MODEL

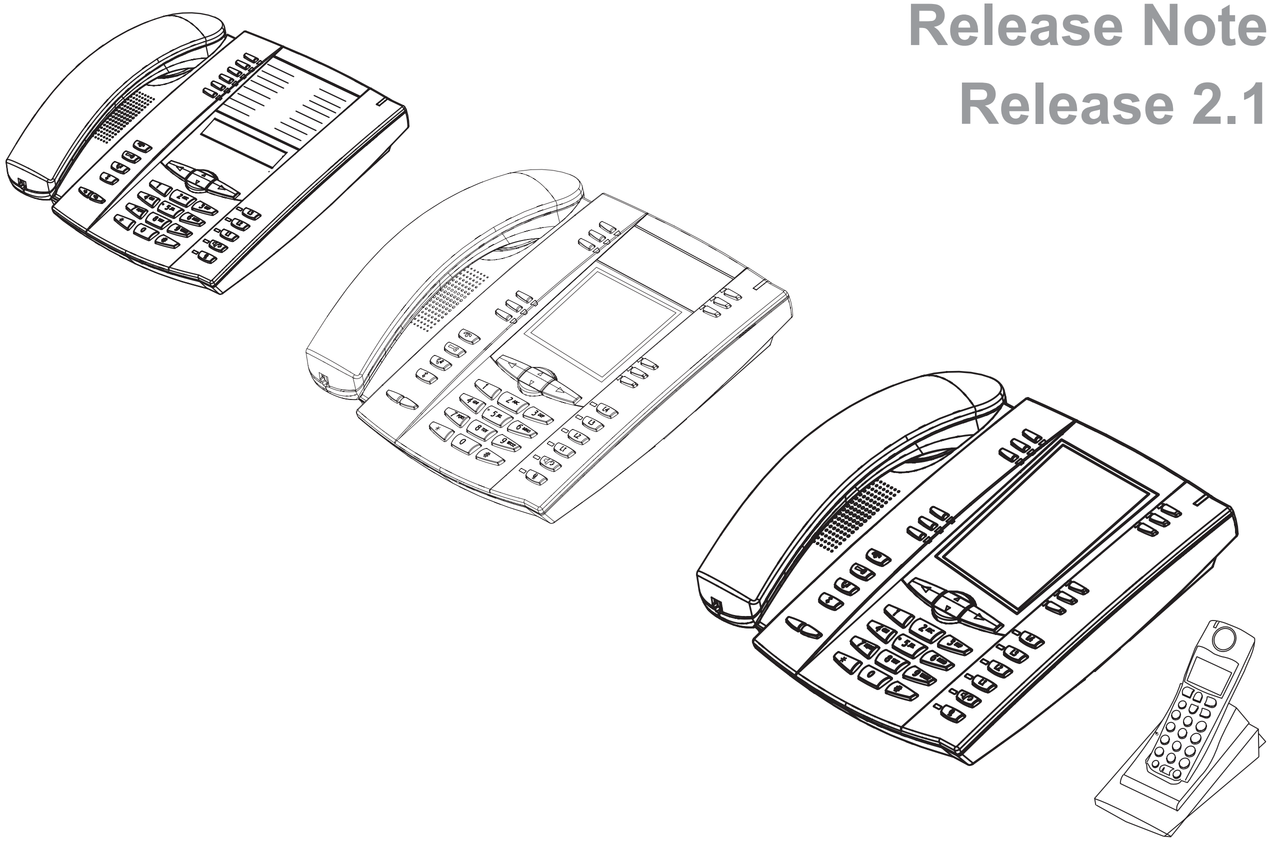
53i, 55i, 57i, and 57i CT

SIP IP PHONE

RN-001029-00
Rev 03

Release Note

Release 2.1



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SIP IP Phone Models 53i, 55i, 57i, and 57i CT Release Note 2.1

About this Document

This document provides an overview of the 2.1 features on the 5i Series phones (53i, 55i, 57i, and 57i CT).

For more detailed information about the features associated with each phone, and for information on how to use the phones, see your model-specific *SIP IP Phone Installation Guide* and the *SIP IP Phone User Guide*. For detailed information about more advanced features, see the *SIP IP Phone Administrator Guide*.

Topics in this release note include:

General Information

Release Content Information

This document provides release content information on the Aastra 53i, 55i, 57i, and 57i CT SIP IP phone firmware.

Model	Release Name	Release Version	Release Filename	Release Date
53i	Generic SIP	2.1	FC-001086-00-04	July 2007
55i	Generic SIP	2.1	FC-001087-00-04	July 2007
57i	Generic SIP	2.1	FC-001088-00-04	July 2007
57i CT	Generic SIP	2.1	FC-001089-00-04	July 2007

Hardware Supported

This release of firmware is compatible with the following Aastra IP portfolio products:

- 53i
- 55i
- 57i
- 57i CT

Bootloader Requirements

This release of firmware is compatible with the following Aastra IP portfolio product bootloader versions:

- 53i - Bootloader 2.0.1.1055 or higher
- 55i - Bootloader 2.0.1.1055 or higher
- 57i - Bootloader 2.0.1.1055 or higher
- 57i CT - Bootloader 2.0.1.1055 or higher

Changes in Release 2.1, Build 2145

Description

This section describes the new features included in Release 2.1 of the 5i Series IP Phones. The following table specifies the 2.1 Features and provides the page number for each feature.

