-Istanbul TT-Port of Spain TW-Taipei	EET AST CST
UA-Kiev US-Eastern US-Central US-Mountain US-Pacific US-Alaska US-Aleutian US-Hawaii UY-Montevideo	EET EST CST MST PST AKS HAS HST UYS
VA-Vatican	CET
YU-Belgrade	CET

Time Server Settings

Parameter – *time server disabled* *NTP Time Servers* (in Web UI)	**Aastra Web UI** **IP Phone UI'** **Configuration Files**	Basic Settings->Preferences-> Time and Date Settings Options->Preferences-> Time and Date->Time Servers 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). **Note:** For DHCP to automatically populate this parameter, your DHCP server must support Option 66.	
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** **IP Phone UI'** **Configuration Files**	Basic Settings->Preferences-> Time and Date Settings Options->Preferences-> Time and Date->Time Servers 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. **Note:** For DHCP to automatically populate this parameter, your DHCP server must support Option 66.	
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** **IP Phone UI'** **Configuration Files**	Basic Settings->Preferences-> Time and Date Settings Options->Preferences-> Time and Date->Time Servers 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** **IP Phone UI'** **Configuration Files**	Basic Settings->Preferences-> Time and Date Settings Options->Preferences-> Time and Date->Time Servers 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	

Custom Time Zone and DST Settings

Note: To use the parameters in this section, the "**time zone name**" parameter must be set to "**Custom**". See page A-50 for more information.

Parameter – *time zone minutes*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	The number of minutes the timezone is offset from UTC (Coordinated Universal Time). This can be positive (West of the Prime Meridian) or negative (East of the Prim Meridian). Eastern Standard Time (EST) has a value of 300 which is the default..
Format	Integer
Default Value	300
Range	Any positive or negative Integers: 720 (GMT minus 12 hours) 660 (GMT minus 11 hours) 600 (GMT minus 10 hours) 540 (GMT minus 9 hours - Alaska Standard Time North America) 480 (GMT minus 8 hours - Pacific Standard Time North America) 420 (GMT minus 7 hours - Mountain Standard Time North America) 360 (GMT minus 6 hours - Central Standard Time North America) 300 (GMT minus 5 hours - Eastern Standard Time North America) 270 (GMT minus 4.5 hours - Venezuela) 240 (GMT minus 4 hours) 210 (GMT minus 3.5 hours - Newfoundland Standard Time North America) 180 (GMT minus 3 hours) 150 (GMT minus 2.5 hours - Newfoundland daylight time) 120 (GMT minus 2 hours) 60 (GMT minus 1 hour) 0 (GMT = 0 hours - Greenwich Mean Time) -60 (GMT + 1 hour - Central European Time) -120 (GMT + 2 hours - Eastern European Time Europe) -180 (GMT + 3 hours) -210 (GMT + 3.5 hours) -240 (GMT + 4 hours) -270 (GMT + 4.5 hours) -300 (GMT + 5 hours) -330 (GMT + 5.5 hours) -345 (GMT + 5.75 hours) -360 (GMT + 6 hours) -390 (GMT + 6.5 hours) -420 (GMT + 7 hours - Christmas Island Time Australia) -480 (GMT + 8 hours - Australian Western Standard Time) -540 (GMT + 9 hours) -570 (GMT + 9.5 hours - Australian Central Standard Time) -600 (GMT + 10 hours - Australian Eastern Standard Time) -660 (GMT + 11 hours) -720 (GMT + 12 hours) -780 (GMT + 13 hours)
Example	time zone minutes: 300

Parameter – *dst minutes*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	The number of minutes to add during Daylight Saving Time. Valid values are a positive integer between 0 to 60.
Format	Integer
Default Value	Not Applicable
Range	0-60
Example	dst minutes: 60

Parameter – *dst [start|end] relative date*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	Specifies how to interpret the start and end day, month, and week parameters - absolute (0) or relative (1).
Format	Boolean
Default Value	Not Applicable
Range	0 (absolute) 1 (relative)
Example	dst [start|end] relative date: 1

Absolute Time

Parameter – *dst start month*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	The month that DST starts. Valid values are 1 to 12 (January to December).
Format	Integer
Default Value	Not Applicable
Range	1 (January) 2 (February) 3 (March) 4 (April) 5 (May) 6 (June) 7 (July) 8 (August) 9 (September) 10 (October) 11 (November) 12 (December)
Example	dst start month: 7

Parameter – *dst end month*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	The month that DST ends. Valid values are 1 to 12 (January to December).
Format	Integer
Default Value	Not Applicable
Range	1 (January) 2 (February) 3 (March) 4 (April) 5 (May) 6 (June) 7 (July) 8 (August) 9 (September) 10 (October) 11 (November) 12 (December)
Example	dst end month: 6

Parameter – *dst start day*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	The day of the month that DST starts. Valid values are 1 to 31.
Format	Integer
Default Value	Not Applicable
Range	1-31
Example	dst start day: 1

Parameter – *dst end day*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	The day of the month that DST ends. Valid values are 1 to 31.
Format	Integer
Default Value	Not Applicable
Range	1-31
Example	dst end day: 31

Parameter – *dst start hour*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	The hour that DST starts. Valid values are an integer from 0 (midnight) to 23.
Format	Integer
Default Value	Not Applicable
Range	0 (midnight) to 23
Example	dst start hour: 0

Parameter – *dst end hour*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	The hour that DST ends. Valid values are an integer from 0 (midnight) to 23.
Format	Integer
Default Value	Not Applicable
Range	0 (midnight) to 23
Example	dst end hour: 23

Relative Time

Parameter – *dst start month*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	The month that DST starts. Valid values are 1 to 12 (January to December).
Format	Integer
Default Value	Not Applicable
Range	1 (January) 2 (February) 3 (March) 4 (April) 5 (May) 6 (June) 7 (July) 8 (August) 9 (September) 10 (October) 11 (November) 12 (December)
Example	dst start month: 6

Parameter – *dst end month*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	The month that DST ends. Valid values are 1 to 12 (January to December).
Format	Integer
Default Value	Not Applicable
Range	1 (January) 2 (February) 3 (March) 4 (April) 5 (May) 6 (June) 7 (July) 8 (August) 9 (September) 10 (October) 11 (November) 12 (December)
Example	dst end month: 12

Parameter – *dst start week*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	The week in the specified month in which DST starts. Valid value is a positive or negative integer from 1 to 5.
Format	Integer
Default Value	Not Applicable
Range	1 = first full week of month -1 = last occurance "dst start day" in the month 2 = second full week of month -2 =second last occurance "dst start day" in the month 3 = third full week of month -3 =third last occurance "dst start day" in the month 4 = fourth full week of month -4 = fourth last occurrance ""dst start day" in the month 5 = fifth full week of month -5 = fifth last occurance "dst start day" in the month
Example	dst start week: 1

Parameter – *dst end week*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	The week in the specified month in which DST ends. Valid value is a positive or negative integer from 1 to 5.
Format	Integer
Default Value	Not Applicable
Range	1 = first full week of month -1 = last occurance "dst start day" in the month 2 = second full week of month -2 =second last occurance "dst start day" in the month 3 = third full week of month -3 =third last occurance "dst start day" in the month 4 = fourth full week of month -4 = fourth last occurrance ""dst start day" in the month 5 = fifth full week of month -5 = fifth last occurance "dst start day" in the month
Example	dst end week: 5

Parameter – *dst start day*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	The day of the specified week in the specified month that DST starts on. Valid values are an integer from 1 to 7.
Format	Integer
Default Value	Not Applicable
Range	1 (Sunday) 2 (Monday) 3 (Tuesday) 4 (Wednesday) 5 (Thursday) 6 (Friday) 7 (Saturday)
Example	dst start day: 1

Parameter – *dst end day*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	The day of the specified week in the specified month that DST ends on. Valid values are an integer from 1 to 7.
Format	Integer
Default Value	Not Applicable
Range	1 (Sunday) 2 (Monday) 3 (Tuesday) 4 (Wednesday) 5 (Thursday) 6 (Friday) 7 (Saturday)
Example	dst end day: 7

Parameter – *dst start hour*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	The hour that DST starts. Valid values are an integer from 0 (midnight) to 23.
Format	Integer
Default Value	Not Applicable
Range	0 (midnight) to 23
Example	dst start hour: 0

Parameter – *dst end hour*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	The hour that DST ends. Valid values are an integer from 0 (midnight) to 23.
Format	Integer
Default Value	Not Applicable
Range	0 (midnight) to 23
Example	dst end hour: 23

Backlight Mode Settings (6755i, 6757i, 6757i CT)

Parameter – *backlight mode* *Backlight* (in Phone UI)	**IP phone UI** Options->Preferences->Display->Backlight **Configuration Files** aastra.cfg, <mac>.cfg
Description	Allows you to turn the backlight on the LCD, Off (always off) or Auto. The “Auto” setting sets the phone to turn off the backlight after a period of inactivity. You can set the amount of time before the backlight goes off using the “Backllight On Time” parameter.
Format	Integer
Default Value	1 (Auto)
Range	**0** (Off - Turns the backlight off constant) **1** (Auto - Turns the backlight off after a period of inactivity.) **Note:** In the IP Phone UI, the options for this parameter are “Off” and “Auto” only.
Example	backlight mode: 0

Parameter – *bl on time* *Backllight On Time* (in Phone UI)	**IP phone UI** Options->Preferences->Display->Backlight **Configuration Files** aastra.cfg, <mac>.cfg
Description	Allows you to set the amount of time, in seconds, that the backlight stays ON before turning OFF because of inactivity. This setting is applicable to the “**Auto**” mode only.
Format	Integer
Default Value	600 seconds (equals 10 minutes)
Range	1-7200 seconds
Example	bl on time: 15

Brightness Level Settings (6739i)

Parameter – *bl on time* *Brightness Timer* (in Phone UI)	**IP phone UI** **Configuration Files**	Options->Display->Brightness Timer aastra.cfg, <mac>.cfg
Description	Allows you to set the amount of time, in seconds, that the backlight stays ON before turning OFF because of inactivity. This setting is applicable to the "**Auto**" mode only.	
Format	Integer	
Default Value	600 seconds (equals 10 minutes)	
Range	1-7200 seconds	
Example	bl on time: 15	

Parameter – *brightness level* *Brightness Level* (in Phone UI)	**IP phone UI** **Configuration Files**	Options->Display->Brightness Level aastra.cfg, <mac>.cfg
Description	Specifies the brightness level during phone activity (e.g. when a user touches the touchscreen).	
Format	Integer	
Default Value	1	
Range	1-5	
Example	brightness level: 3	

Parameter – *inactivity brightness level*	**Configuration Files**	aastra.cfg, <mac>.cfg
Description	Specifies the brightness level during phone inactivity.	
Format	Integer	
Default Value	1	
Range	0...4 (where "0" represents screen off)	
Example	inactivity brightness level: 0	

DHSG Settings (6753i, 6755i, 6757i, 6757i CT)

Parameter – *dhsg*	**Configuration Files** aastra.cfg, <mac>.cfg **IP Phone UI:** Options->Preferences->Set Audio->DHSG
Description	Enables and disables the DHSG headset support on the phone. **Note:** The phones that support DHSG are: 6753i, 6755i, 6757i, and 6757i CT. For more information about installing a DHSG headset on your phone, see your *IP Phone-specific Installation Guide*.
Format	Boolean
Default Value	0
Range	0 (disable - DHSG support is OFF) 1 (enable - DHSG support is ON)
Example	dhsg: 1

Live Dialpad Settings

Parameter – *live dialpad* *Live Dialpad* (in Phone UI)	**IP phone UI** Options->Preferences->Live Dialpad **Configuration Files** aastra.cfg, <mac>.cfg
Description	Turns the “Live Dialpad” feature ON or OFF. With live dial pad ON, the IP phone automatically dials out and turns ON Handsfree mode as soon as a dial pad key or softkey is pressed. With live dial pad OFF, if you dial a number while the phone is on-hook, lifting the receiver or pressing the  initiates a call to that number.
Format	Boolean
Default Value	0 (Off)
Range	0 (Off) 1 (On)
Example	live dialpad: 1

SIP Local Dial Plan Settings

<table>
<tr><td>Parameter –
sip dial plan

Local Dial Plan
(in Web UI)</td><td>Aastra Web UI Basic Settings->Preferences
Configuration Files aastra.cfg, <mac>.cfg</td></tr>
<tr><td>Description</td><td>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:
<table>
<tr><td>Symbol</td><td>Description</td></tr>
<tr><td>0, 1, 2, 3, 4, 5, 6, 7, 8, 9</td><td>Digit symbol</td></tr>
<tr><td>X</td><td>Match any digit symbol (wildcard)</td></tr>
<tr><td>*, #, .</td><td>Other keypad symbol; # can terminate a dial string</td></tr>
<tr><td>|</td><td>Expression inclusive OR</td></tr>
<tr><td>+</td><td>0 or more of the preceding digit symbol or [] expression</td></tr>
<tr><td>[]</td><td>Symbol inclusive OR</td></tr>
<tr><td>-</td><td>Used only with [], represent a range of acceptable symbols; For example, [2-8]</td></tr>
<tr><td>“,” (open/close quotes)</td><td>In the configuration files, enter the sip dial plan value using quotes.</td></tr>
<tr><td>.;.</td><td>Used when a secondary dial tone is required on the phone. (For example, “9;xxxxxx”, when a user has to dial “9” to get and outside line and needs a secondary dial tone presented.</td></tr>
</table>

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”.</td></tr>
<tr><td>Format</td><td>Alphanumeric characters</td></tr>
<tr><td>Default Value</td><td>X+#|XX+*</td></tr>
<tr><td>Range</td><td>Up to 512 alphanumeric characters.

If a User enters a dial plan longer than 512 characters, or a parsing error occurs, the phone uses the default dial plan of “x+#|xx+*”. If this is the case, the Administrator cannot change the dial plan from the configuration files. The dial plan must be changed from the Aastra Web UI.</td></tr>
<tr><td>Example</td><td>sip dial plan: "X+#|XXX+*"</td></tr>
</table>

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 Outbound Support

Parameter – *sip outbound support*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	Enables and disables whether or not the phone uses Draft 15 (draft-ietf-sip-outbound-15) support for SIP outbound packets. A SIP User Agent (UA) behind a firewall, reuses an existing connection (usually the REGISTER outbound connection) for the inbound request if the proxy supports it. The UA uses keep-alive packets to monitor the connection status. **Notes**: **1.** You must restart the phone after setting a value for this parameter. **2.** TLS and TCP both support this feature. **3.** If the Global SIP parameter “Persistent TLS” is set on the phone, then only one TLS persistent connection can be established since the phone uses the local port 5061 for connection. If the Global SIP parameter “TLS” is set on the phone, more than one connection can be setup since the phone uses a random local port for connection. **4.** This parameter must be enabled for this feature to start keep-alive for a particular transport.
Format	Boolean
Default Value	0
Range	0 (disabled) 1 (enabled)
Example	sip outbound support: 1

Contact Header Matching

Parameter – *sip contact matching*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	Specifies the method for which the phone uses to match the Contact header in a SIP registration packet.
Format	Integer
Default Value	0
Range	0 (default) URI matching of username, domain name, port, and transport 1 matching of port only 2 matching of username only 3 matching of port and username only
Example	sip contact matching: 1

SIP Basic, Global Settings

SIP Global Authentication Settings

Parameter – *sip screen name* *Screen 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 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** **Aastra Web UI** **Configuration Files**	Options->Administrator Menu->SIP Settings Advanced Settings->Global SIP-> Basic SIP Settings 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** **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 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** **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 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** **Configuration Files**	Advanced Settings->Global SIP-> Basic SIP Settings 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** **Configuration Files**	Advanced Settings->Global SIP-> Basic SIP Settings 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) • **BLA** - Bridged Line Appearance (BLA) line.	
Format	Integer	
Default Value	0	
Range	Valid values are: 0 - Generic 1 - BroadSoft SCA 2 - (reserved) 3 - BLA	
Example	sip mode: 2	

Call Waiting Settings

Parameter – *call waiting* *Call Waiting* (in Web UI)	**Aastra Web UI** **Configuration Files**	Basic Settings->Preferences->General 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 6757i 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	

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

Parameter – *call waiting tone period* *Call Waiting Tone Period* (in Web UI)	**Aastra Web UI**: Basic Settings->Preferences **Configuration Files** aastra.cfg, <mac>.cfg
Description	Specifies the time period, in seconds, that the call waiting tone is audible on an active call when another call comes in. When enabled, the call waiting tone plays at regular intervals for the amount of time set for this parameter. For example, if set to “30” the call waiting tone plays every 30 seconds. When set to “0”, the call waiting tone is audible only once on the active call.
Format	Integer
Default Value	0
Range	0-4294967295
Example	call waiting tone period: 30

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** **Configuration Files**	Advanced Settings->Global SIP-> Advanced SIP Settings 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** **Aastra Web UI** **Configuration Files**	Options->Administrator Menu->SIP Settings Advanced Settings->Global SIP-> Basic SIP Settings 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	0-2147483647	
Example	sip registration period: 3600	

Backup Outbound Proxy (Global Settings)

Parameter – *sip backup outbound proxy*	**Aastra Web UI**	Advanced Settings->Global SIP-> Advanced SIP Settings
	Configuration Files	aastra.cfg, <mac>.cfg
Description	The IP address or domain name of the backup outbound SIP proxy server for which the IP phone uses when the primary SIP proxy is unavailable. Use this parameter to configure the sip backup outbound proxy on a global basis.	
Format	IP address or fully qualified Domain Name	
Default Value	0.0.0.0	
Range	Not Applicable	
Example	sip backup outbound proxy: drax.us.aastra.com	

Parameter – *sip backup outbound proxy port*	**Aastra Web UI**	Advanced Settings->Global SIP-> Advanced SIP Settings
	Configuration Files	aastra.cfg, <mac>.cfg
Description	The backup outbound proxy port on the backup outbound proxy server to which the IP phone sends all SIP messages. Use this parameter to configure the sip backup outbound proxy port on a global basis.	
Format	Integer	
Default Value	0	
Range	0 - 65535	
Example	sip backup outbound proxy port: 5060	

SIP Basic, Per-Line Settings

The following parameters are SIP per-line settings. The value of "*N*" is 1 - 9 or 1-6 depending on your model phone.