Hardware Support

53i Supports 536M Expansion Module

The 53i IP Phone now supports the optional 536M Expansion Module (previously only supported on the 55i, 57i, and 57i CT). This module adds 36 additional softkeys to the IP phone. The 536M provides paper labels for each softkey. Up to 3 modules can be piggy-backed to provide up to 108 additional softkeys for the phone.

55i Supports 560M Expansion Module

The 55i IP Phone now supports the optional 560M Expansion Module (previously only supported on the 57i and 57i CT). This module adds 60 additional softkeys to the IP phone (using the 3 function keys on the bottom right of the unit). The 560M module provides an LCD display for displaying softkey labels. Up to 3 modules can be piggy-backed to provide up to 180 additional softkeys for the phone.

560M Expansion Module

560M Expansion Module

Usability Features

New Options Menu

For IP Phone Release 2.1, Series 5i and Series 3i phones include a new "Options" menu that has been redesigned to improve navigation. The new Options Menu applies to IP phones operating in SIP mode. As part of the redesign, some menu titles have been added, or been renamed.

Generic SIP Mode Options Menu

For Aastra IP Phones operating in SIP mode, the Options Menu, Submenus, and Options available in Release 2.1 are as follows:

1. Call Forward
- 1. Cfwd Number (53i)
- 2. Cfwd Mode (53i)
- 3. No. Rings (53i)

2. Preferences
- 1. Tones
  - 1. Ring Tone
  - 2. Tone Set
- 2. Display (55i, 57i, 57i CT, Contrast Level for all others)
  - 1. Contrast Level
  - 2. Backlight (55i, 57i, 57i CT)
  - 3. Backlight Time
- 3. Live DialPad
- 4. Set Audio
  - 1. Audio Mode
  - 2. Headset Mic Volume (Headset Mic Vol for 53i)
- 5. Handset Pairing (only for 57i CT)
- 6. Time and Date
  - 1. Time Zone
  - 2. Daylight Savings
  - 3. Time Format
  - 4. Date Format

SIP Mode Options Menu (continued)

5. Time Server
    1. Time Server 1
    2. Time Server 2
    3. Time Server 3
6. Set Time
7. Set Date

7. Language

3. Phone Status

1. IP&MAC Addresses
2. LAN Port
3. PC Port
4. Firmware Info

4. User Password

5. Administrator Menu (Admin Menu for 53i)

1. Configuration Server (Config Server for 53i)

1. Download Protocol

2. TFTP Settings
    1. Primary TFTP
    2. Alternate TFTP
    3. Select TFTP
3. FTP Settings
    1. FTP Server
    2. FTP Username
    3. FTP Password
4. HTTP Settings
    1. HTTP Server
    2. HTTP Path
5. HTTPS Settings
    1. HTTPS Client
        1. Download Server
        2. Download Path
        3. Client Method
    2. HTTPS Server
        1. HTTP->HTTPS
        2. XML HTTP POSTs

SIP Mode Options Menu (continued)

2. Network Settings
- 1. DHCP
- 2. IP Address
- 3. Subnet Mask
- 4. Gateway
- 5. DNS
  - 1. Primary DNS (53i)
  - 2. Secondary DNS (53i)
- 6. NAT Settings
  - 1. Nortel NAT
  - 2. Static NAT
    - 1. NAT IP
    - 2. NAT SIP Port
    - 3. NAT RTP Port
  - 3. UPnP
- 7. VLAN Settings
  - 1. VLAN Enable
  - 2. Phone VLAN
    - 1. Phone VLAN ID
    - 2. VLAN Priority
      - 1. SIP Priority
      - 2. RTP Priority
      - 3. RTCP Priority
      - 4. Other Priority
  - 3. PC Port VLAN
    - 1. PC Port VLAN ID
    - 2. PC Port Priority
- 8. Type of Service DSCP (53i)
  - 1. Type of Service SIP (53i)
  - 2. Type of Service RTP (53i)
  - 3. Type of Service RTCP (53i)
- 9. Ethernet
  - 1. LAN Port Link
  - 2. PC Port Link

SIP Mode Options Menu (continued)

3. SIP Settings

53i IP Phone:

1. Proxy Server
2. Proxy Port
3. Registrar Server
4. Registrar Port
5. SIP Register
6. User Name
7. Display Name
8. Screen Name
9. Authentic. Name
10. Password
11. RTP Port Base

55i, 57i, and 57iCT IP Phones:

1. Proxy IP/Port
2. Registrar IP/Port
3. SIP Register
4. User Name
5. Display Name
6. Screen Name
7. Authentication Name
8. Password
9. RTP Port Base

3. SIP Settings

1. Proxy IP/Port (Proxy Server for 53i)
2. Proxy Port (53i)
3. Registrar IP/Port (Registrar Server for 53i)
4. Registrar Port (53i)
5. SIP Register
6. User Name
7. Display Name
8. Screen Name
9. Authentication Name (Authentic. Name for 53i)
10. Password

11. RTP Port Base

4. Factory Default

5. Erase Local Config. (Erase Local Cfg. for 53i)

6. Restart Phone

7. Phone Lock

IP Phone UI-based Speeddial Keys

IP phone users can now use the IP Phone UI to change an empty softkey, or programmable key to a Speeddial key. In addition, if the IP phone has an Expansion Module attached, you can change unassigned Expansion Module keys to Speeddial keys.