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-89. **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** Advanced Settings->Line 1 thru 9 **Configuration Files** aastra.cfg, <mac>.cfg
Description	Used in the username field of the Authorization header field of the *SIP REGISTER* request.
Format	Text
Default Value	Not Applicable
Range	Up to 20 alphanumeric characters
Example	sip line1 auth name: 5553456

Parameter – *sip lineN password* *Password* (in Web UI)	**Aastra Web UI** Advanced Settings->Line 1 thru 9 **Configuration Files** 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** Advanced Settings->Line 1 thru 9 **Configuration Files** aastra.cfg, <mac>.cfg
Description	Allows you to assign a phone number that is shared on specific lines on the IP phone. For Sylantro Server: When configuring the BLA feature on a Sylantro server, the value set for the **sip lineN bla number** parameter shall be the same value set for the **sip lineN user name** parameter for all the phones in the group. For example, if sip lineN user name is 1010, you would configure BLA on a per-line basis for the Sylantro server as follows: sip line1 user name: 1010(# for all the phones) sip line1 bla number: 1010 For ININ Server: When configuring the BLA feature on an ININ server, the value set for the **sip lineN bla number** parameter shall be the same value set for the **sip lineN user name** parameter without the incremented digit added to the phone #. For example, if the sip lineN user name for the first phone is 10101, and the sip lineN user name for the second phone is 10102, etc. you would configure BLA on a per-line basis for the ININ server as follows: sip line1 user name: 10101(# for phone 1 with) sip line1 bla number: 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

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) • **BLA** - Bridged Line Appearance (BLA) line. • If the softkeys on the 6757i/6757i CT or the programmable keys on the 6753i 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 - (Reserved) 3 - BLA	
Example	sip line1 mode: 2	

SIP Per-Line Call Waiting Setting

Parameter – *sip lineN call waiting* *Call Waiting* (in Web UI)	**Aastra Web UI** **Configuration Files**	Basic SIP Authentication Settings->Line 1 thru 9 aastra.cfg, <mac>.cfg
Description	Allows you to enable or disable Call Waiting on the IP phone on a per line basis. 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 6757i 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	Global	
Range	Global 0 (disabled) 1 (enabled)	
Example	sip line1 call waiting: 0 sip line2 call waiting: 1 sip line3 call waiting: 0	

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 lineN 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 lineN 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** **Configuration Files**	Advanced Settings->Line 1 thru 9 aastra.cfg, <mac>.cfg
Description	The registrar's port number	
Format	Integer	
Default Value	0	
Range	Not Applicable	
Example	sip line1 registrar port: 5060	

Parameter – *sip lineN backup registrar ip* *Backup Registrar Server* (in Web UI)	**Aastra Web UI** **Configuration Files**	Advanced Settings->Line*N*-> Basic SIP Network Settings 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 lineN 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	

Backup Outbound Proxy (Per-line Settings)

Parameter – *sip lineN backup outbound proxy*	**Aastra Web UI** **Configuration Files**	Advanced Settings->Global SIP-> Advanced SIP Settings aastra.cfg, <mac>.cfg
Description	The IP address or domain name of the backup outbound SIP proxy server for which the IP phone uses when the primary SIP proxy is unavailable. Use this parameter to configure the sip backup outbound proxy on a per-line basis.	
Format	IP address or fully qualified Domain Name	
Default Value	0.0.0.0	
Range	Not Applicable	
Example	sip line1 backup outbound proxy: drax.us.aastra.com	

Parameter – *sip lineN backup outbound proxy port*	**Aastra Web UI** **Configuration Files**	Advanced Settings->Global SIP-> Advanced SIP Settings aastra.cfg, <mac>.cfg
Description	The backup outbound proxy port on the backup outbound proxy server to which the IP phone sends all SIP messages. Use this parameter to configure the sip backup outbound proxy port on a per-line basis.	
Format	Integer	
Default Value	0	
Range	0 - 65535	
Example	sip line1 backup outbound proxy port: 5060	

BLA Support for MWI

Parameter – *sip mwi for bla account* *MWI for BLA Account* (in Web UI)	**Configuration Files** aastra.cfg, <mac>.cfg **Aastra Web UI** Advanced Settings->Global SIP->Advanced SIP Settings
Description	Enables or disables a BLA configured line to send an MWI SUBSCRIBE message for the BLA account. **Notes**: **1.** If you change the setting on this parameter, you must reboot the phone for it to take affect. **2.** Both the "**sip explicit mwi subscription**" and "**sip mwi for bla account**" parameters must be enabled in order for the MWI subscription for BLA to occur. **3.** The MWI re-subscription for the BLA account uses the value set for the "**sip explicit mwi subscription period**" parameter to re-subscribe. **4.** Whether or not the "**sip mwi for bla account**" parameter is enabled, the priority for displaying MWI does not change.
Format	Boolean
Default Value	0 (disabled)
Range	0 (disabled) 1 (enabled)
Example	sip mwi for bla account: 1

Shared Call Appearance (SCA) Call Bridging

Global Setting

Parameter – *sip sca bridging*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	Enables/disables SCA bridging on the phone-side on a global basis. **Note:** You must restart the phone after setting a value for this parameter.
Format	Boolean
Default Value	0
Range	0 (disabled) 1 (enabled)
Example	sip sca bridging: 1

Per-Line Setting

Parameter – *sip lineN sca bridging* (N is a line number from 1 to 9)	**Configuration Files** aastra.cfg, <mac>.cfg
Description	Enables/disables SCA bridging on the phone-side on a per-account basis using a specific SCA-configured line. **Note:** You must restart the phone after setting a value for this parameter.
Format	Boolean
Default Value	0
Range	0 (disabled) 1 (enabled)
Example	sip line1 sca bridging: 1

Centralized Conferencing Settings

Global Setting

Parameter – *sip centralized conf* *Conference Server URI* (in Web UI)	**Aastra Web UI** Advanced->Global SIP Settings-> Basic SIP Network Settings **Configuration Files** 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 Setting

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

SIP Join Feature for 3-Way Conference

Parameter – *sip join support*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	Enables or disables the phone to allow a conference to be set up with a join header as described in RFC 3911.
Format	Boolean
Default Value	0 (disabled)
Range	0 (disabled) 1 (enabled)
Example	sip join support: 1

HTTP/HTTPS Authentication Support for Broadsoft CMS

Parameter – *http digest username*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	Specifies the username to use for HTTP/HTTPS digest authentication. The server uses this username for authentication purposes when loading configuration to the phone over HTTP/HTTPS. This parameter initiates a “Username/Password” screen after pressing the **Log In** softkey. **Notes**: **1.** The Username field accepts special characters, such as, @, #, %, =, _, etc. You can also specify domain names (i.e., user@domain). **2.** You must reboot the phone after setting the HTTP/HTTPS digest authentication parameters.
Format	String
Default Value	Not Applicable
Range	Up to 40 alphanumeric characters
Example	http digest username: mysuername

Parameter – *http digest password*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	Specifies the password to use for HTTP/HTTPS digest authentication. The server uses this password for authentication purposes when loading configuration to the phone over HTTP/HTTPS. This parameter initiates a “Username/Password” screen after pressing the **Log In** softkey. **Notes**: **1.** The Password field accepts special characters, such as, @, #, %, =, _, etc. **2.** You must reboot the phone after setting the HTTP/HTTPS digest authentication parameters.
Format	String
Default Value	Not Applicable
Range	Up to 20 alphanumeric characters
Example	http digest password: mypassword

Parameter – *http digest force login*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	Enables and disables the display of a LOGIN key on the phone's idle screen. **Note:** After the server has authenticated the phone, this parameter must be set to "**0**" in order for the server to send the default profile to the phone.
Format	Boolean
Default Value	0 (disable)
Range	0 (disable) 1 (enable)
Example	http digest force login: 1

Advanced SIP Settings

Parameter – *sip explicit mwi subscription* *Explicit MWI Subscription* (in Web UI)	**Aastra Web UI** Advanced Settings->Global SIP-> Advanced SIP Settings **Configuration Files** 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 subscription period: 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:** Advanced Settings->Global SIP->Advanced SIP Settings **Configuration Files** 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 line number that is being registered.
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** Advanced Settings->Global SIP->Advanced SIP Settings **Configuration Files** 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. **3.** This parameter implies keep-alive mechanism. For more information about Persistent TLS, see "Transport Layer Security (TLS) Settings" on page A-114.	
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	30-1800	
Example	sip registration retry timer: 30	

Parameter – *sip registration timeout retry timer* *Registration Timeout Retry Timer* (in Web UI)	**Aastra Web UI** Advanced Settings->Global SIP->Advanced SIP Settings **Configuration Files** 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	30-214748364
Example	sip registration timeout retry timer: 150

Parameter – *sip registration renewal timer* Registration Renewal Timer (in Web UI)	**Aastra Web UI** Advanced Settings->Global SIP->Advanced SIP Settings **Configuration Files** 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	0-214748364 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	

Parameter– *sip bla subscription period* *BLA Subscription Period* (in Web UI)	**Configuration Files** **Aastra Web UI**	aastra.cfg, <mac>.cfg Advanced Settings->Global SIP-> Advanced SIP Settings
Description	Specifies the amount of time, in seconds, that the phone waits to receive a BLA subscribe message from the server. If you specify zero (0), the phone uses the value specified for the BLA expiration in the subscribe message received from the server. If no value is specified, the phone uses the default value of 300 seconds.	
Format	Integer	
Default Value	300	
Range	0-3700 **Note:** When set to zero (0), the phone uses BLA expiry value specified in subscribe message.	
Example	sip bla subscription period: 0	

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	0-99999999	
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** Advanced Settings->Global SIP-> Advanced SIP Settings **Configuration Files** 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

As-Feature-Event Subscription Settings

Parameter – *sip lineN as-feature-event subscription*	**Aastra Web UI:** Advanced Settings->LineN-> Advanced SIP Settings **Configuration Files** aastra.cfg, <mac>.cfg
Description	Enables or disables the specified line with the BroadSoft's server-side DND, CFWD, or ACD features.
Format	Boolean
Default Value	0 (disable)
Range	0 (disable) 1 (enable)
Example	sip line1 as-feature-event subscription: 1

Parameter – *sip as-feature-event subscription period*	**Aastra Web UI:** Basic Settings->Global SIP-> Advanced SIP Settings **Configuration Files** aastra.cfg, <mac>.cfg
Description	Specifies the amount of time, in seconds, between resubscribing. If the phone does not resubscribe in the time specified for this parameter, it loses subscription.
Format	Integer
Default Value	3600
Range	5-2147483648
Example	sip as-feature-event subscription period: 600

Transport Layer Security (TLS) Settings

To configure TLS, you must enter the "**sip transport protocol**" parameter with a value of "**4**" (TLS). See the "sip transport protocol" description on page A-107.

Also enter the following parameters in the configuration files to configure TLS:

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

<table>
<tr><td>Parameter –
sips root and intermediate certificates

Root and Intermediate Certificates Filename
(in Web UI)</td><td>Aastra Web UI
Configuration Files</td><td>Advanced Settings->TLS Support
aastra.cfg, <mac>.cfg</td></tr>
<tr><td>Description</td><td colspan="2">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.)

You can use this parameter in three ways:
• To download no certificates
• To download a certificate from the original configuration server
• To download a certificate from another specified server

To download a specific file, the string value MUST HAVE A FILENAME at the end of the string. For example:

sips root and intermediate certificates:ftp://admin:admin!@1.2.3.4:50/path/phonesRootCert.pem

where “path” is the directory and “phonesRootCert.pem” is the filename. If you do not specify a filename, the download fails.

See examples for each below.

Note: The certificate files must use the format “.pem”. To create custom certificate files to use on your IP phone, contact Aastra Technical Support.</td></tr>
<tr><td>Format</td><td colspan="2"><file name>.pem</td></tr>
<tr><td>Default Value</td><td colspan="2">Not Applicable</td></tr>
<tr><td>Range</td><td colspan="2">Not Applicable</td></tr>
</table>

Example	The following example downloads no root and intermediate certificate file: sips root and intermediate certificates: The following example downloads the root and intermediate certificate file from the original configuration server. sips root and intermediate certificates: phonesRootCert.pem The following example uses FTP to download the firmware file "phonesRootCert.pem" (root and intermediate certificate file) from the "path" directory on server 1.2.3.4 using port 50: sips root and intermediate certificates:ftp://admin:admin!@1.2.3.4:50/path/phonesRootCert.pem

<table>
<tr><td>Parameter –
sips local certificate

Local Certificate Filename
(in Web UI)</td><td>Aastra Web UI
Configuration Files</td><td>Advanced Settings->TLS Support
aastra.cfg, <mac>.cfg</td></tr>
<tr><td>Description</td><td colspan="2">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.)

You can use this parameter in three ways:
• To download no certificates
• To download a certificate from the original configuration server
• To download a certificate from another specified server

To download a specific file, the string value MUST HAVE A FILENAME at the end of the string. For example:

sips local certificate:ftp://admin:admin!@1.2.3.4:50/path/ phonesLocalCert.pem

where “path” is the directory and “phonesLocalCert.pem” is the filename. If you do not specify a filename, the download fails.

See examples for each below.

Note: The certificate file must use the format “.pem”. To create specific certificate files to use on your IP phone, contact Aastra Technical Support.</td></tr>
<tr><td>Format</td><td colspan="2"><file name>.pem</td></tr>
<tr><td>Default Value</td><td colspan="2">Not Applicable</td></tr>
<tr><td>Range</td><td colspan="2">Not Applicable</td></tr>
<tr><td>Example</td><td colspan="2">The following example downloads no local certificate file:

sips local certificate:

The following example downloads the local certificate file from the original configuration server.

sips local certificate: phonesLocalCert.pem

The following example uses FTP to download the firmware file “phonesLocalCert.pem” (local certificate file) from the “path” directory on server 1.2.3.4 using port 50:

sips local certificate:ftp://admin:admin!@1.2.3.4:50/path/ phonesLocalCert.pem</td></tr>
</table>

<table>
<tr><td>Parameter –
sips private key

Private Key Filename
(in Web UI)</td><td>Aastra Web UI
Configuration Files</td><td>Advanced Settings->TLS Support
aastra.cfg, <mac>.cfg</td></tr>
<tr><td>Description</td><td colspan="2">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.)

You can use this parameter in three ways:
• To download no private key
• To download a private key from the original configuration server
• To download a private key from another specified server

To download a specific file, the string value MUST HAVE A FILENAME at the end of the string. For example:

sips trusted certificates:ftp://admin:admin!@1.2.3.4:50/path/
phonesPrivatekey.pem

where “path” is the directory and “phonesPrivateKey.pem” is the filename. If you do not specify a filename, the download fails.

See examples for each below.

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.</td></tr>
<tr><td>Format</td><td colspan="2"><file name>.pem</td></tr>
<tr><td>Default Value</td><td colspan="2">Not Applicable</td></tr>
<tr><td>Range</td><td colspan="2">Not Applicable</td></tr>
<tr><td>Example</td><td colspan="2">The following example downloads no private key file:

sips private key:

The following example downloads the private key file from the original configuration server.

sips private key: phonesPrivateKey.pem

The following example uses FTP to download the firmware file “phonesPrivateKey.pem” (private key file) from the “path” directory on server 1.2.3.4 using port 50:

sips private key: ftp://admin:admin!@1.2.3.4:50/path/
phonesPrivateKey.pem</td></tr>
</table>

Parameter – *sips trusted certificates* *Trusted Certificates Filename* (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. You can use this parameter in three ways: • To download no certificates • To download a certificate from the original configuration server • To download a certificate from another specified server To download a specific file, the string value MUST HAVE A FILENAME at the end of the string. For example: sips trusted certificates:ftp://admin:admin!@1.2.3.4:50/path/phonesTrustedCert.pem where "path" is the directory and "phonesTrustedCert.pem" is the filename. If you do not specify a filename, the download fails. See examples for each below. **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	Not Applicable	
Range	Not Applicable	

Example	The following example downloads no trusted certificate file: sips trusted certificates: The following example downloads the trusted certificate file from the original configuration server. sips trusted certificates: phonesTrustedCert.pem The following example uses FTP to download the firmware file “phonesTrustedCert.pem” (trusted certificate file) from the “path” directory on server 1.2.3.4 using port 50: sips trusted certificates:ftp://admin:admin!@1.2.3.4:50/path/ phonesTrustedCert.pem

802.1x Support Settings

Use the following parameters to configure the 802.1x Protocol on your phone using the configuration files.

For EAP-MD5 use:

- **eap type**
- **identity**
- **md5 password**
- **pc port passthrough enabled**

For EAP-TLS use:

- **eap type**
- **identity**
- **802.1x root and intermediate certificates**
  (use 1 root and 0 or 1 intermediate certificates)
- **802.1x local certificate**
  (use 1 local certificate)
- **802.1x private key**
  (1 private key that corresponds to local certificate)
- **802.1x trusted certificates**
  (0 or more trusted certificates (a maximum of 2))
- **pc port passthrough enabled**

Parameter – *pc port passthru enabled* *PC Port PassThru Enable/ Disable* (in Web UI)	**IP phone UI** **Aastra Web UI** **Configuration Files**	Options->Administrator Menu-> Network Settings->Ethernet Link Advanced Settings->Network-> Basic Network Settings aastra.cfg, <mac>.cfg
Description	Enables or disables the PC port.	
Format	Integer	
Default Value	1 (enable)	
Range	0 (disable) 1 (enable)	
Example	pc port passthru enabled: 1	

Parameter – *eap type* *EAP Type* (in Web UI)	**IP Phone UI:** **Aastra Web UI:** **Configuration Files**	Options->Administrator Menu-> Network Settings->Ethernet Link-> 802.1x Settings->802.1x Mode Advanced Settings->802.1x Support->General aastra.cfg, <mac>.cfg
Description	Specifies the type of authentication to use on the IP Phone.	
Format	Integer	
Default Value	0 (disable)	
Range	0 (disable) 1 (MD5) 2 (TLS)	
Example	eap type: 1	

Parameter – *identity* *Identity* (in Web UI)	**IP Phone UI:** **Aastra Web UI:** **Configuration Files**	Options->Administrator Menu-> Network Settings->Ethernet Link-> 802.1x Settings->EAP-MD5 Settings Advanced Settings->802.1x Support->General aastra.cfg, <mac>.cfg
Description	Specifies the identity or username used for authenticating the phone. **Note:** The value you enter for this parameter also displays in the Aastra Web UI at the path Advanced Settings-> 802.1x Support->General->Identity.	
Format	String	
Default Value	Not Applicable	
Range	Not Applicable	
Example	identity: phone1	

Parameter – *md5 password* *MD5 Password* (in Web UI)	**IP Phone UI:** **Aastra Web UI:** **Configuration Files**	Options->Administrator Menu-> Network Settings->Ethernet Link-> 802.1x Settings->EAP-MD5 Settings Advanced Settings->802.1x Support-> EAP-MD5 Settings aastra.cfg, <mac>.cfg
Description	Specifies the password used for the MD5 authentication of the phone. **Note:** The value you enter for this parameter also displays in the Aastra Web UI at the path Advanced Settings-> 802.1x Support->EAP-MD5 Settings->MD5 Password. The password displays as "*******".	
Format	String	
Default Value	Not Applicable	
Range	Not Applicable	
Example	md5 password: password1	

Parameter – *802.1x root and intermediate certificates* *Root and Intermediate Certificates* (in Web UI)	**Aastra Web UI:** Advanced Settings->802.1x Support-> EAP-TLS Settings **Configuration Files** aastra.cfg, <mac>.cfg
Description	Specifies the file name that contains the root and intermediate certificates related to the local certificate. You can use this parameter in three ways: • To download no certificates • To download a certificate from the original configuration server • To download a certificate from another specified server To download a specific file, the string value MUST HAVE A FILENAME at the end of the string. For example: 802.1x root and intermediate certificates:ftp://admin:admin!@1.2.3.4:50/path/phones802RootCert.pem where "path" is the directory and "phones802RootCert.pem" is the filename. If you do not specify a filename, the download fails. See examples for each below.
Format	String
Default Value	Not Applicable
Range	Not Applicable
Example	The following example downloads no 802.1x root and intermediate certificate file: 802.1x root and intermediate certificates: The following example downloads the 802.1x root and intermediate certificate file from the original configuration server. 802.1x root and intermediate certificates: phones802RootCert.pem The following example uses FTP to download the firmware file "phones802RootCert.pem" (802.1x root and intermediate certificate file) from the "path" directory on server 1.2.3.4 using port 50: 802.1x root and intermediate certificates:ftp://admin:admin!@1.2.3.4:50/path/phones802RootCert.pem

Parameter – *802.1x local certificate* *Local Certificate* (in Web UI)	**Aastra Web UI:** **Configuration Files**	Advanced Settings->802.1x Support->EAP-TLS Settings aastra.cfg, <mac>.cfg
Description	Specifies the file name that contains the local certificate. You can use this parameter in three ways: • To download no certificates • To download a certificate from the original configuration server • To download a certificate from another specified server To download a specific file, the string value MUST HAVE A FILENAME at the end of the string. For example: 802.1x local certificate:ftp://admin:admin!@1.2.3.4:50/path/phones802LocalCert.pem where "path" is the directory and "phones802LocalCert.pem" is the filename. If you do not specify a filename, the download fails. See examples for each below.	
Format	String	
Default Value	Not Applicable	
Range	Not Applicable	
Example	The following example downloads no local certificate file: 802.1x local certificate: The following example downloads the local certificate file from the original configuration server. 802.1x local certificate: phones802LocalCert.pem The following example uses FTP to download the firmware file "phones802LocalCert.pem" (802.1x local certificate file) from the "path" directory on server 1.2.3.4 using port 50: 802.1x local certificate:ftp://admin:admin!@1.2.3.4:50/path/phones802LocalCert.pem	

Parameter – *802.1x private key* *Private Key* (in Web UI)	**Aastra Web UI:** **Configuration Files**	Advanced Settings->802.1x Support-> EAP-TLS Settings aastra.cfg, <mac>.cfg
Description	Specifies the file name that contains the private key.	
Format	String	
Default Value	Not Applicable	
Range	Not Applicable	
Example	802.1x private key: filename.pem	

<table>
<tr><td>Parameter –
802.1x trusted certificates

Trusted Certificates
(in Web UI)</td><td>Aastra Web UI: Advanced Settings->802.1x Support-> EAP-TLS Settings
Configuration Files aastra.cfg, <mac>.cfg</td></tr>
<tr><td>Description</td><td>Specifies the file name that contains the trusted certificates.

You can use this parameter in three ways:
• To download no certificates
• To download a certificate from the original configuration server
• To download a certificate from another specified server

To download a specific file, the string value MUST HAVE A FILENAME at the end of the string. For example:

802.1x trusted certificates:ftp://admin:admin!@1.2.3.4:50/path/phones802TrustedCert.pem

where “path” is the directory and “phones802TrustedCert.pem” is the filename. If you do not specify a filename, the download fails.

See examples for each below.</td></tr>
<tr><td>Format</td><td>String</td></tr>
<tr><td>Default Value</td><td>Not Applicable</td></tr>
<tr><td>Range</td><td>Not Applicable</td></tr>
<tr><td>Example</td><td>The following example downloads no 802.1x trusted certificate file:

802.1x trusted certificates:

The following example downloads the 802.1x trusted certificate file from the original configuration server.

802.1x trusted certificates: phones802TrustedCert.pem

The following example uses FTP to download the firmware file “phones802TrustedCert.pem” (802.1x trusted certificate file) from the “path” directory on server 1.2.3.4 using port 50:

802.1x trusted certificates:ftp://admin:admin!@1.2.3.4:50/path/phones802TrustedCert.pem</td></tr>
</table>

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. **Note:** The phones support decoding and playing out DTMF tones sent in SIP INFO requests. The following DTMF tones are supported: • Support signals 0-9, #, * • Support durations up to 5 seconds	
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 (G.711 u-Law, G.711 a-Law, G.729). 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** **Configuration Files**	Advanced Settings->Global SIP->RTP Settings 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* *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 select the preferred Codecs for this IP phone. You can enter up to 10 codec preferences. **Note:** Enabling or disabling silence suppression (**sil supp**) enables/ disables it for all codecs in the customized list.
Format	Comma-separated list of semicolon-separated values
Default Value	Not Applicable
Range	Valid values for the syntax are: **payload**: **Configuration Files** / **Web UI** 0 - G711u/8000 — G711u (8K) 8 - G711a/8000 — G711a (8K) 98 - G726-16/8000 — G726-16 97 - G726-24/8000 — G726-24 115 - G726-32/8000 — G726-32 96 - G726-40/8000 — G726-40 18 - G729/8000 — G729 106 - BV16/8000 — BV16 (8K) 107 - BV32/16000 — BV32 (16K) 110 - G711u/16000 — G711u (16K) 111 - G711a/16000 — G711a (16K) 9 - G722/8000 — G722 113 - L16/16000 — L16 (16K) 112 - L16/8000 — L16 (8K) Leave blank for all codecs — All (Codec 1 only) Basic (Codec 1 only) None (Codecs 2 thru 10 only) **ptime** (in milliseconds) — 5, 10, 15, 20........90 **silsupp** — on, off **Note:** The “silsupp” value is either ON for all codecs or OFF for all codecs.
Example	sip customized codec: payload=18;ptime=10;silsupp=on,payload=0;ptime=10; silsupp=on

Parameter – *sip dtmf method* *DTMF Method* (in Web UI)	**Aastra Web UI** Advanced Settings->Global SIP->RTP Settings **Configuration Files** aastra.cfg, <mac>.cfg
Description	Sets the Dual-tone multifrequency (DTMF) method to use on the IP phone.
Format	Boolean
Default Value	0 (RTP)
Range	0 (RTP) 1 (SIP INFO) 2 (BOTH)
Example	sip dtmf method: 1

Parameter – *sip srtp mode* *RTP Encryption* (in Web UI)	**Aastra Web UI** Advanced Settings->Global SIP-> RTP Settings **Configuration Files** 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** Advanced Settings->Global SIP->RTP Settings **Configuration Files** aastra.cfg, <mac>.cfg
Description	Silence suppression is enabled by default on the IP phones. The phone negotiates whether or not to use silence suppression. Disabling this feature forces the phone to ignore any negotiated value.
Format	Boolean
Default Value	1 (enabled)
Range	0 (disabled) 1 (enabled)
Example	sip silence suppression: 0

Per-Line Settings

Parameter – *sip lineN dtmf method* *DTMF Method* (in Web UI)	**Aastra Web UI** Advanced Settings->Global SIP->RTP Settings **Configuration Files** aastra.cfg, <mac>.cfg
Description	Sets the Dual-tone multifrequency (DTMF) method to use on the IP phone for a specific line.
Format	Integer
Default Value	0 (RTP)
Range	0 (RTP) 1 (SIP INFO) 2 (BOTH)
Example	sip line1 dtmf method: 1

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

Parameter – *sip autodial timeout* *Autodial Timeout* (in Web UI)	**Aastra Web UI** Advanced Settings->Global SIP->Autodial Settings **Configuration Files** 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	0-120
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	0-120
Examples	sip line1 autodial timeout: 30

Voicemail Settings

Parameter – *sip lineN vmail* **Note:** The value of "*N*" is 1 - 9 or 1-6 depending your model phone.	**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	String
Default Value	Not Applicable
Range	0-99
Example	sip line1 vmail: *97 **Note:** In the above example, the user would dial *97 to access the voicemail account.

Parameter – *sip vmail*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	Specifies the phone number of the voicemail system connected to the sip account. This parameter specifies the phone number you dial from your phone to retrieve your voicemail. Configuring this parameter allows you to call the voice mail system directly from the "voicemail" application via the IP Phone UI under the "Services" menu of the IP Phone.
Format	Integer
Default Value	Not Applicable
Range	Not Applicable
Example	sip vmail: 5553435

Directory Settings

Parameter – *directory 1* *Directory List* (in Web UI)	**Aastra Web UI** Operation->Directory **Configuration Files** aastra.cfg, <mac>.cfg
Description	The name of a directory list that you can download from the configuration server. **Note:** You can use this parameter in three ways: • To download no directory • To download a directory from the original configuration server • To download a directory from another specified server To download a specific file, the string value MUST HAVE A FILENAME at the end of the string. For example: directory 1: tftp://10.30.102.158/path/companylist.csv where “path” is the directory and “companylist.csv” is the filename. If you do not specify a filename, the download fails. See examples for each below.
Format	Alphanumeric characters
Default Value	Not Applicable
Range	Not Applicable
Example	The following example downloads no directory: directory 1: The following example downloads a company directory from the original configuration server: directory 1:companylist.csv The following example downloads a company directory file from the specified server in the “path” directory: directory 1: tftp://10.30.102.158/path/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. **Note:** You can use this parameter in three ways: • To download no directory • To download a directory from the original configuration server • To download a directory from another specified server To download a specific file, the string value MUST HAVE A FILENAME at the end of the string. For example: directory 1: tftp://10.30.102.158/path/companylist.csv where "path" is the directory and "companylist.csv" is the filename. If you do not specify a filename, the download fails. See examples for each below.	
Format	Alphanumeric characters	
Default Value	Not Applicable	
Range	Not Applicable	
Example	The following example downloads no directory: directory 2: The following example downloads a company directory from the original configuration server: directory 2:companylist.csv The following example downloads a company directory file from the specified server in the "path" directory: directory 2: tftp://10.30.102.158/path/companylist.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 6757i and 6757i 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 6757i and 6757i 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	Not Applicable
Range	Not Applicable
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	Not Applicable
Range	Not Applicable
Example	callers list script: http://10.50.100.234/test.xml

Call Forward Settings

Parameter – *call forward disabled*	**IP Phone UI** **Aastra Web UI** **Configuration Files**	Options->Call Forward Basic Settings->Account Configuration 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	

Call Forward Key Mode Settings

Parameter – *call forward key mode*	**Aastra Web UI:** Basic Settings->Preferences->General **Configuration Files** aastra.cfg, <mac>.cfg
Description	Sets the mode for how the phone uses "call forwarding" (CFWD) • **account** The account mode allows you to configure CFWD on a per account basis. Pressing a configured CFWD key applies to the account in focus. • **phone** The Phone mode allows you to set the same CFWD configuration for all accounts (**All**, **Busy**, and/or **No Answer**). When you configure the initial account, the phone applies the configuration to all other accounts. (In the Aastra Web UI, only the account you configured is enabled. All other accounts are grayed out but set to the same configuration.) Using the Aastra Web UI, if you make changes to that initial account, the changes apply to all accounts on the phone. • **custom** The Custom mode allows you to configure CFWD for a specific account or all accounts. You can configure a specific mode (**All**, **Busy**, and/or **No Answer**) for each account independently or all accounts. On 3-Line LCD phones, you can set all accounts to **ALL On** or **ALL Off**. On 8 and 11-Line LCD phones, you can set all accounts to **All On**, **All Off**, or copy the configuration for the account in focus to all other accounts using a **CopytoAll** softkey. **Notes**: **1.** If there is no CFWD key configured on the phone or it is removed, you can still set the CFWD modes via the IP Phone UI at the path *Options->Call Forward.* **2.** If there is only one account configured on the phone, then the mode setting is ignored and the phone behaves as if the mode was set to "Phone". **3.** When configuring a CFWD mode (**All**, **Busy**, **No Answer**) for an account, you must configure a CFWD number for that mode in order for the mode to be enabled.
Format	Integer
Default Value	0 (account)
Range	0 (account) 1 (phone) 2 (custom)
Example	call forward key mode: 2

Example

The following is an example of configuring the CFWD key mode in the configuration files:

```
call forward key mode: 2
softkey1 type: callforward
softkey1 states: idle connected incoming outgoing busy
```

In the above example, softkey 1 is configured for CFWD on line 1 (account 1) with a "**custom**" configuration. Pressing softkey 1 displays CFWD screens for which you can customize on the phone.