Changing Programmable Keys to Speeddial Keys (53i, 55i)

Use the following procedure to change an empty programmable key on your IP phone to a Speeddial key.

Aastra IP Phone UI

Step	Action
1	Press and hold an unassigned programmable key. After a few seconds, the IP Phone UI prompts you to assign a number for the Speeddial key.
2	Enter the extension/number for the Speeddial key, then press **<Save>**.
3	Select a line number, then press **<Save>**. The programmable key now functions as a Speeddial key. **Note:** Use the Aastra Web UI to edit or delete this speeddial key.

Changing Expansion Module Programmable Keys to Speeddial Keys (53i)

Use the following procedure to change an empty programmable key an Expansion Module to a Speeddial key.

Aastra IP Phone UI	
Step	**Action**
1	Press and hold an unassigned programmable key. After a few seconds, the IP Phone UI prompts you to assign a number for the Speeddial key.
2	Enter the extension/number for the Speeddial key, then press **<Save>**.
3	Select a line number, then press **<Save>**. The programmable key now functions as a Speeddial key. **Note:** Use the Aastra Web UI to edit or delete this speeddial key.

Changing Softkeys to Speeddial Keys (55i, 57i, 57i CT)

Use the following procedure to change an empty softkey (or the More softkey) on your IP phone to a Speeddial key.

Aastra IP Phone UI	
Step	**Action**
1	Press and hold an unassigned softkey, or the **More** softkey. After a few seconds, the IP Phone UI prompts you to assign a name/number to the Speeddial key.
2	Enter the name assigned to the Speeddial key. **Note:** Use the △ and ▽ keys to save your changes, and to move between entry fields.
3	Enter the number assigned to the Speeddial key.

Aastra IP Phone UI	
Step	**Action**
4	Do one of the following actions: • Accept the default line number **1**, or • To select a different line on which to apply the Speeddial key, press the **<Change>** softkey and select a different line, or use the key pad to select a number (1-9), or use the arrow keys to make your selection.
5	Press **<Save>** to save your changes. The softkey key now functions as a Speeddial key. If you look at the IP Phone UI, the Speeddial key displays the name you specified in Step 2. **Note:** Use the Aastra Web UI to edit or delete this speeddial key.

Changing Expansion Module SoftKeys to Speeddial Keys (55i, 57i, 57i CT)

Use the following procedure to change an empty softkey on an Expansion Module to a Speeddial key

Aastra IP Phone UI	
Step	**Action**
1	Press and hold an unassigned softkey on the Expansion Module.
2	Enter the name assigned to the Speeddial key. **Note:** Use the △ and ▽ keys to save your changes, and to move between entry fields.
3	Enter the number assigned to the Speeddial key.

Aastra IP Phone UI

Step	**Action**
4	Do one of the following actions: • Accept the default line number **1**, or • To select a different line on which to apply the Speeddial key, press the **<Change>** softkey and select a different line, or use the key pad to select a number (1-9), or use the arrow keys to make your selection.
5	Press **<Save>** to save your changes. The softkey key now functions as a Speeddial key. **Note:** Use the Aastra Web UI to edit or delete this speeddial key.

Ability to Disable Call Waiting

Currently on the IP phones, the call waiting feature notifies the user currently on the phone, of a new incoming call. Release 2.1 allows a User or Administrator the ability to disable the call waiting feature, so that the new incoming call is automatically rejected by the phone with a busy message.

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

- transfer the currently active call

  or

- accept transferred calls if there is no active calls.

If call waiting is disabled:

- on the 57i CT 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 disable call waiting using the configuration files or the Aastra Web UI.

Enabling/Disabling Call Waiting using the Configuration Files

Use the following parameter to enable or disable call waiting on the phone using the configuration files.

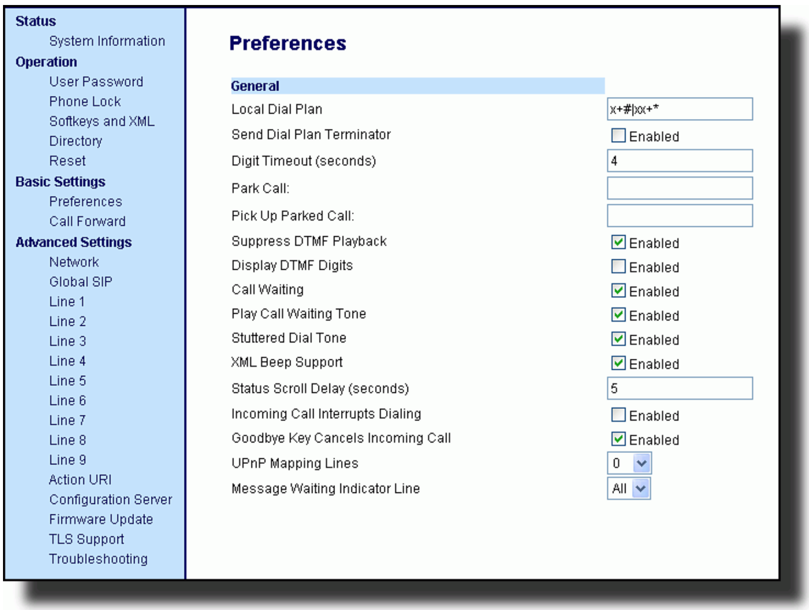
- **`call waiting`**

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 57i CT base, and the handset is currently on a call, all additional incoming calls are rejected on the handset. • intercom calls are treated as regular incoming calls and are rejected. • pre-dialing with live dial pad disabled still accepts incoming calls. • the “Incoming Call Cancels Dialing” parameter is ignored because the incoming call is automatically rejected. • the Missed Calls List does not get updated with details of calls. • the Blind Transfer feature on the phone may not work if two calls are made to the phone at one time.	
Format	Boolean	
Default Value	1 (enabled)	
Range	0 (disabled) 1 (enabled)	
Example	call waiting: 0	

Enabling/Disabling Call Waiting using the Aastra Web UI

Use the following parameter to enable or disable call waiting on the phone using the Aastra Web UI.

Aastra Web UI

1	Click on **Basic Settings-> Preferences->General.** 
2	The "**Call Waiting**" field is enabled by default. To disable this field, uncheck the box. This feature allows you to enable or disable the call waiting feature on the IP phone.
3	Click Save Settings to save your changes.

Mexico Tone Set Added to the IP Phones

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. A new ring tone set for Mexico has been added in Release 2.1.

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

A User and Administrator can configure ring tone sets using the Aastra Web UI and the IP Phone UI. Additionally, an Administrator can configure ring tone sets using the configuration files.

See your IP Phone-specific *User Guide* or the *IP Phone Administrator's Guide* for information about configuring ring tone sets.

Customizable Callers List and Services Keys

The IP phones currently have a Callers List key (all 5i Series phones) and a Services key (55i, 57i, and 57i CT). In Release 2.1, two new parameters have been added that allow you to specify URI overrides for these keys. These parameters are:

- **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

You use the following parameters to customize the Callers List and Services function.

Parameter – *services script*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	Allows you to specify a specific URI for accessing services after pressing the Services key. When this parameter is set, it overrides the standard function of the Services key.
Format	Alphanumeric characters
Default Value	N/A
Range	N/A
Example	services script: http://10.50.100.234/test.xml

Parameter – *callers list script*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	Allows you to specify a specific URI for accessing the Callers List after pressing the Callers List key. When this parameter is set, it overrides the standard function of the Callers List key.
Format	Alphanumeric characters
Default Value	N/A
Range	N/A
Example	callers list script: http://10.50.100.234/test.xml

Configuration Parameters No Longer Case Sensitive

Previously, the configuration parameters an Administrator entered in the configuration files were case sensitive and had to be entered exactly the way they appeared in the *IP Phone Administrator's Guide* (all lowercase).

In Release 2.1, the parameters are no longer case sensitive and can be entered in either upper or lower case as desired.

Deployability Features

Server and Protocol Identification via DHCP Feature

The IP Phones now support additional download protocols according to RFC2131 and RFC1541 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. In addition to supporting the IP address of a TFTP server in this field, the phone now supports a URI format for specifying the other configuration server types.

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

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

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

For more information about setting the download Protocol on the IP phones, see the *SIP IP Phone Administrator Guide*.

Security Features

Transport Layer Security (TLS) Support

The phones now support a new transport protocol called **Transport Layer Security (TLS)** and **Persistent TLS**. TLS is a protocol that ensures communication privacy between the SIP phones and the Internet. TLS ensures that no third party may eavesdrop or tamper with any message.

TLS is composed of two layers: the TLS Record Protocol and the TLS handshake protocol. The TLS Record Protocol provides connection security with some encryption method such as the Data Encryption Standard (DES). The TLS Handshake Protocol allows the server and client to authenticate each other and to negotiate an encryption algorithm and cryptographic keys before data is exchanged. TLS requires the use of specific security certificate files to perform TLS handshake:

- Root and Intermediate Certificates
- Local Certificate
- Private Key
- Trusted Certificate

When the phones use **TLS** to authenticate with the server, each individual call must setup a new TLS connection. This can take more time when placing each call. Thus, the IP phones also have a feature that allows you to setup the connection to the server once and re-use that one connection for all calls from the phone. It is called **Persistent TLS**. The setup connection for Persistent TLS is established during the registration of the phone. If the phones are set to use Persistent TLS, and a call is made from the phone, this call and all subsequent calls use the same authenticated connection. This significantly reduces the delay time when placing a call.