LLDP-MED and ELIN Settings

Parameter– *lldp* *LLDP Support* (IP Phone UI) *LLDP* (in Web UI)	**Configuration Files** aastra.cfg, <mac>.cfg **IP Phone UI** Options->Administrator Menu-> Network Settings->Ethernet&VLAN-> LLDP Support **Aastra Web UI** Advanced Settings->Network-> Advanced Network Settings
Description	Enables or disables Link Layer Discovery Protocol for Media Endpoint Devices (LLDP-MED) on the IP Phone.
Format	Boolean
Default Value	1 (enabled)
Range	0 (disabled) 1 (enabled)
Example	lldp: 0

Parameter– *lldp interval* *LLDP Packet Interval* (in Web UI)	**Configuration Files** aastra.cfg, <mac>.cfg **Aastra Web UI** Advanced Settings->Network-> Advanced Network Settings
Description	The amount of time, in seconds, between the transmission of LLDP Data Unit (LLDPDU) packets. The value of zero (0) disables this parameter.
Format	Integer
Default Value	30
Range	0-2147483647
Example	lldp interval: 60

Parameter– *use lldp elin* *Use LLDP ELIN* (in Web UI)	**Configuration Files** **Aastra Web UI**	aastra.cfg, <mac>.cfg Basic Settings->Preferences->General
Description	Enables or disables the use of an Emergency Location Identification Number (ELIN) received from LLDP as a caller ID for emergency numbers.	
Format	Boolean	
Default Value	1 (enabled)	
Range	0 (disabled) 1 (enabled)	
Example	use lldp elin: 0	

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

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** **Configuration Files**	Operation->Softkeys and XML->Services 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** **Configuration Files**	Basic Settings->Preference**s** 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** Advanced Settings->Action URI **Configuration Files** aastra.cfg, <mac>.cfg
Description	Specifies the URI for which the phone executes a GET on when a startup event occurs. This parameter can use any of the following variables: $$REMOTENUMBER$$ $$DISPLAYNAME$$ $$SIPUSERNAME$$ $$SIPAUTHNAME$$ $$INCOMINGNAME$ $$PROXYURL$$ $$LINESTATE$$ $$LOCALIP$$ $$DISPLAYNAME$$ $$CALLDURATION$$ $$CALLDIRECTION$$
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** Advanced Settings->Action URI **Configuration Files** 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: $$SIPUSERNAME$$ $$SIPAUTHNAME$$ $$PROXYURL$$ $$LINESTATE$$ $$LOCALIP$$ **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 registration event* *Registration Event* (in Web UI)	**Configuration Files** **Aastra Web UI**	aastra.cfg, <mac>.cfg Advanced Settings->Action URI
Description	Specifies the URI that the phone executes a GET on, when a registration event change occurs. This parameter uses the following variables to determine the state of the event: $$REGISTRATIONSTATE$$ $$REGISTRATIONCODE$$ **Note:** This action URI is not called when the same event is repeated (for example, a timeout occurs again when registration is already in a timeout state.)	
Format	String	
Default Value	Not Applicable	
Range	Any valid URI	
Example	action uri registration event: http://10.30.100.39/PHPtests/ actionuri.php?action=RegEvt&regstate=$$REGISTRATIONSTATE$$&regco de=$$REGISTRATIONCODE$$	

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: $$REMOTENUMBER$$ $$DISPLAYNAME$$ $$SIPUSERNAME$$ $$INCOMINGNAME$ $$LINESTATE$$ $$CALLDURATION$$ $$CALLDIRECTION$$ $$LOCALIP$$	
Format	Fully qualified URI	
Default Value	Not Applicable	
Range	Up to 128 ASCII characters	
Example	action uri incoming: http://10.50.10.140/ incoming.php?number=$$REMOTENUMBER$$	

Parameter – *action uri outgoing* *Outgoing Call* (in Web UI)	**Aastra Web UI** Advanced Settings->Action URI **Configuration Files** 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: $$REMOTENUMBER$$ $$SIPUSERNAME$$ $$LINESTATE$$ $$LOCALIP$$
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** Advanced Settings->Action URI **Configuration Files** aastra.cfg, <mac>.cfg
Description	Specifies the URI for which the phone executes a GET on when an offhook event occurs. This parameter can use the following variables: $$LINESTATE$$ $$LOCALIP$$
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** Advanced Settings->Action URI **Configuration Files** aastra.cfg, <mac>.cfg
Description	Specifies the URI for which the phone executes a GET on when an onhook event occurs. This parameter can use the following variables: $$LOCALIP$$ $$CALLDURATION$$ $$CALLDIRECTION$$ $$LINESTATE$$ **Note:** The “LocalIP”, “CallDuration”, and “CallDirection” variables allow for enhanced information in call records and billing applications.
Format	Fully qualified URI
Default Value	Not Applicable
Range	Up to 128 ASCII characters
Example	action uri onhook: http://10.50.10.140/onhook

Parameter– *action uri disconnected*	**Aastra Web UI** Advanced Settings->Action URI->Event **Configuration Files** aastra.cfg, <mac>.cfg
Description	Specifies the URI that the phone executes a GET on, when it transitions from the incoming, outgoing, calling, or connected state into the idle state. This parameter uses the following variables to determine the state of the line: $$LINESTATE$$ $$LOCALIP$$
Format	String
Default Value	Not Applicable
Range	Any valid URI
Example	action uri disconnected: http://fargo.ana.aastra.com/ disconnected.xml?state=$$LINESTATE$$

XML SIP Notify Settings

Parameter – *sip xml notify event*	**Aastra Web UI:** **Configuration Files**	Advanced Settings->Global SIP-> Advanced SIP Settings aastra.cfg, <mac>.cfg
Description	Enables or disables the phone to accept or reject an aastra-xml SIP NOTIFY message. **Note:** To ensure the SIP NOTIFY is coming from a trusted source, it is recommended that you enable the Whitelist feature (**Whitelist Proxy** parameter) on the IP phone. If enabled, and the phone receives a SIP NOTIFY from a server that is NOT on the whitelist (i.e. untrusted server), the phone rejects the message.	
Format	Boolean	
Default Value	0	
Range	0 - disabled 1 - enabled	
Example	sip xml notify event: 1	

Parameter – *action uri xml sip notify*	**Aastra Web UI:** **Configuration Files**	Advanced Settings->Action URI aastra.cfg, <mac>.cfg
Description	Specifies the URI to be called when an empty XML SIP NOTIFY is received by the phone. This parameter can use the following variable: $$LOCALIP$$ **Note:** The **sip xml notify event** parameter must be enabled.	
Format	HTTP(s) server path or Fully Qualified Domain Name	
Default Value	Not Applicable	
Range	Not Applicable	
Example	action uri xml sip notify: http://myserver.com/myappli.xml	

Polling Action URI Settings

Parameter – *action uri poll*	**Aastra Web UI** Advanced Settings->Action URI **Configuration Files** aastra.cfg, <mac>.cfg
Description	Specifies the URI to be called every "action uri poll interval" seconds.
Format	HTTP(s) server path or Fully Qualified Domain Name
Default Value	Not Applicable
Range	Not Applicable
Example	action uri poll: http://myserver.com/myappli.xml

Parameter – *action uri poll interval*	**Aastra Web UI** Advanced Settings->Action URI **Configuration Files** aastra.cfg, <mac>.cfg
Description	Specifies the interval, in seconds, between calls from the phone to the "action uri poll".
Format	Integer
Default Value	0 (disabled)
Range	Not Applicable
Example	action uri poll interval: 60

Ring Tone and Tone Set Global Settings

Parameter – *ring tone* *Global Ring Tone* (in Web UI)	**IP Phone UI** Options->Preferences->Tones->Set Ring Tone **6739i Phone** Options->Audio->Ring Tone **Aastra Web UI:** Basic Settings->Preferences->Ring Tones **Configuration Files** aastra.cfg, <mac>.cfg
Description	Globally sets the type of ring tone on the IP phone. Ring tone can be set to one of six distinct rings.
Format	Integer
Default Value	Aastra Web UI: Tone 1 IP Phone UI: Tone 1 Configuration Files: 0 (Tone 1)
Range	**Aastra Web UI & IP Phone UI** Tone 1 Tone 2 Tone 3 Tone 4 Tone 5 Silent **Configuration Files** 0 (Tone 1) 1 (Tone 2) 2 (Tone 3) 3 (Tone 4) 4 (Tone 5) 5 (Silent)
Example	ring tone: 3

Parameter – *tone set* *Tone Set* (in Web UI)	**IP Phone UI** Options->Preferences->Tones->Tone Set **6739i Phone** Options->Audio->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 Brazil Russia Malaysia UK (United Kingdom) US (also used in Canada)
Example	tone set: Germany

Ring Tone Per-Line Settings

Parameter – *lineN ring tone* *N=1 through 9* *Line N* (in Web UI)	**Aastra Web UI:** Basic Settings->Preferences->Ring Tones **Configuration Files** aastra.cfg, <mac>.cfg
Description	Sets the type of ring tone on the IP phone on a per-line basis. Ring tone can be set to one of six distinct rings. **Note:** LineN is applicable to lines 1 through 9.
Format	Integer
Default Value	Aastra Web UI: Global Configuration Files: -1 (Global)
Range	Aastra Web UI Global Tone 1 Tone 2 Tone 3 Tone 4 Tone 5 Silent Configuration Files -1 (Global) 0 (Tone 1) 1 (Tone 2) 2 (Tone 3) 3 (Tone 4) 4 (Tone 5) 5 (Silent)
Example	line1 ring tone: 3

Incoming Call Interrupts Dialing Setting

Parameter – *incoming call cancels dialing* *Incoming Call Interrupts Dialing* (in Web UI)	**Aastra Web UI:** **Configuration Files**	Basic Settings->Preferences->General aastra.cfg, <mac>.cfg
Description	Enables 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 = disable), which is the default, the phone does not interrupt the outgoing call during dialing and the number you were dialling continues to display in the LCD. The phones sends the incoming call to a free line on the phone (or sends busy signal if all remaining lines are busy) and the LED for that line blinks. 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 3-Line LCD phones, 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.	

Description (Cont'd)	**Transfer/Conference Call Behavior** If you are dialing the phone to transfer or conference a call, and your phone receives an incoming call, your dialing is never interrupted (regardless of whether the "Incoming Call Interrupts Dialing" is enabled or disabled). For Transfer and Conference, the incoming calls always go to an available line (other than the one you are using for dialing) and the incoming call's line LED blinks. The LCD still displays your dialing screen. **Intercom Behavior** If "**Incoming Call Interrupts Dialing"** (or **incoming call cancels dialing** in config files) is enabled and you are dialing an outgoing Intercom call, the enabled interrupt setting takes precedence over an enabled "**Allow Barge In**" setting. The incoming call interrupts your dialing on an outgoing intercom call. On an incoming intercom call, the enabled "**Allow Barge In**" and "**Auto-Answer**" occurs while you are dialing to transfer or conference the call. However, the incoming call goes to an available idle line, and the LED blinks while you are dialing the second half of the conference or transfer. If "**Incoming Call Interrupts Dialing"** (or **incoming call cancels dialing** in config files) is disabled, an incoming intercom goes to an available idle line and the LED blinks for that line. The phone answers the call under all conditions.
Format	Boolean
Default Value	0 (disable)
Range	0 (disable) 1 (enable)
Example	incoming call cancels dialing: 1

Status Code on Ignoring Incoming Calls

Parameter – *sip ignore status code*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	Specifies the status code that is sent in the response to the server when a user ignores an incoming call.
Format	Integer
Default Value	486
Range	Valid SIP final negative response code (Refer to RFC3261)
Example	sip ignore status code: 486

Switch Focus to Ringing Line

Parameter – *switch focus to ringing line*	**Aastra Web UI:** **Configuration Files**	Basic Settings->Preferences->General aastra.cfg, <mac>.cfg
Description	Specifies whether or not the UI focus is switched to a ringing line while the phone is in the connected state.	
Format	Boolean	
Default Value	1 (enable)	
Range	0 (disable) 1 (enable)	
Example	switch focus to ringing line: 1	

Call Hold Reminder During Active Calls

Parameter – *call hold reminder during active calls* *Call Hold Reminder During Active Calls* (in Web UI)	**Aastra Web UI:** **Configuration Files**	Basic Settings->Preferences->Ring Tones aastra.cfg, <mac>.cfg
Description	Enables or disables the ability for the phone to initiate a continuous reminder tone on the active call when another call is on hold. For example, when the call on Line 1 is on hold, and the User answers a call on Line 2 and stays on that line, a reminder tone is played in the active audio path on Line 2 to remind the User that there is still a call on hold on Line 1. When this feature is disabled, a ring splash is heard when the active call hangs up and there is still a call on hold.	
Format	Boolean	
Default Value	0	
Range	0 (disable) 1 (enable)	
Example	call hold reminder during active calls: 1	

Call Hold Reminder

Parameter – *call hold reminder* *Call Hold Reminder* (in Web UI)	**Aastra Web UI**: Basic Settings->Preferences **Configuration Files** aastra.cfg, <mac>.cfg
Description	Enables or disables the reminder ring splash timer to start as soon as you put a call on hold (even when no other calls are active on the phone). When enabled, the phone initiates a reminder ring splash periodically for the single call on hold. When disabled, no reminder ring splash is audible.
Format	Boolean
Default Value	0
Range	0 (disable) 1 (enable)
Example	call hold reminder: 1

Call Hold Reminder Timer

Parameter – *call hold reminder timer*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	Specifies the time delay, in seconds, that a ring splash is heard on an active call when another call was placed on hold. For example, if a call comes into Line 1, and then a call comes into Line 2 and you answer Line 2, Line 1 is automatically placed on hold. While on the active Line 2, after 7 seconds, a ring splash audio sounds on the line reminding you that the call on Line 1 is still on hold. This timer begins to increment after Line 2 is answered. **Notes**: **1.** This parameter is used with the “call hold reminder frequency” parameter. **2.** You must enable this “call hold reminder timer” parameter for it to work. **3.** A value of “0” disables the call hold reminder feature.
Format	Integer
Default Value	7
Range	0-4294967295 seconds
Example	call hold reminder timer: 10

Call Hold Reminder Frequency

Parameter – *call hold reminder frequency*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	Specifies the time interval, in seconds, between each ring splash sound on the active line. For example, if a call comes into Line 1, and then a call comes into Line 2 and you answer Line 2, Line 1 is automatically placed on hold. While on the active Line 2, after 7 seconds, a ring splash audio sounds on the line reminding you that the call on Line 1 is still on hold (determined by the "call hold reminder timer" parameter), and then the ring splash is heard again after 60 seconds (determined by this parameter). **Notes**: **1.** You must enable the "call hold reminder" and/or "call hold reminder during active calls" parameter(s), and the "call hold reminder timer" parameter, for this parameter to work. **2.** A value of "0" prevents additional rings.
Format	Integer
Default Value	60
Range	0-4294967295
Example	call hold reminder frequency: 50

Preferred Line and Preferred Line Timeout

Parameter– *preferred line* *Preferred Line* (in Web UI)	**Configuration Files** **Aastra Web UI**	aastra.cfg, <mac>.cfg Basic Settings->Preferences->General
Description	Specifies the preferred line to switch focus to when incoming or outgoing calls end on the phone.	
Format	Integer	
Default Value	1	
Range	0 (none - disables the preferred line focus feature) 1-9	
Example	preferred line: 2	