Notes:

1. Persistent TLS requires the **outbound proxy server** and **outbound proxy port** parameters be configured in either the configuration files or the Aastra Web UI (*Advanced Settings->Global SIP->Basic SIP Network Settings*). There can be only one persistent TLS connection created per phone. The phone establishes the TLS connection to the configured outbound proxy.

2. If you configure the phone to use Persistent TLS, you must also specify the Trusted Certificate file to use. The Root and Intermediate Certificates, Local Certificate, and Private Key files are optional.

On the IP phones, an Administrator can configure TLS and Persistent TLS on a global-basis only, using the configuration files or the Aastra Web UI.

Configuring TLS Using 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**

Parameter – *sip transport protocol* *Transport Protocol* (in Web UI)	**Aastra Web UI** Advanced Settings->Global SIP-> Advanced SIP Settings **Configuration Files** 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.
Format	Integer
Default Value	1 - UDP
Range	Valid values are: 0 - User Datagram Protocol (UDP) and Transmission Control Protocol (TCP) 1 - UDP 2 - TCP 4- Transport Layer Security (TLS)
Example	sip transport protocol: 4

Parameter – *sips persistent tls*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	Enables or disables the use of Persistent Transport Layer Security (TLS). Persistent TLS sets up the connection to the server once and re-uses that connection for all calls from the phone. The setup connection for Persistent TLS is established during the registration of the phone. If the phones are set to use Persistent TLS, and a call is made from the phone, this call and all subsequent calls use the same authenticated connection. This significantly reduces the delay time when placing a call. **Notes:** **1.** Persistent TLS requires the **outbound proxy server** and **outbound proxy port** parameters be configured in either the configuration files or the Aastra Web UI (*Advanced Settings->Global SIP->Basic SIP Network Settings*). There can be only one persistent TLS connection created per phone. The phone establishes the TLS connection to the configured outbound proxy. **2.** If you configure the phone to use Persistent TLS, you must also specify the **Trusted Certificate** file to use. The Root and Intermediate Certificates, Local Certificate, and Private Key files are optional.
Format	Boolean
Default Value	0 (disabled)
Range	0 (disabled) 1 (enabled)
Example	sips persistent tls: 1

Parameter – *sips root and intermediate certificates* *Root and Intermediate Certificates* (in Web UI)	**Aastra Web UI** **Configuration Files**	Advanced Settings->TLS Support aastra.cfg, <mac>.cfg
Description	Allows you to specify the SIP Root and Intermediate Certificate files to use when the phone uses the TLS transport protocol to setup a call. The Root and Intermediate Certificate files contain one root certificate and zero or more intermediate certificates which must be placed in order of certificate signing with root certificate being the first in the file. If the local certificate is signed by some well known certificate authority, then that authority provides the user with the Root and Intermediate Certificate files (most likely just CA root certificate). This parameter is required when configuring TLS (optional for Persistent TLS.) **Note:** The certificate files must use the format “.pem”. To create custom certificate files to use on your IP phone, contact Aastra Technical Support.	
Format	<file name>.pem	
Default Value	N/A	
Range	N/A	
Example	sips root and intermediate certificates: cacert_openser.pem	

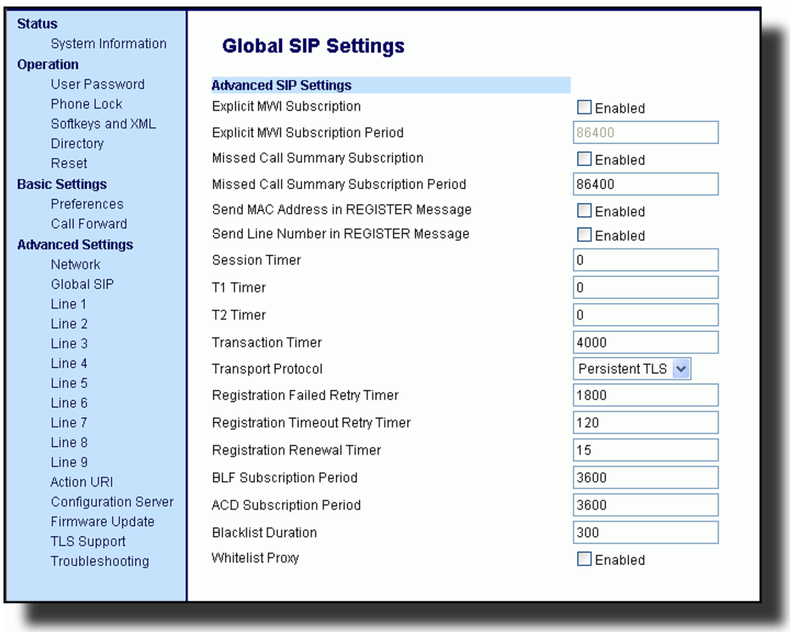
Parameter – *sips local certificate* *Local Certificate* (in Web UI)	**Aastra Web UI** **Configuration Files**	Advanced Settings->TLS Support aastra.cfg, <mac>.cfg
Description	Allows you to specify the Local Certificate file to use when the phone uses the TLS transport protocol to setup a call. This parameter is required when configuring TLS (optional for Persistent TLS.) **Note:** The certificate file must use the format ".pem". To create specific certificate files to use on your IP phone, contact Aastra Technical Support.	
Format	<file name>.pem	
Default Value	N/A	
Range	N/A	
Example	sips local certificate: phonesLocalCert.pem	

Parameter – *sips private key* *Private Key* (in Web UI)	**Aastra Web UI** **Configuration Files**	Advanced Settings->TLS Support aastra.cfg, <mac>.cfg
Description	Allows you to specify a Private Key file to use when the phone uses the TLS transport protocol to setup a call. This parameter is required when configuring TLS (optional for Persistent TLS.) **Note:** The key file must use the format ".pem". To create specific private key files to use on your IP phone, contact Aastra Technical Support.	
Format	<file name>.pem	
Default Value	N/A	
Range	N/A	
Example	sips private key: phone-privkey.pem	

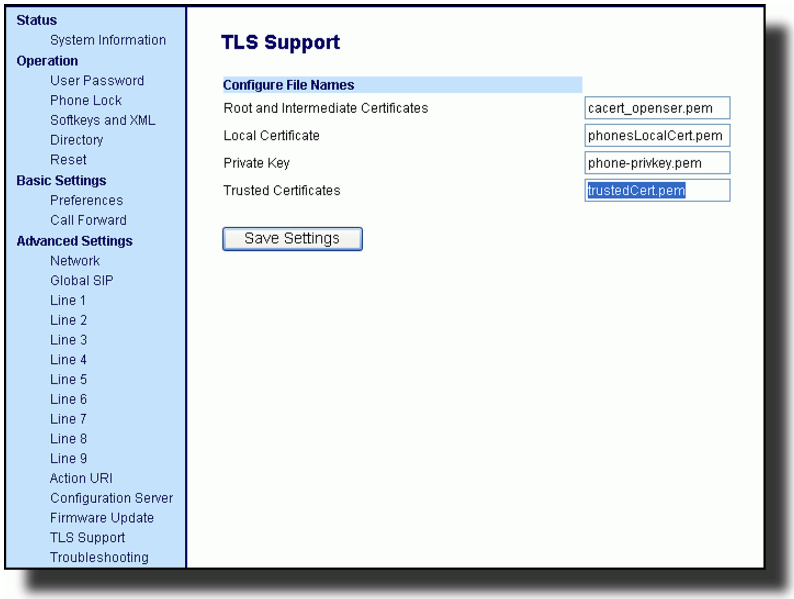
Parameter – *sips trusted certificates* *Trusted Certificates* (in Web UI)	**Aastra Web UI** **Configuration Files**	Advanced Settings->TLS Support aastra.cfg, <mac>.cfg
Description	Allows you to specify the Trusted Certificate files to use when the phone uses the TLS transport protocol to setup a call. The Trusted Certificate files define a list of trusted certificates. The phone's trusted list must contain the CA root certificates for all the servers it is connecting to. For example, if the phone is connecting to server A which has a certificate signed by CA1, and server B which has a certificate signed by CA2, the phone must have CA1 root certificate and CA2 root certificate in its Trusted Certificate file. This parameter is required when configuring TLS or Persistent TLS. **Note:** The certificate files must use the format ".pem". To create custom certificate files to use on your IP phone, contact Aastra Technical Support.	
Format	<file name>.pem	
Default Value	N/A	
Range	N/A	
Example	sips trusted certificates: trustedCert.pem	

Configuring TLS Using the Aastra Web UI

To configure TLS using the Aastra Web UI, you must enable TLS or Persistent TLS first. Then you must define the TLS certificate file names that you want the phone to use. Use the following procedure to configure TLS using the Aastra Web UI.

	Aastra Web UI
1	Click on **Advanced Settings->Global SIP->Advanced SIP Settings**. 
2	In the "**Transport Protocol**" field, select **TLS** or **Persistent TLS**. **Note:** If configuring **Persistent TLS**, you must go to *Advanced Settings->Global SIP->Basic Network Settings* and configure the "**Outbound Proxy Server**" and "**Outbound Proxy Port**" parameters.
3	Click Save Settings to save your changes.