Parameter– *preferred line timeout* *Preferred Line Timeout (seconds)* (in Web UI)	**Configuration Files** **Aastra Web UI**	aastra.cfg, <mac>.cfg Basic Settings->Preferences->General
Description	Specifies the time, in seconds, that the phone switches back to the preferred line after a call (incoming or outgoing) ends on the phone, or after a duration of inactivity on an active line.	
Format	Integer	
Default Value	0 (the phone returns the line to the preferred line immediately)	
Range	0-999	
Example	preferred line timeout: 30	

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 8 and 11-Line LCD phones:** 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 3-Line LCD phones:** 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	stutter disabled: 1	

Message Waiting Indicator Settings

Parameter – *mwi led line* *Message Waiting Indicator Line* (in Web UI)	**Aastra Web UI** **Configuration Files**	Basic Settings->Preferences->General 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 -9	
Example	mwi led line: 3	

Message Waiting Indicator Request URI Setting

Parameter– *sip linex mwi request uri*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	Specifies the URI to use when specifying which proxy server **Notes**: **1.** Quotes (“) must be used to enclose the value when specifying it with this parameter. **2.** Sip Explicit MWI Subscription must be enabled to use this feature.
Format	“sip:user@host:port”
Default Value	Not Applicable
Range	Not Applicable
Example	sip line1 mwi request uri: “sip:1020@10.50.224.53”

DND Key Mode Settings

Parameter – *dnd key mode*	**Aastra Web UI:** Basic Settings->Preferences->General **Configuration Files** aastra.cfg, <mac>.cfg
Description	Sets the mode for how the phone uses “do not disturb” (DND): • **account** Sets DND for a specific account. DND key toggles the account in focus on the IP Phone UI, to ON or OFF. • **phone** Sets DND ON for all accounts on the phone. DND key toggles all accounts on the phone to ON or OFF. • **custom** Sets the phone to display custom screens after pressing the DND key, that list the account(s) on the phone. The user can select a specific account for DND, turn DND ON for all accounts, or turn DND OFF for all accounts. **Notes**: **1.** If there is only one account configured on the phone, then the mode setting is ignored and the phone behaves as if the mode was set to “Phone”. **2.** You must configure a DND key on the phone to use this feature. To configure a DND key, see “Softkey/Programmable Key/Feature Key/ Expansion Module Key Parameters” on page A-202.
Format	Integer
Default Value	1 (phone)
Range	0 (account) 1 (phone) 2 (custom)
Example	dnd key mode: 2

Example

The following is an example of configuring the mode for DND in the configuration files:

```
dnd key mode: 2
softkey1 type: dnd
softkey1 states: idle connected incoming outgoing busy
```

In the above example, softkey 1 is configured for DND for line 1 only, with a “**custom**” configuration. Pressing softkey 1 displays DND screens for which you can customize on the phone.

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	

Parameter – *alert community 1* *community-1* (in Web UI)	**Aastra Web UI:** Basic Settings->Preferences-> Priority Alerting Settings **Configuration Files** 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:** Basic Settings->Preferences-> Priority Alerting Settings **Configuration Files** 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:** Basic Settings->Preferences-> Priority Alerting Settings **Configuration Files** aastra.cfg, <mac>.cfg
Description	When an "alert priority" keyword appears in the header of the INVITE request, the configured Bellcore ring tone is applied to the IP phone.
Format	Integer
Default Value	0 Normal ringing
Range	0 Normal ringing (default) 1 Bellcore-dr2 2 Bellcore-dr3 3 Bellcore-dr4 4 Bellcore-dr5 5 Silent
Example	alert priority: 4

Bellcore Cadence Settings

Parameter– *bellcore cadence dr2*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	Sets the cadence for Bellcore-dr2. **Note:** You can define up to 8 cadence rings. The value of -1 indicates “do not repeat”.
Format	Integer
Default Value	800,400, 800,4000
Range	Not Applicable
Example	bellcore cadence dr2: 800, 400, 800, 4000

Parameter– *bellcore cadence dr3*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	Sets the cadence for Bellcore-dr3. **Note:** You can define up to 8 cadence rings. The value of -1 indicates “do not repeat”.
Format	Integer
Default Value	400,200,400,200,800,4000
Range	Not Applicable
Example	bellcore cadence dr3: 400,200,400,200,800,4000

Parameter– *bellcore cadence dr4*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	Sets the cadence for Bellcore-dr4. **Note:** You can define up to 8 cadence rings. The value of -1 indicates “do not repeat”.
Format	Integer
Default Value	300,200,1000,200,300,4000
Range	Not Applicable
Example	bellcore cadence dr4: 300,200,1000,200,300,300,200,4000

Parameter– *bellcore cadence dr5*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	Sets the cadence for Bellcore-dr5. **Note:** You can define up to 8 cadence rings. The value of -1 indicates “do not repeat”.
Format	Integer
Default Value	500,-1
Range	Not Applicable
Example	bellcore cadence dr5: 500,-1

SIP Diversion Display

Global Setting

Parameter – *sip diversion display*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	Global parameter that enables/disables the display of the Caller ID and/or caller name, and reason for diversion on the phones LCD for an outgoing call that is being diverted to another destination. The phone that receives the diverted call displays the Caller ID of the original call destination. **Note:** You must restart the phone after setting a value for this parameter.
Format	Boolean
Default Value	1
Range	0 (disabled - do not display diversion information to the phone’s LCD) 1 (enabled - display diversion information to the phone’s LCD)
Example	sip diversion display: 0

Per-Line Setting

Parameter – *sip lineN diversion display* (N is a line number from 1 to 9)	**Configuration Files** aastra.cfg, <mac>.cfg
Description	For a specific line on the phone, this parameter enables/disables the display of the Caller ID and/or caller name, and reason for diversion on the phones LCD for an outgoing call that is being diverted to another destination. The phone that receives the diverted call displays the Caller ID of the original call destination. **Note:** You must restart the phone after setting a value for this parameter.
Format	Boolean
Default Value	1
Range	0 (disabled - do not display diversion information to the phone's LCD) 1 (enabled - display diversion information to the phone's LCD)
Example	sip line1 diversion display: 0

Display of Call Destination for Incoming Calls

Parameter – *show call destination name*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	Enable/disables the display of the call destination name to the LCD on the phone during incoming calls.
Format	Boolean
Default Value	0
Range	0 (disabled) 1 (enabled)
Example	show call destination name: 1

Language Settings

Parameter – *language* *Webpage Language* (in Web UI)	**IP Phone UI** **Aastra Web UI** **Configuration File**	Options->Language Basic Settings->Preferences-> Language Settings->Webpage Language aastra.cfg, <mac>.cfg
Description	The language you want to display in the IP Phone UI and the Aastra Web UI. Valid values for all phones are: 0 (English) default 1-4 The values 1-4 are dependent on the "Language N" parameter. For example, if "language 1: lang_fr.txt", then "language: 1" would set the webpage language to French. Valid values for CT cordless handsets are: 0 (English) 1-2 **Note:** Values 1-2 can only be set to either French or 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 -40.	
Format	Integer	
Default Value	0	
Range	0 to 4 (all phones) 0 to 2 (for CT handsets)	
Example	language: 1	

Parameter– *input language* *Input Language* (in Web UI)	**IP Phone UI:** **Aastra Web UI:** **Configuration Files**	Language->Input Language Basic Settings->Preferences->Language Settings aastra.cfg, <mac>.cfg
Description	Allows you to specify the language to use for inputs on the IP Phone. Entering a language value for this parameter allows users to enter text and characters in the IP Phone UI, Aastra Web UI, and in XML applications via the keypad on the phone, in the language(s) specified.	
Format	Text	
Default Value	English	
Range	Valid values are: • English • French • Français • German • Deutsch • Italian • Italiano • Spanish • Español • Portuguese • Português • Russian • Русский • Nordic	
Example	input language: French	

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_de.txt (German) lang_dk.txt (Danish) lang_es.txt (Spanish) lang_es_mx.txt (Mexican Spanish) lang_fi.txt (Finnish) lang_fr.txt (French) lang_fr_ca.txt (Canadian French) lang_it.txt (Italian) lang_no.txt (Norwegian) lang_pt.txt (Portuguese) lang_pt_br.txt (Brazillian Portuguese) lang_ru.txt (Russian) lang_sv.txt (Swedish) You can use this parameter in three ways: • To download no language packs • To download a language pack(s) from the original configuration server • To download a language pack(s) from another specified server **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 -40. **2.** You must reboot the phone to load a language pack. **3.** To download a specific file, the string value MUST HAVE A FILENAME at the end of the string. For example: language 1:ftp://admin:admin!@1.2.3.4:50/path/lang_de.txt where "path" is the directory and "lang_de.txt" is the filename. If you do not specify a filename, the download fails. See examples for each below.
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-179 and "Country Codes (from Standard ISO 3166)" on page A-179.
Default Value	Not Applicable

Range	Not Applicable
Example	The following example downloads no language pack file: language 1: The following example downloads the German language pack to the phones from the original configuration server: language 1: lang_de.txt The following example uses FTP to download the firmware file “lang_de.txt” (German language pack) from the “path” directory on server 1.2.3.4 using port 50: language 1:ftp://admin:admin!@1.2.3.4:50/path/lang_de.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
European French	fr
French Canadian	fr_ca
European Spanish	es
Mexican Spanish	es_mx
German	de
Italian	it
Portuguese	pt
Brazillian Portuguese	br_pt
Russian	ru
Swedish	sv
Danish	dk
Finnish	fi

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 BAHRAIN BANGLADESH BARBADOS BELARUS BELGIUM BELIZE BENIN BERMUDA BHUTAN BOLIVIA BOSNIA AND HERZEGOVINA BOTSWANA BOUVET ISLAND BRAZIL BRITISH INDIAN OCEAN TERRITORY BRUNEI DARUSSALAM BULGARIA BURKINA FASO BURUNDI	BS BH BD BB BY BE BZ BJ BM BT BO BA BW BV BR IO BN BG BF BI
CAMBODIA CAMEROON CANADA CAPE VERDE CAYMAN ISLANDS CENTRAL AFRICAN REPUBLIC CHAD CHILE CHINA CHRISTMAS ISLAND COCOS (KEELING) ISLANDS COLOMBIA COMOROS CONGO CONGO, THE DEMOCRATIC REPUBLIC OF THE COOK ISLANDS COSTA RICA CÔTE D'IVOIRE CROATIA CUBA CYPRUS CZECH REPUBLIC	KH CM CA CV KY CF TD CL CN CX CC CO KM CG CD CK CR CI HR CU CY CZ
DENMARK Dhcp (see Chapter 4, the section, "DHCP Time Offset (Option 2) Support" on page 5-23) DJIBOUTI DOMINICA DOMINICAN REPUBLIC	DK DP DJ DM DO

Country	Country Code
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
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

Country	Country Code
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
MACAO MACEDONIA, THE FORMER YUGOSLAV REPUBLIC OF MADAGASCAR MALAWI MALAYSIA MALDIVES MALI MALTA MARSHALL ISLANDS MARTINIQUE MAURITANIA MAURITIUS MAYOTTE MEXICO MICRONESIA, FEDERATED STATES OF MOLDOVA, REPUBLIC OF MONACO MONGOLIA MONTENEGRO MONTSERRAT MOROCCO MOZAMBIQUE MYANMAR	MO MK MG MW MY MV ML MT MH MQ MR MU YT MX FM MD MC MN ME MS MA MZ MM
NAMIBIA NAURU NEPAL NETHERLANDS NETHERLANDS ANTILLES NEW CALEDONIA NEW ZEALAND NICARAGUA NIGER NIGERIA NIUE NORFOLK ISLAND NORTHERN MARIANA ISLANDS NORWAY	NA NR NP NL AN NC NZ NI NE NG NU NF MP NO
OMAN	OM

Country	Country Code
PAKISTAN	PK
PALAU	PW
PALESTINIAN TERRITORY, OCCUPIED	PS
PANAMA	PA
PAPUA NEW GUINEA	PG
PARAGUAY	PY
PERU	PE
PHILIPPINES	PH
PITCAIRN	PN
POLAND	PL
PORTUGAL	PT
PUERTO RICO	PR
QATAR	QA
RÉUNION	RE
ROMANIA	RO
RUSSIAN FEDERATION	RU
RWANDA	RW
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

Country	Country Code
TAIWAN, PROVINCE OF CHINA TAJIKISTAN TANZANIA, UNITED REPUBLIC OF THAILAND TIMOR-LESTE TOGO TOKELAU TONGA TRINIDAD AND TOBAGO TUNISIA TURKEY TURKMENISTAN TURKS AND CAICOS ISLANDS TUVALU	TW TJ TZ TH TL TG TK TO TT TN TR TM TC TV
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	1 (enabled)
Range	0 (disabled) 1 (enabled)
Example	suppress dtmf playback: 0

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** Basic Settings->Preferences->Outgoing Intercom Settings **Configuration Files** 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** Basic Settings->Preferences->Outgoing Intercom Settings **Configuration Files** 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	Not Applicable
Range	Not Applicable
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 intercom warning tone* *Play Warning Tone* (in Web UI)	**Aastra Web UI** **Configuration Files**	Basic Settings->Preferences-> Incoming Intercom Settings 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 intercom warning tone: 0	

Parameter – *sip intercom allow barge in* *Allow Barge In* (in Web UI)	**Aastra Web UI:** **Configuration Files**	Basic Settings->Preferences-> Incoming Intercom Settings 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	

Enable Microphone During Early Media

Parameter – *sip early media mute mic*	**Configuration Files**aastra.cfg, <mac>.cfg
Description	Enables or disables the microphone while in early media.
Format	Boolean
Default Value	0
Range	0-1 0 (disables mic during early media) 1 (enables mic during early media)
Example	sip early media mute mic: 1

Group Paging RTP Settings

Parameter – *paging group listening*	**Aastra Web UI:** **Configuration Files**	Basic Settings->Preferences-> Group Paging RTP Settings aastra.cfg, <mac>.cfg
Description	Specifies the multicast address(es) and the port on which the phone listens for incoming multicast RTP packets. **Note:** If this field is blank, Paging listening capability is disabled on the phone.	
Format	IP Address in dotted decimal format/Port #	
Default Value	Not Applicable	
Range	Not Applicable	
Example	paging group listening: 224.0.0.2:10000,239.0.1.20:15000 0	

Example

The following is an example of configuring RTP streaming for Paging applications using the configuration files:

```
paging group listening: 224.0.0.2:10000,239.0.1.20:15000
softkey1 type: paging
softkey1 label: group 1
softkey1 value: 224.0.0.2:10000,239.0.1.20:15000
```

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

Disable User Login to Aastra Web UI

Parameter – *web interface enabled*	**Configuration Files**aastra.cfg, <mac>.cfg
Description	Specifies whether or not to disable the web user interface
Format	Integer
Default Value	1 (admin/user enabled)
Range	0 (admin/user disabled) 1 (admin/user enabled) 2 (only admin enabled)
Example	web interface enabled: 0

Minimum Ringer Volume

Parameter – *ringer volume minimum*	**Configuration Files**aastra.cfg, <mac>.cfg
Description	Specifies the minimum ringer volume level
Format	Integer
Default Value	0
Range	0-9
Example	ringer volume minimum: 1

Terminated Calls Indicator

Parameter – *far end disconnect timer*	**Configuration Files**aastra.cfg, <mac>.cfg
Description	Specifies whether or not the phone displays an indication of a terminated call. If set to 0, this feature is disabled and the phone does not display the “Call Terminated” screen. If you specify a value for this paramter other than “0”, the “Call Terminated” screen displays for the configured time interval. The audible busy tone also plays for the configured time interval specified.
Format	Integer
Default Value	0 (disable)
Range	0 to 86400 seconds
Example	far end disconnect timer: 5

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** Basic Settings->Preferences-> Directed Call Pickup Settings **Configuration Files** 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	

Mapping Key Settings

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** **Configuration Files**	Basic Settings->Preferences->Key Mapping 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 6757i CT Base Station, the **Redial** key on the 6757i CT handset retains its original functionality. The **Redial** key on the handset is not configured for speeddial.	
Format	Integer	
Default Value	Not Applicable	
Range	Not Applicable	
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 6757i CT Base Station, the **Conf** key on the 6757i CT handset retains its original functionality. The **Conf** key on the handset is not configured for speeddial.	
Format	Integer	
Default Value	Not Applicable	
Range	Not Applicable	
Example	map conf key to: 5551267	

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
9143i	-	Not Applicable	7	9	-
9480i	6	Not Applicable	-	9	-
9480i CT	6	Not Applicable	-	9	15
6730i	-	Not Applicable	8	6	-
6731i	-	Not Applicable	8	6	-
6739i	55	36 to 108* (Model M670i) 60 to 180** (Model M675i)	-	9	-
6753i	-	36 to 108* (Model M670i)	6***	9	-
6755i	6 Bottom Keys	36 to 108* (Model M670i) 60 to 180** (Model M675i)	6 Top Keys	9	-
6757i	12 Top and Bottom Softkeys	36 to 108* (Model M670i) 60 to 180** (Model M675i)	-	9	-
6757i CT	12 Top and Bottom Softkeys	36 to 108* on Base Station 60 to 180** on Base Station (Model M675i)	-	9	15

*The M670i expansion module consists of 36 softkeys. You can have up to 3 expansion modules on an IP phone totaling 108 softkeys. Valid for 6739i, 6753i, 6755i, 6757i, and 6757i CT phones.

**The M675i expansion module consists of 60 softkeys. You can have up to 3 expansion modules on an IP phone totaling 180 softkeys. Valid for 6739i, 6755i, 6757i, and 6757i CT phones.

***On the 6753i, two of the 6 programmable keys are the DELETE and SAVE keys and can be programmed only if Administrator allows.

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

Softkey Settings for 8 and 11-Line LCD phones

The value of "*N*" for the following parameters is dependent on the number of softkeys available on the 9480i, 9480i CT, 6755i, 6757i, and 6757i CT models. See the table above for applicable values.

<table>
<tr><td>Parameter –
softkeyN type

Type
(in Web UI)</td><td>Aastra Web UI
Configuration Files</td><td>Operation->Softkeys and XML
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 key is disabled.
• line - Indicates softkey is configured for line use.
• speeddial - Indicates softkey is configured for speeddial use. Speeddial is applicable to the M670i and M675i 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 is configured for do not disturb on the phone. This option is "Do Not Disturb" in the Aastra Web UI). You must also set the DND key mode. See
• blf - Indicates softkey 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 M670i and M675i also.
• 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. You can also use the “BLF List URI” parameter to specify a URI for the phone to access for the BLF List.
• 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.</td></tr>
</table>

	• **flash** - Indicates the softkey is set to generate a flash event when it is pressed, or a feature key is pressed on the CT handsets. 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. • **callforward** - Indicates the softkey is configured for accessing the Call Forward features on the phone. A Call Forwarding Mode must be enabled to use this softkey. "Account" mode is the default. • **blfxfer** - Indicates the softkey is configured to transfer calls AND configured for BLF on a single key. • **speeddialxfer** - Indicates the softkey is configured to transfer calls AND configured for speeddialing to a specific number. • **speeddialconf** - Indicates the softkey is configured to be used as a speeddial key AND as a conference key. • **directory** - Indicates the softkey is set for accessing the Directory List. • **callers** - Indicates the softkey is set for accessing the Callers 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. • **paging** - Indicates the softkey is set for Group Paging on the phone. Pressing this key automatically sends a Real Time Transport Protocol (RTP) stream to pre-configured multicast address(es) without involving SIP signaling. • **empty** - Indicates the softkey 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 callforward blfxfer speeddialxfer speeddialconf callers (“Callers List” in Aastra Web UI) directory icom services phonelock paging 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

<table>
<tr><td>Parameter –
softkeyN label

Label
(in Web UI)</td><td>Aastra Web UI Operation->Softkeys and XML
Configuration Files aastra.cfg, <mac>.cfg</td></tr>
<tr><td>Description</td><td>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
• blfxfer
• speeddialxfer
• speeddialconf
• directory
• callers
• icom
• services
• paging

Notes:
1. For 8 and 11-Line LCD 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.</td></tr>
<tr><td>Format</td><td>Text</td></tr>
<tr><td>Default Value</td><td>Not Applicable</td></tr>
<tr><td>Range</td><td>For line, blf types - Up to 9 characters.
For speeddial type - Up to 11 characters.</td></tr>
<tr><td>Example</td><td>softkey1 label: “Line 9”
softkey2 label: “info”
softkey3 label: flash
softkey4 label: “johnsmith”</td></tr>
</table>

Parameter – *softkeyN value* *Value* (In Web UI)	**Aastra Web UI** **Configuration Files**	Operation->Softkeys and XML aastra.cfg, <mac>.cfg
Description	This is the value you assign to the softkey. 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 • blfxfer • speeddialxfer • speeddialconf • paging **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 -209. For Park/Pickup examples, see Chapter 5, the section, "Examples for Models with 8 and 11-Line LCDs" on page -208. **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: • $$SIPUSERNAME$$ • $$SIPAUTHNAME$$ • $$PROXYURL$$ • $$LINESTATE$$ • $$LOCALIP$$ • $$REMOTENUMBER$$ • $$DISPLAYNAME$$ • $$SIPUSERNAME$$ • $$INCOMINGNAME$$ • $$CALLDURATION$$ • $$CALLDIRECTION$$	
Format	String	
Default Value	Not Applicable	
Range	Not Applicable	
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 • blfxfer • speeddialxfer • speeddialconf	
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, callforward,blfxfer, speeddialxfer, speeddialconf, Directory, Callers List, Icom, Services, empty**: idle, connected, incoming, outgoing **For softkey type - Flash**: All states disabled **For softkey type - 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 9143i, 6730i, 6731i, 6753i, and 6755i

The value of "*N*" for the following parameters is dependent on the number of programmable keys available on the phone models. See the table on page -202 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. You can also use the "BLF List URI" parameter to specify a URI for the phone to access for the BLF List. • **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. • **icom** - Indicates programmable key is set to be used as the Intercom key.

	• **flash** - Indicates programmable key is set to generate a flash event when it is pressed. 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. • **callforward** - Indicates the programmable key is configured for accessing the Call Forward features on the phone. A Call Forwarding Mode must be enabled to use this softkey. “Account” mode is the default. • **blfxfer** - Indicates the programmable key is configured to transfer calls AND configured for BLF on a single key. • **speeddialxfer** - Indicates the programmable key is configured to transfer calls AND configured for speeddialing to a specific number. • **speeddialconf** - Indicates the programmable key is configured to be used as a speeddial key AND as a conference key. • **directory** - Indicates programmable key is configured to access the Directory List. • **callers** - Indicates programmable key is configured to access the Callers List. • **conf** - Indicates programmable key is configured as a conference key. Enter as “**conf**” in configuration files. • **xfer**- Indicates programmable key is configured as a Transfer key for transferring calls. Enter as “**xfer**” in configuration files. • **services** - Indicates the programmable key is set to be used as the Services key. • **phonelock** - Indicates the programmable key is set to be used to lock/unlock the phone. • **paging** - Indicates the softkey is set for Group Paging on the phone. Pressing this key automatically sends a Real Time Transport Protocol (RTP) stream to pre-configured multicast address(es) without involving SIP signaling. • **empty** - Indicates programmable key is configured to force a blank entry on the IP phone display for a specific programmable key.
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 callforward directory callers(“Callers List” in Aastra Web UI) conf xfer blfxfer speeddialxfer speeddialconf phonelock services paging empty
Example	prgkey3 type: speeddial

Parameter – *prgkeyN value* *Value* (in Web UI)	**Aastra Web UI** Operation->Programmable Keys **Configuration Files** aastra.cfg, <mac>.cfg
Description	This is the value you assign to the programmable key. 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 • blfxfer • speeddialxfer • speeddialconf • paging **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 - You can specify a URI to use for this XML softkey. The variables you can use with the XML softkey URI are: • $$SIPUSERNAME$$ • $$SIPAUTHNAME$$ • $$PROXYURL$$ • $$LINESTATE$$ • $$LOCALIP$$ • $$REMOTENUMBER$$ • $$DISPLAYNAME$$ • $$SIPUSERNAME$$ • $$INCOMINGNAME$$ • $$CALLDURATION$$ • $$CALLDIRECTION$ **6.** For Park, Pickup - For valid values, see Chapter 5, the section, "Park/Pickup Call Server Configuration Values" on page -209. For Park/Pickup examples, see Chapter 5, the section "Examples for Models with 3-Line LCDs" on page -208.
Format	String
Default Value	Not Applicable
Range	Not Applicable
Example	prgkey3 value: 411 prgkey4 value: 123456+ (example of a speeddial prefix)

Parameter– *prgkeyN line* *Line* (in Web UI)	**Aastra Web UI** Operation->Programmable Keys **Configuration Files** aastra.cfg, <mac>.cfg
Description	This is the line associated with the programmable key you are configuring. The "**prgkeyN line**" parameter can be set for the following softkey types only: • speeddial • blf • blf/List • acd • dcp • park • pickup • lcr • blfxfer • speeddialxfer • speeddialconf
Format	Integer
Default Value	1
Range	1 through 9
Example	prgkey3 line: 1 prgkey4 line: 5

Top Softkey Settings for 6757i and 6757i CT

Parameter – *topsoftkeyN type* *Top Softkeys->Type* (in Web UI)	**Aastra Web UI** **Configuration Files**	Operation->Softkeys and XML 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. You can also use the "BLF List URI" parameter to specify a URI for the phone to access for the BLF List. • **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, or a feature key is pressed on the CT handsets. 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. • **callforward** - Indicates the softkey is configured for accessing the Call Forward features on the phone. A Call Forwarding Mode must be enabled to use this softkey. "Account" mode is the default. • **blfxfer** - Indicates the softkey is configured to transfer calls AND configured for BLF on a single key. • **speeddialxfer** - Indicates the softkey is configured to transfer calls AND configured for speeddialing to a specific number. • **speeddialconf** - Indicates the softkey is configured to be used as a speeddial key AND as a conference key. • **callers** - 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. • **paging** - Indicates the softkey is set for Group Paging on the phone. Pressing this key automatically sends a Real Time Transport Protocol (RTP) stream to pre-configured multicast address(es) without involving SIP signaling. • **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 callforward park pickup blfxfer speeddialxfer speeddialconf lcr callers (“Callers List” in Aastra Web UI) directory icom services phonelock paging 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 • blfxfer • speeddialxfer • speeddialconf • paging • directory • callers list • icom • services **Notes**: **1.** For the 6757i and 6757i 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 • blfxfer • speeddialxfer • speeddialconf • paging **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 -209. For Park/Pickup examples, see Chapter 5, the section, "Examples for Models with 8 and 11-Line LCDs" on page -208. **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: • $$SIPUSERNAME$$ • $$SIPAUTHNAME$$ • $$PROXYURL$$ • $$LINESTATE$$ • $$LOCALIP$$ • $$REMOTENUMBER$$ • $$DISPLAYNAME$$ • $$SIPUSERNAME$$ • $$INCOMINGNAME$$ • $$CALLDURATION$$ • $$CALLDIRECTION$$	
Format	String	
Default Value	Not Applicable	
Range	Not Applicable	
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)	

<table>
<tr><td>Parameter –
topsoftkeyN line

Top Softkeys->Line
(in Web UI)</td><td>Aastra Web UI
Configuration Files</td><td>Operation->Softkeys and XML
aastra.cfg, <mac>.cfg</td></tr>
<tr><td>Description</td><td colspan="2">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
• blfxfer
• speeddialxfer
• speeddialconf</td></tr>
<tr><td>Format</td><td colspan="2">Integer</td></tr>
<tr><td>Default Value</td><td colspan="2">1</td></tr>
<tr><td>Range</td><td colspan="2">1 through 9</td></tr>
<tr><td>Example</td><td colspan="2">topsoftkey1 line: 1
topsoftkey2 line: 5</td></tr>
</table>

Handset Feature Key Settings (9480i CT and 6757i 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. Value can be: line1, line2, line3, line4, line5, line6, line7, line8, or line9. • **xfer**- Indicates feature key is configured for transferring a call. • **conf** - 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 6757i 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** - 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 CT handsets. 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 line1, line2, line3, line4, line5, line6, line7, line8, line9 transfer conference public icom directory callers (“Callers List” in Aastra Web UI) park pickup flash
Example	featurekey1 type: line3 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 6757i 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	

Expansion Module Key Settings for M670i (6753i, 6755i, 6757i, 6757i CT) and M675i (for 6755i, 6757i, 6757i CT)

Parameter – *expmodX keyN type* *Type* (in Web UI)	**Aastra Web UI** **Configuration Files**	Operation->Expansion Module N 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. You can also use the "BLF List URI" parameter to specify a URI for the phone to access for the BLF List. • **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, or a feature key is pressed on the CT handsets. 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	• **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. • **callforward** - Indicates the softkey is configured for accessing the Call Forward features on the phone. A Call Forwarding Mode must be enabled to use this softkey. “Account” mode is the default. • **blfxfer** - Indicates the softkey is configured to transfer calls AND configured for BLF on a single key. • **speeddialxfer** - Indicates the softkey is configured to transfer calls AND configured for speeddialing to a specific number. • **speeddialconf** - Indicates the softkey is configured to be used as a speeddial key AND as a conference key. • **callers** - 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 6753i) 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. • **paging** - Indicates the softkey is set for Group Paging on the phone. Pressing this key automatically sends a Real Time Transport Protocol (RTP) stream to pre-configured multicast address(es) without involving SIP signaling. • **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 • flash • sprecode • park • pickup • lcr • callforward • blfxfer • speeddialxfer • speeddialconf • callers (“Callers List” in Aastra Web UI) • directory • icom • services (not available on the 6753i) • phonelock • paging • empty
Example	expmod1 key1 type: line expmod1 key2 type: speeddial expmod1 key3 type: blf expmod1 key4 type: list

<table>
<tr><td>Parameter –
expmodX keyN label

Label
(in Web UI)</td><td>Aastra Web UI Operation->Expansion Module N
Configuration Files aastra.cfg, <mac>.cfg</td></tr>
<tr><td>Description</td><td>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
• flash
• sprecode
• park
• pickup
• directory
• callers list
• icom
• services
• blfxfer
• speeddialxfer
• speeddialconf
• paging

Note: For 8 and 11-Line LCD phones, an icon appears beside the soft key label that indicates the status of the line.</td></tr>
<tr><td>Format</td><td>Text</td></tr>
<tr><td>Default Value</td><td>Not Applicable</td></tr>
<tr><td>Range</td><td>For line, blf types - Up to 9 characters.
For speeddial type - Up to 11 characters.</td></tr>
<tr><td>Example</td><td>expmod1 key1 label: “Line 9”
expmod2 key1 label: “info”
expmod3 key1 label: “johnsmith”</td></tr>
</table>

<table>
<tr><td>Parameter –
expmodX keyN value

Value
(In WEb UI)</td><td>Aastra Web UI Operation->Expansion Module N
Configuration Files aastra.cfg, <mac>.cfg</td></tr>
<tr><td>Description</td><td>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
• sprecode
• park
• pickup
• directory
• callers (“Callers List” in Aastra Web UI)
• icom
• services (not available on 6753i)
• blfxfer
• speeddialxfer
• speeddialconf
• paging

Notes:
1. For the 6757i and 6757i 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.
3. For xml - You can specify a URI to use for this XML softkey. The variables you can use with the XML softkey URI are:
• $$SIPUSERNAME$$
• $$SIPAUTHNAME$$
• $$PROXYURL$$
• $$LINESTATE$$
• $$LOCALIP$$
• $$REMOTENUMBER$$
• $$DISPLAYNAME$$
• $$SIPUSERNAME$$
• $$INCOMINGNAME$$
• $$CALLDURATION$$
• $$CALLDIRECTION$$</td></tr>
<tr><td>Format</td><td>String</td></tr>
<tr><td>Default Value</td><td>Not Applicable</td></tr>
<tr><td>Range</td><td>Not Applicable</td></tr>
<tr><td>Example</td><td>expmod1 key1 value: 9
expmod1 key2 value: 411
expmod1 key3 value: 123456+ (example of a speeddial prefix)</td></tr>
</table>

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 • park • pickup • blfxfer • speeddialxfer • speeddialconf	
Format	Integer	
Default Value	1	
Range	1 through 9	
Example	expmod1 key1 line: 1 expmod1 key2 line: 5	

Customizing the Key Type List

Softkeys, Programmable Keys, Expansion Module Keys

Parameter – *softkey selection list*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	Allows you to specify which key types to display and the order in which to display them in the "**Type**" list for softkeys, programmable keys, and/or expansion module keys when configuring the keys in the Aastra Web UI. If no value is specified for this "**softkey selection list**" parameter, the key "**Type**" list displays ALL of the key types by default in the Aastra Web UI. **Notes**: **1.** Any key types configured that do not apply to the phone's environment are ignored. **2.** The SAVE and DELETE keys appear by default as Keys 1 and 2 on the 6753i and 9143i, and as 5 and 6 on the 6730i and 6731i unless specifically changed by your Administrator. **3.** An Administrator must use the English value when configuring the key types in the configuration files. **4.** Any key type already configured on a phone displays in that key's "**Type**" list, in addition to the values specified for this parameter. **5.** After configuring specific key types for a phone, the key types in the Aastra Web UI display the same for both the User and Administrator Web interfaces for that phone.
Format	Alpha Characters in a comma separated list
Default Value	(see list below)
Range	Any of the key types in the "Default Value" field above.
Example	softkey selection list: blf, speeddial, line, xml

Default Value:

• none	• callforward
• line	• blf/xfer
• speeddial	• speeddial/xfer
• dnd	• speeddial/conf
• blf	• directory
• list	• callers
• acd	• conf (6730i, 6731i, 6753i, 9143i)
• dcp	• xfer (6730i, 6731i, 6753i, 9143i)
• xml	• icom
• flash	• services
• sprecode	• phonelock
• park	• paging
• pickup	• empty (6755i, 6757i, 6757 CT, 9480i, 9480i CT)
• lcr	

Handset Feature Keys (for CT Models ONLY)

<table>
<tr><td>Parameter –
feature key selection list</td><td>Configuration Files aastra.cfg, <mac>.cfg</td></tr>
<tr><td>Description</td><td>Allows you to specify which key types to display and the order in which to display them in the “Type” list for the feature keys (CT models only) when configuring the keys in the Aastra Web UI.