	Aastra Web UI
4	Click on **Advanced Settings->TLS Support.**



Aastra Web UI	
5	Enter the certificate file names and the private key file name in the appropriate fields. The Root and Intermediate Certificate files contain one root certificate and zero or more intermediate certificates which must be placed in order of certificate signing with root certificate being the first in the file. If the local certificate is signed by some well known certificate authority, then that authority provides the user with the Root and Intermediate Certificate files (most likely just CA root certificate). The Trusted Certificate files define a list of trusted certificates. The phone's trusted list must contain the CA root certificates for all the servers it is connecting to. For example, if the phone is connecting to server A which has a certificate signed by CA1, and server B which has a certificate signed by CA2, the phone must have CA1 root certificate and CS2 root certificate in its Trusted Certificate file. **Notes**: **1.** If configuring TLS, you must specify the files for Root and Intermediate Certificates, the Local Certificate, the Private Key, and the Trusted Certificates in order for the phone to receive calls. **2.** If configuring Persistent TLS, you must specify the Trusted Certificates (which contains the trusted certificate list). All other certificates and the Private Key are optional. **3.** The certificate files and Private Key file names must use the format ".pem". **4.** To create custom certificate files and private key files to use on your IP phone, contact Aastra Technical Support.
6	Click Save Settings to save your changes.

Secure Real-Time Transfer Protocol (SRTP) Support with SDES Key Exchange

Release 2.1 includes 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 (default): 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 Preferred: 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 Only: IP phone generates and accepts RTP secured calls only; all other calls are rejected (fail).

An Administrator 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**.

An Administrator can configure SRTP on a global or per-line basis using the configuration files and the Aastra Web UI.

Configuring SRTP Using Configuration Files

You use the following parameters to configure SRTP in the configuration files:

Global Parameter

- **`sip srtp mode`**

Per-Line Parameter

- **`sip lineN srtp mode`**

Global Parameter

Parameter – *sip srtp mode*	**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 (disabled)
Range	0 1 2
Example	sip srtp mode: 1

Per-Line Parameter

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

Configuring SRTP Using the Aastra Web UI

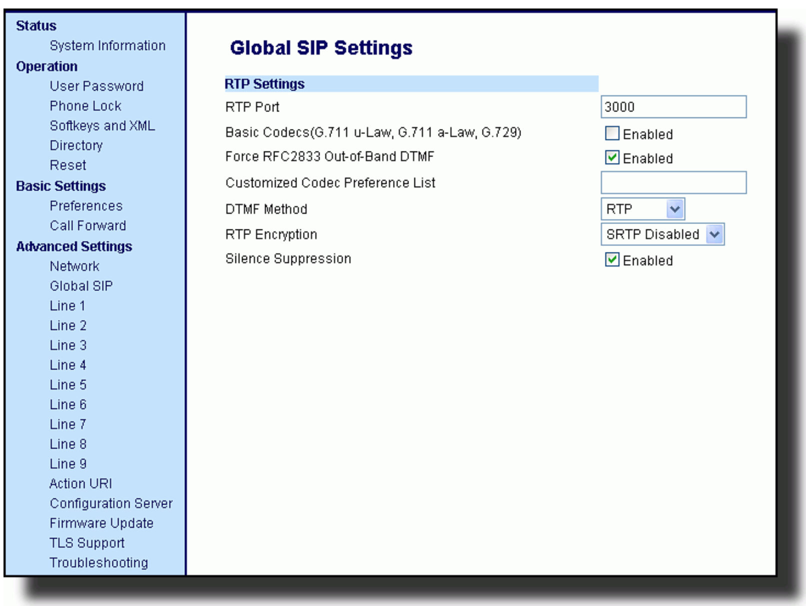
When you configure SRTP using the Aastra Web UI, you must first globally configure SRTP support for the IP phone. Then, if you wish, you can configure SRTP support on a per-line basis.

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

Aastra Web UI

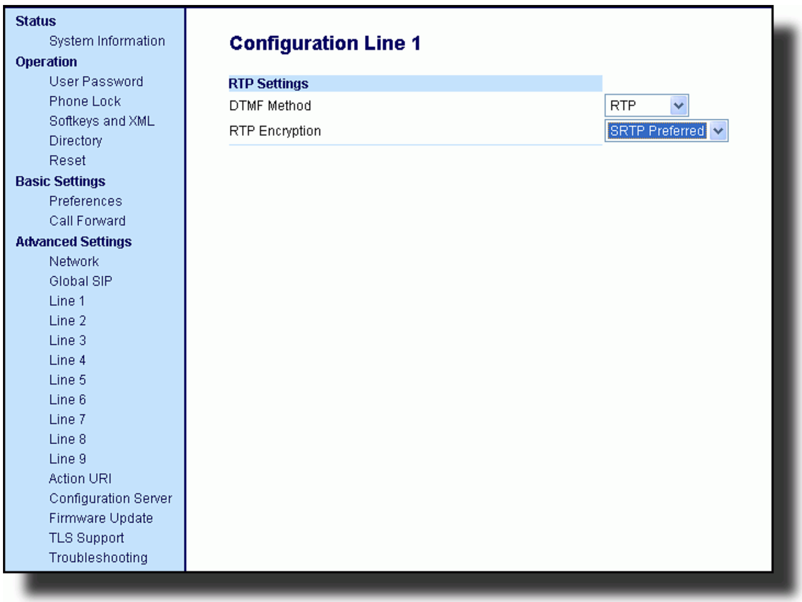
Global Configuration

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



2 Set the **“RTP Encryption”** field to one of the following settings:

- **SRTP Disabled**: IP phone generates and receives nonsecured RTP calls.
- **SRTP Preferred**: IP phones generates RTP secured calls, and accepts both secured and non-secured RTP calls.
- **SRTP Only:** IP phones generates and accepts RTP secured calls only; all other calls are rejected.