If no value is specified for this “feature key selection list” parameter, the key “Type” list displays ALL of the key types by default in the Aastra Web UI.

:
1. Any key types configured that do not apply to the phone’s environment are ignored.
2. An Administrator must use the English value when configuring the key types in the configuration files.
3. Any key type already configured on a phone displays in that key’s “Type” list, in addition to the values specified for this parameter.
4. After configuring specific key types for a phone, the key types in the Aastra Web UI display the same for both the User and Administrator Web interfaces for that phone.</td></tr>
<tr><td>Format</td><td>Alpha Characters in a comma separated list</td></tr>
<tr><td>Default Value</td><td><table>
<tr><td>• none</td><td>• park</td></tr>
<tr><td>• line</td><td>• pickup</td></tr>
<tr><td>• icom</td><td>• conf</td></tr>
<tr><td>• directory</td><td>• private</td></tr>
<tr><td>• callers</td><td>• public</td></tr>
<tr><td>• xfer</td><td>• flash</td></tr>
</table></td></tr>
<tr><td>Range</td><td>Any of the key types in the “Default Value” field above.</td></tr>
<tr><td>Example</td><td>feature key selection list: line, directory, callers</td></tr>
</table>

Locking Softkeys and Programmable Keys

Parameter– *softkeyN locked*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	Locks the specified softkey on the 9480i, 9480i CT, 6755i, 6757i, or 6757i 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 6757i or 6757i 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 programmable key on the 9143i, 6730i, 6731i, 6753i, or 6755i 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 feature key on the CT handset. 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 keyN 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 locked: 1

Locking the SAVE and DLETE Keys (6753i)

Parameter– *prgkey1 locked*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	Allows you to lock or unlock the **Save** key on the 6753i IP Phone. When the **Save** key is unlocked, a User can change the function of the key using the Aastra Web UI. An Administrator can change the function of the key using the Aastra Web UI or the configuration files. **Note**: **1.** The save function on the 6753i IP Phone is limited to Key 1 only. **2.** Changing the function from the **Save** key to another function, removes the ability to save items on the IP phone.
Format	Boolean
Default Value	1 (lock)
Range	0 (unlock) 1 (lock)
Example	prgkey1 locked: 0

Parameter– *prgkey2 locked*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	Allows you to lock or unlock the **Delete** key on the 6753i IP Phone. When the **Delete** key is unlocked, a User can change the function of the key using the Aastra Web UI. An Administrator can change the function of the key using the Aastra Web UI or the configuration files. **Note**: **1.** The delete function on the 6753i IP Phone is limited to Key 2 only. **2.** Changing the function from the **Delete** key to another function, removes the ability to delete items on the IP phone.
Format	Boolean
Default Value	1 (lock)
Range	0 (unlock) 1 (lock)
Example	prgkey2 locked: 0

Enabling/Disabling Ability to Add/Edit Speeddial Keys

Parameter – *speeddial edit*	**Configuration Files**	aastra.cfg, <mac>.cfg
Description	Allows you to enable or disable the ability to add a speeddial key or edit a speeddial key. The default is enabled (Yes) allowing you to create and edit speeddial keys on the phone using the Press-and-hold feature, softkeys, programmable keys, expansion module keys and key pad, speeddial menu in the IP Phone UI, and the SAVE TO key. If this parameter is set to disabled (No), it blocks the user from using any of the features on the phone to create or edit a speeddial key.	
Format	Boolean	
Default Value	1 (Enabled)	
Range	0 (Disabled) 1 (Enabled)	
Example	speeddial edit: 0	

BLF List URI Settings

Parameter– *list uri* *BLF List URI* (in Web UI)	**Aastra Web UI** **Configuration Files**	Operation->Softkeys and XML->Services Operation->Programmable Keys->Services Operation->Expansion Module Keys->Services aastra.cfg, <mac>.cfg
Description	Specifies the URI that the phone uses to access the BLF list on the Broadsoft server when the BLF list key is pressed. When you specify a URI for this parameter, the phone uses the Internet to access the BLF list on the Broadsoft server.	
Format	HTTP server path or Fully Qualified Domain Name	
Default Value	Not Applicable	
Range	Not Applicable	
Example	list uri: my6757i-blf-list@as.broadworks.com	

Customizing M675i Expansion Module Column Display

Expansion Module 1 through 3

Parameter– *expmodXpageNleft*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	Allows you to specify a customized heading for the M675i expansion module, in the left column of a specific page. You can specify the following options for this parameter: **Expansion Module 1** expmod1page1left (Expansion Module 1, Page 1, left column) expmod1page2left (Expansion Module 1, Page 2, left column) expmod1page3left (Expansion Module 1, Page 3, left column) **Expansion Module 2** expmod2page1left (Expansion Module 2, Page 1, left column) expmod2page2left (Expansion Module 2, Page 2, left column) expmod2page3left (Expansion Module 2, Page 3, left column) **Expansion Module 3** expmod3page1left (Expansion Module 3, Page 1, left column) expmod3page2left (Expansion Module 3, Page 2, left column) expmod3page3left (Expansion Module 3, Page 3, left column)
Format	Text String
Default Value	Not Applicable
Range	Not Applicable
Example	expmod1page1left: Personnel Ext

Parameter– *expmodXpageNright*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	Allows you to specify a customized heading for the M675i expansion module, in the right column of a specific page. You can specify the following options for this parameter: **Expansion Module 1** expmod1page1right (Expansion Module 1, Page 1, right column) expmod1page2right (Expansion Module 1, Page 2, right column) expmod1page3right (Expansion Module 1, Page 3, right column) **Expansion Module 2** expmod2page1right (Expansion Module 2, Page 1, right column) expmod2page2right (Expansion Module 2, Page 2, right column) expmod2page3right (Expansion Module 2, Page 3, right column) **Expansion Module 3** expmod3page1right (Expansion Module 3, Page 1, right column) expmod3page2right (Expansion Module 3, Page 2, right column) expmod3page3right (Expansion Module 3, Page 3, right column)
Format	Text String
Default Value	Not Applicable
Range	Not Applicable
Example	expmod1page1right: Operations Ext

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	

Semi-Attended Transfer Settings

Parameter – *sip refer-to with replaces*	**Configuration Files**	aastra.cfg, <mac>.cfg
Description	Flag for controlling the mode of a semi-attended transfer.	
Format	Boolean	
Default Value	0	
Range	0 or1	
Example	sip refer-to with replaces: 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 Settings.

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 Setting

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 6757i CT and 9480i CT base unit and the handset. When this feature is enabled (set to 1), you can make separate active calls from the 6757i CT and 9480i 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 Setting

Parameter – *sip blacklist duration* *Blacklist Duration (Aastra Web UI)*	**Aastra Web UI** Advanced Settings->Global SIP Settings->Advanced SIP Settings **Configuration Files** 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 Setting

Parameter – *sip whitelist* *Whitelist Proxy* (Aastra Web UI)	**Aastra Web UI** Advanced Settings->Global SIP->Advanced SIP Settings **Configuration Files** 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

XML Key Redirection Settings (for Redial, Xfer, Conf, Icom, Voicemail)

Parameter– *redial script*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	Specifies a redial script for the phone to use. When this parameter is set, pressing the Redial key GETs the specified URI from the server to use in performing the redial action.
Format	String
Default Value	empty
Range	Any valid URI
Example	redial script: http://bluevelvet.ana.aastra.com/redial.php

Parameter– *xfer script*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	Specifies an Xfer script for the phone to use. When this parameter is set, pressing the Xfer key GETs the specified URI from the server instead of starting the transfer action.
Format	String
Default Value	empty
Range	Any valid URI
Example	xfer script: http://bluevelvet.ana.aastra.com/xfer.php

Parameter– *conf script*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	Specifies an Conf script for the phone to use. When this parameter is set, pressing the Conf key GETs the specified URI from the server to use in performing the conference action.
Format	String
Default Value	empty
Range	Any valid URI
Example	conf script: http://bluevelvet.ana.aastra.com/conf.php

Parameter– *icom script*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	Specifies an Icom script for the phone to use. When this parameter is set, pressing the Icom key GETs the specified URI from the server to use in performing the Intercom action.
Format	String
Default Value	empty
Range	Any valid URI
Example	icom script: http://bluevelvet.ana.aastra.com/icom.php

Parameter– *voicemail script*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	Specifies an Voicemail script for the phone to use. When this parameter is set, selecting the voicemail option from the Services Menu GETs the specified URI from the server instead of starting the Voicemail application.
Format	String
Default Value	empty
Range	Any valid URI
Example	voicemail script: http://bluevelvet.ana.aastra.com/voicemail.php

Options Key Redirection Setting

Parameter– *options script*	**Configuration Files**	aastra.cfg, <mac>.cfg
Description	Specifies an Options script for the phone to use. When this parameter is set, pressing the Options Key GETs the specified URI from the server. **Note:** Pressing and holding the Options key displays the local Options Menu on the phone.	
Format	String	
Default Value	empty	
Range	Any valid URI	
Example	options script: http://fargo.ana.aastra.com/options.xml	

Off-Hook and XML Application Interaction Setting

Parameter– *auto offhook*	**Configuration Files**	aastra.cfg, <mac>.cfg
Description	Specifies whether or not the phone is prevented from entering the off-hook/ dialing state, if the handset is off-hook for more than 2 seconds, and the call ends.	
Format	Boolean	
Default Value	0 (disabled)	
Range	0 (disabled - phone is prevented from entering the off-hook dialing state) 1 (enabled - allows phone to enter the off-hook dialing state)	
Example	auto offhook: 1	

XML Override for a Locked Phone Setting

Parameter– *xml lock override*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	Specifies the method to use for overriding a locked phone when XML applications are sent to the phone. There are three settings for this parameter: **0** Phone prevents XML POSTs and XML GETs from being received or sent. **1** Phone allows XML POSTs; however, XML GETs by pressing the XML keys (softkeys/programmable keys/extension module keys) are not allowed. **2** Phone allows XML POSTs to the phone as well as XML GETs to/from the phone by pressing the XML keys (softkeys/programmable keys/extension module keys).
Format	Integer
Default Value	0
Range	0 to 2
Example	xml lock override: 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. **Note**: This parameter should be disabled according to M5T.
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

GRUU and sip.instance Support

Parameter – *sip gruu*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	Enables or disables Globally Routable User-Agent URI (GRUU) support on the IP Phone according to draft-ietf-sip-gruu-15. If this parameter is disabled, parsing of inbound GRUU's for transfer are still enabled.
Format	Boolean
Default Value	1 (enabled)
Range	0 (disabled) 1 (enabled)
Example	sip gruu: 0

DNS Query Setting

Parameter– *sip dns query type*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	Specifies the Domain Name Service (DNS) query method to use when the phone performs a DNS lookup.
Format	Integer
Default Value	1
Range	**0** **A only** - The phone sends “A” (Host IP Address) lookup for the IP address and uses the default port number of 5060. **1** **SRV & A** - The phone sends “SRV” (Service Location Record) lookup to get the port number. Most often, the IP address is included in the response from the DNS server to avoid extra queries. If there is no IP address returned in the response, the phone sends out the “A” DNS lookup to find the IP address. **2** **NAPTR & SRV & A** - First, the phone sends "**NAPTR**" (Naming Authority Pointer) lookup to get the “**SRV**” pointer and service type (such as "aastra.com SIP+2DT .... _sip.tcp.aastra.net", which means the service prefers to use TCP and "_sip.tcp.aastra.net" for the SRV query instead of the default "_sip._tcp.aastra.com"). If the NAPTR record is returned empty then the default value is used, so in the same case, the phone will use "_sip._udp.aastra.com" for the next step lookup. Next, the phone does SRV lookup to get the IP address and port number. If there is no IP address in the SRV response then it sends out and “**A**” lookup to get it. **Note:** On the phone side, if you configure the phone with a Fully- Qualified Domain Name (FQDN) proxy and specified port, the phone always sends “A only” lookups to find the Host IP Address of the proxy.
Example	sip dns query type: 2

Ignore Out of Order SIP Requests

Parameter– *sip accept out of order requests*	**Configuration Files**	aastra.cfg, <mac>.cfg
Description	Enables a workaround for non-compliant SIP devices (for example, Asterisk) which do not increment the CSeq numbers in SIP requests sent to the phone.	
Format	Boolean	
Default Value	0 (disabled)	
Range	0 (disabled) 1 (enabled)	
Example	sip accept out of order requests: 1	

Optional “Allow” and “Allow-Event” Headers

Parameter – *sip notify opt headers*	**Configuration Files**	aastra.cfg, <mac>.cfg
Description	Enables and disables whether or not the “Allow” and Allow-Events” optional headers are included in the SIP NOTIFY messages sent from the phone to the server.	
Format	Boolean	
Default Value	1	
Range	0 (disabled - optional headers are removed from the SIP NOTIFY message) 1 (enabled - no change; optional headers are included in SIP NOTIFY message)	
Example	sip notify opt headers: 0	

P-Asserted Identity (PAI)

Parameter – *sip pai*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	Enables or disables whether the SIP PAI displays to the IP phone.
Format	Boolean
Default Value	1
Range	0 (disabled) 1 (enabled)
Example	sip pai: 0

Route Header in SIP Packet

Parameter – *sip remove route*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	Enables and disables the addition of the Route header in a SIP packet. Enable this parameter for outbound proxies that do not support Route headers. **Note:** When enabled this will break all support for SIP routing, so if some other device in the network attempts to add itself to the route it will fail.
Format	Boolean
Default Value	0
Range	0 (disable - adds the Route header to the packet) 1 (enable - removes the Route header from the packet)
Example	sip remove route: 1

Compact SIP Header

Parameter – *sip compact headers*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	Enables or disables the IP phones to use compact SIP headers in the SIP packets sent from the phone.
Format	Boolean
Default Value	0 (disabled- uses long SIP header format)
Range	0 (disabled- uses long SIP header format) 1 (enabled- uses short (compact) SIP header format)
Example	sip compact headers: 1

Rejection of INV or BYE

Parameter – *sip enforce require hdr*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	Enables and disables the the rejection of an INV or BYE with a “420 Bad Extension” if the INV or BYE contains an unsupported value in the REQUIRE header.
Format	Integer
Default Value	0
Range	0 (disable) 1 (enable)
Example	sip enforce require hdr: 1

Configuration Encryption Setting

Parameter– *config encryption key*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	Specifies the phone-specific encryption key that the configuration server uses to encrypt in a MAC-specific configuration file.
Format	String
Default Value	Not applicable
Range	String length of 4 to 32 alphanumeric characters
Example	config encryption key: 123abcd

DNS Host File

Parameter – *sip dns host file*	**Configuration Files**	aastra.cfg, <mac>.cfg
Description	The UNIX-format host file on the configuration server. The phone(s) download this file to perform DNS lookups on the local network instead of the service provider's public network. **Note:** If using a text file on a PC to enter this value, you must enter a carriage return (CR) after entering the host file name.	
Format	UNIX format using Carriage Return (CR) or Carriage Return + Line Feed (CRLF) to terminate each line	
Default Value	Not Applicable	
Range	File name allows Alpha-numeric characters	
Example	sip dns host file: hostfile.txt	

DNS Server Query

Parameter – *sip dns srvX name* **Note**: The "X" indicate a record number with values from 1 to 4.	**Configuration Files**	aastra.cfg, <mac>.cfg
Description	The fully qualified URI of the DNS SRV record	
Format	Fully qualified URI including service prefix	
Default Value	Not Applicable	
Range	Not Applicable	
Example	sip dns srv1 name: _sip._udp.example.com	

Parameter – *sip dns srvX priority* **Note**: The "X" indicate a server number with values from 1 to 4.	**Configuration Files**	aastra.cfg, <mac>.cfg
Description	The priority level assigned to this DNS server. After this parameter is downloaded from the configuration server to the phone, the phone uses the DNS server with the lowest numbered priority first to perform DNS lookups.	
Format	Integer	
Default Value	0	
Range	0 to 65535	

Example	sip dns srv1 priority: 10

Parameter – *sip dns srvX weight* **Note**: The “X” indicate a server number with values from 1 to 4.	**Configuration Files**	aastra.cfg, <mac>.cfg
Description	The weight level assigned to this server. If a service has multiple SRV records with the same priority value, the phones use the weight field to determine which host to use. The weight value is relevant only in relation to other weight values for the service, and only among records with the same priority value. **Note**: The “sip dns srvX weight” parameter must be configured but the phones will support this feature in a future release.	
Format	Integer	
Default Value	0	
Range	0 to 65535	
Example	sip dns srv1 weight: 60	

Parameter – *sip dns srvX port* **Note**: The “X” indicate a server number with values from 1 to 4.	**Configuration Files**	aastra.cfg, <mac>.cfg
Description	The port number on the target host.	
Format	Integer	
Default Value	0	
Range	0 to 65535	
Example	sip dns srv1 port: 5060	

Parameter – *sip dns srvX target* **Note**: The “X” indicate a server number with values from 1 to 4.	**Configuration Files**	aastra.cfg, <mac>.cfg
Description	The host name of the target.	
Format	Host name or fully qualified domain name	
Default Value	Not Applicable	
Range	Not Applicable	
Example	sip dns srv1 target: bigbox.example.com	

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** **Configuration Files**	Advanced Settings->Troubleshooting-> Log Settings 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** **Configuration Files**	Advanced Settings->Troubleshooting-> Log Settings 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	

<table>
<tr><td>Parameters –
log module <module name>

Module/Debug Level
(in Web UI)</td><td>Aastra Web UI Advanced Settings->Troubleshooting->Module/Debug Level
Configuration Files aastra.cfg, <mac>.cfg</td></tr>
<tr><td>Description</td><td>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:

<u>Module Name (configuration files)</u>

• linemgr (line manager information)
• user interface
• misc (miscellaneous)
• sip (call control SIP stack)
• dis (display driver)
• dstore (delayed storage)
• ept (endpoint)
• ind (indicator)
• kbd (keyboard)
• net (network)
• provis (provisioning)
• rtpt (Real Time Transport)
• snd (sound)
• prof (profiler)
• xml (Extension Markup Language)
• stun (Simple Traversal of User Datagram Protocol (UDP) through Network Address Translation (NAT)</td></tr>
<tr><td>Format</td><td>Integer</td></tr>
<tr><td>Default Value</td><td>1 (fatal errors)</td></tr>
<tr><td>Range</td><td><table><tr><th>Debug Level</th><th>Value</th></tr><tr><td>Fatal Errors</td><td>1 (default)</td></tr><tr><td>Errors</td><td>2</td></tr><tr><td>Warnings</td><td>4</td></tr><tr><td>Init</td><td>8</td></tr><tr><td>Functions</td><td>16</td></tr><tr><td>Info</td><td>32</td></tr><tr><td>All debug levels OFF</td><td>0</td></tr><tr><td>All Debug Levels ON</td><td>65535</td></tr></table></td></tr>
</table>

Examples	Enter a debug level value in the "**Debug Level**" field for a module. **Example 1** To turn two or more debug levels on at the same time, you add the value associated with each level. For example, Fatal Errors + Errors + Warnings = 1 + 2 + 4 = 7 log module linemgr: 7 log module user interface: 7 log module sip: 7 In the above example, fatal errors, general errors, and warnings are logged for the line manager, user interface, and SIP call control modules. **Example 2** Functions and Info = 16 + 32 = 48 log module dis: 48 log module net: 48 log module snd: 48 In the above example, functions and general information are logged for the display drivers, network, and sound modules. **Example 3** log module rtpt: 0 log module ind: 65535 In the above example, all debug levels are OFF for the Real Time Transport module. All debug levels are ON for the indicator module. You can set the Module/Debug Levels using the configuration files or the Aastra Web UI.

WatchDog Settings

Parameter– *watchdog enable* *WatchDog* (in Web UI)	**Configuration Files** **Aastra Web UI**	aastra.cfg, <mac>.cfg Advanced Settings->Troubleshooting
Description	Enables/disables the use of the WatchDog task for the IP Phones.	
Format	Boolean	
Default Value	1 (enabled)	
Range	0 (disabled) 1 (enabled)	
Example	watchdog enable: 0	

Crash File Retrieval

Parameter– *upload system info server*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	Specifies the server for which the phone sends the system and crash files (server.cfg, local.cfg, and crash.cfg).
Format	IP address or qualified domain name. Example: tftp://0.0.0.0 ftp://username:password@hostname:port/path
Default Value	Not Applicable
Range	Not Applicable
Example	upload system info server: tftp://132.432.0.43

Parameter– *upload system info manual option*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	Enables and disables the ability to manually upload support information from the IP Phone UI and Aastra Web UI. IP Phone UI: ***Options->Phone Status->Upload System Info*** Aastra Web UI: ***Status->System Information->Support Information*** When this parameter is enabled, an "**Upload System Info**" option displays on the IP Phone UI AND an <**Upload**> button displays on the System Information page in the Aastra Web UI.
Format	Boolean
Default Value	1
Range	0 (disabled) 1 (enabled)
Example	upload system info manual option: 1

Parameter– *upload system info on crash*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	Enables and disables the watchdog to automatically reboot the phone and send a crash file to the pre-defined server.
Format	Boolean
Default Value	0
Range	0 (disabled) 1 (enabled)
Example	upload system info on crash: 1

Appendix B 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:

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 6757i 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 6757i phone.

Table 1:

exten -> 1234,1,Dial(SIP/phone1,20)

This definition completes configuring the 6757i 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.

Table 2:

Name/username	*Host*	*Mask*	*Port*	*Status*
phone1/phone1	*192.168.1.1*	*(D) 255.255.255.255*	*5060*	*Unmonitored*

This completes the basic set-up for the 6757i phone with 1234 extension at the Asterisk IP PBX system. Refer to Asterisk documentation for set-up on extended or advanced features such as voice mail and call forwarding, etc.

Appendix C
Sample Configuration Files

About this Appendix

Introduction

This appendix provides sample configuration files for the 6757i, 6757i CT, and 6753i.

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.

6757i Sample Configuration File

```
# Sample Configuration File
# =========================

# Date: October 20th, 2005

# Phone Model: 6757i

# 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 6757i, 6757iCT, and 6753i 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 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 2:
#
#  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: 6757iaastra


## 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 6757i and 6757iCT
#     L1 to L3 on the 6753i
#
#   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 - Reserved

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
#  6757i or 6757iCT 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
```

6757i CT Sample Configuration File

```
# Sample Configuration File
# =========================

# Date: October 26th, 2005

# Phone Model: 6757iCT

# 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 6757i, 6757iCT, and 6753i 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 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 2:
#
#  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: 6757iaastra
```

```
## 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 6757i and 6757iCT
#     L1 to L3 on the 6753i
#
#   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 - Reserved

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
#  6757i or 6757iCT 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 6757i
#  phone, following are the parameters specific to the 6757i Cordless
#  phones' handset. These parameters can be defined either int the
#  aastra.cfg or the <mac>.cfg files.
#
#  The feature keys are displayed when the user presses the F button
#  on the cordless phone's hand set.  If any changes to the features
#  keys are made using these parameters the feature keys that exist on
#  the hand set have to be refreshed. To refresh the feature keys
#  simply open a new line or press one of the feature keys that are
#  available from the hand set. After a couple of seconds the cordless
#  should get the new list from the base set.  There are 15 feature
#  keys that can be configured for the cordless hand set. Each feature
#  key has the following settings. N corresponds to the feature key
#  that is being configured for and ranges from 0-14.  Feature key N
#  En label: "String" Feature key N Fr label: "Fr-String" Feature key
#  N Sp label: "Sp-String" Feature key N control: 1     #Takes an
#  integer value Feature key N hs event: 1    #Takes an integer value
#  Feature key N base event: 1  #Takes an integer value


#key list version: 1
#  The parameter value has to be incremented by one whenever the
#  parameters that carry the feature keys change. The range is from
#  1-254. After reaching 254 start over from 1.

#Feature key 0 En label: "Line 1"
#  English label for the key. Displayed when the phone's language is
#  set to use English

#Feature key 0 Fr label: "Fr-Line 1"
#  French label for the key. Displayed when the phone's language
#  is set to use French

#Feature key 0 Sp label: "Sp-Line 1"
#  Spanish label for the key. Displayed when the phone's language
#  is set to use Spanish

Feature key 0 Gr label: "Gr-Line 1"
#  German label for the key. Displayed when the phone's language
#  is set to use German
```

```
Feature key 0 It label: "It-Line 1"
#  Italian label for the key. Displayed when the phone's language
#  is set to use Italian

#Feature key 0 control: 1
#  1 - Make the key configurable by the user through the phone and
#      the phone's web client
#  2 - Locks the key from user modifications. User cannot modify
#      this key from the handset or the phone's web client.
#  4 - Hide this key. Do not show it in the Feature keys list in the
#      cordless handset
#  6 - Lock and hide this key. Do not show it in the Feature keys
#      list in the cordless handset and do not let the user modify
#      this key using the phone or the web client.

#Feature key 0 hs event: 7
#  These events are for handset specific events. Events can be local
#  to the handset like directory/caller's list, intercom etc. or may
#  be an event that is sent to the base set for fruther processing.
#  When this key is configured as a base event then the base set
#  will process the value of this key in conjunction with the value
#  configured for the "Feature key N base event". Where N is the
#  feature key is being configured for.
#  In addition to the values listed below the valid values are
#  [7-23]. The values [7-23] indicate generic handset events. If
#  you are using values within this range make sure to use the value
#  only once.
#  The events local to the handset are as follows:
#
#    58 - Menu (Options)
#    59 - Feature Key
#    60 - Redial
#    61 - Directory
#    62 - Callers' list
#    63 - Services
#    86 - Icom

#Feature key 0 base event: 1
#  Indicates a corresponding action to perform on the base set when
#  the "Feature key N hs event" is set to any value between 7-23.
#
#   1 - Seize base set's line1
```

```
#   2 - Seize base set's line2
#   3 - Seize base set's line3
#   4 - Seize base set's line4
#   5 - Seize base set's line5
#   6 - Seize base set's line6
#   7 - Seize base set's line7
#   8 - Seize base set's line8
#   9 - Seize base set's line9
#   10 - Seize base set's line0
#   11 - Send the base set's transfer event
#   12 - Send the base set's conference event
#   13 - Make feature list public


# Example configuration
key list version: 1

Feature key 0 En label: "Line 1"
Feature key 0 Fr label: "Fr-Line 1"
Feature key 0 Sp label: "Sp-Line 1"
Feature key 0 control: 0
Feature key 0 hs event: 7
Feature key 0 base event: 1

Feature key 1 En label: "Conf."
Feature key 1 Fr label: "Fr-Conf."
Feature key 1 Sp label: "Sp-Conf."
Feature key 1 control: 1
Feature key 1 hs event: 8
Feature key 1 base event: 12

Feature key 2 En label: "Xfer"
Feature key 2 Fr label: "Fr-Xfer."
Feature key 2 Sp label: "Sp-Xfer."
Feature key 2 control: 2
Feature key 2 hs event: 9
Feature key 2 base event: 11

Feature key 3 En label: "Icom"
Feature key 3 Fr label: "Fr-Icom"
Feature key 3 Sp label: "Sp-Icom"
Feature key 3 control: 1
```

```
Feature key 3 hs event: 86
Feature key 3 base event: 13

Feature key 4 En label: "Opt"
Feature key 4 Fr label: "Fr-Opt"
Feature key 4 Sp label: "Sp-Opt"
Feature key 4 hs event: 58
Feature key 4 control: 1
Feature key 4 base event: 13

Feature key 5 En label: "Callers"
Feature key 5 Fr label: "Fr-Callers"
Feature key 5 Sp label: "Sp-Callers"
Feature key 5 hs event: 62
Feature key 5 control: 1
Feature key 5 base event: 13


Feature key 6 En label: "Top"
Feature key 6 Fr label: "Fr-Top"
Feature key 6 Sp label: "Sp-Top"
Feature key 6 hs event: 17
Feature key 6 control: 1
Feature key 6 base event: 13

Feature key 7 En label: "Redial"
Feature key 7 Fr label: "Fr-Redial"
Feature key 7 Sp label: "Sp-Redial"
Feature key 7 hs event: 60
Feature key 7 control: 4
Feature key 7 base event: 13

Feature key 8 En label: "Dir."
Feature key 8 Fr label: "Fr-Dir."
Feature key 8 Sp label: "Sp-Dir."
Feature key 8 hs event: 61
Feature key 8 control: 2
Feature key 8 base event: 13

Feature key 9 En label: "Services"
Feature key 9 Fr label: "Fr-Services"
Feature key 9 Sp label: "Sp-Services"
Feature key 9 hs event: 63
```

```
Feature key 9 control: 1
Feature key 9 base event: 13
```

6753i Sample Configuration File

```
# Sample Configuration File
#= = = = = = = = = = = = = = = = = = = = = =

# Date: October 26th, 2005

# Phone Model: 6753i

# 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 6757i, 6757iCT, and 6753i 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 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: 6757iaastra


## 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 6757i and 6757iCT
#     L1 to L3 on the 6753i
#
#   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 - Reserved

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 6753i 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 D
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:

Sample BLF Softkey Settings

Asterisk BLF

The following are sample softkey and programmable key configurations to enable Asterisk BLF support on Aastra IP phones.

6757i and 6757i CT Configuration Parameters for Asterisk BLF

```
softkey1 type: blf
softkey1 value: 9995551212
softkey1 label: John
softkey1 line: 1
```

6753i Configuration Parameters for Asterisk BLF

```
prgkey1 type: blf
prgkey1 value: 9995551212
prgkey1 label: John
prgkey1 line: 1

prgkey7 type: blf
prgkey7 value: 9995551313
prgkey7 label: Jane
prgkey7 line: 1
```

BroadSoft BroadWorks BLF

The following are sample softkey and programmable key configurations to enable Broadsoft BroadWorks Busy Lamp Field support on Aastra IP phones.

6757i and 6757i 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:my6757i-blf-list@as.broadsoft.com
```

6753i 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.

```
prgkey5 type: list
prgkey5 line: 1

prgkey6 type: list
prgkey7 line: 1

list uri: sip:my53i-blf-list@as.broadsoft.com
```

Appendix E
Sample Multiple Proxy Server Configuration

About this Appendix

Introduction

This appendix provides a sample multiple proxy server configuration.

Topics

This appendix covers the following topics:

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.

Note: The phones include support for RFC3327, a SIP extension header called “PATH” for phones to discover intermediate proxies. This feature is always enabled on the phone.

```
#sip settings
sip proxy ip: #.#.#.#
sip proxy port: 5060
sip registrar ip: #.#.#.#
sip registrar port: 5060
sip registration period:3600

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


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
```

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