	Aastra Web UI
3	Click Save Settings to save your changes.
Per-Line Configuration	
1	**Note:** Setting a per-line SRTP configuration overrides the global SRTP configuration setting. Click on **Advanced Settings->Line <1-9>->RTP Settings**. 
2	Set the **“RTP Encryption”** field to one of the following settings: • **SRTP Disabled**: IP phone generates and receives nonsecured RTP calls on this line. • **SRTP Preferred**: IP phones generates RTP secured calls, and accepts both secured and non-secured RTP calls on this line. • **SRTP Only:** IP phones generates and accepts RTP secured calls only on this line; all other calls are rejected.
3	Click Save Settings to save your changes.

XML Features

New “doneAction” Attribute for XML Text Screen Object

You use the **AastraIPPhoneTextScreen** object to display text to the LCD screen on the IP Phone. The screen text wraps appropriately and can scroll to display a message longer then four lines.

After implementing this object, text displays to the LCD on the IP phone. A user can scroll through the screens as required. If you use the “**destroyOnExit**” attribute in the XML script, when the user exits the XML screens, the screens are destroyed.

A new feature in Release 2.1 allows specific text screens to redisplay for redirection to a new page by using the “**doneAction**” attribute and specifying the new page to go to in the XML script.

Notes:
1. You can use the “destroyOnExit” attribute with any XML object as required.
2. You can use the “**doneAction**” attribute with the **AastraIPPhoneTextScreen** and **AastraIPPhoneFormattedTextScreen** objects only**.**
3. For all available parameters you can use for the Text Screen object, and for an explanation of each parameter, see Aastra Telecom’s "*XML Developer’s Guide*".

Implementation

The following is how you would implement the Text Screen object.

***Softkey*:**

- 6=Done

***XML Description*:**

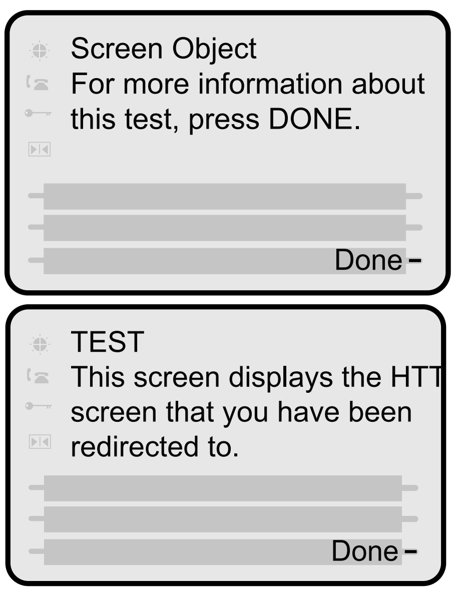
```
<AastraIPPhoneTextScreen doneAction="HTTP address">
   <Title>Screen Title</Title>
   <Text> The screen text goes here</Text>
</AastraIPPhoneTextScreen>
```

***XML Example*:**

```
<AastraIPPhoneTextScreen doneAction="http://10.50.10.117/
test.xml">
   <Title>Screen Object</Title>
   <Text>For more information about this test, press Done.</Text>
</AastraIPPhoneTextScreen>
```

Note: This example displays text that you can scroll through on the LCD screen. As you scroll the screen, and then press **DONE** (55i, 57i, 57i CT) or the **RIGHT ARROW** key (53i), the screen redirects you to the location specified in the script. After pressing **DONE** or the **RIGHT ARROW** key, the phone checks if a "**doneAction**" exists in the XML script. If it does, the screen gets redirected to the location specified. If it does not exist, then the scrolled screens use the "**destroyOnExit**" attribute and destroy the screens.

***XML Screen Example*:**



XML Support for Answer and Ignore Softkeys

In Release 2.1, when the IP phone receives an XML application (either via a post or an incoming action URI) while a call is coming into the phone, the user can either answer or ignore the call with new softkeys that display (55i, 57i, and 57i CT), or press the left and right arrow keys (53i), without canceling the XML application.

For a 55i, 57i, and 57i CT, an Administrator can use the "**Answer**" and "**Ignore**" attributes in an XML script to implement this feature. For a 53i, an Administrator can use the "**allowAnswer**" attribute with the **AastraIPPhoneTextScreen** XML object. Valid values for the "allowAnswer" attribute are "yes" or "no" (default).

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

- The **Answer** and **Ignore** softkeys display on the LCD when the phone has an incoming call at the same time it receives an XML application.
- XML applications are destroyed if the phone receives a call after the XML has been rendered.

When the **Answer** softkey displays, you can press it to answer the incoming call without disturbing the current XML application. When you answer the call, the softkey disappears from the LCD. Pressing the **Ignore** softkey ignores the incoming call without disturbing the current XML application.

Implementation (55i, 57i, 57i CT)

***<u>Softkeys</u>*:**

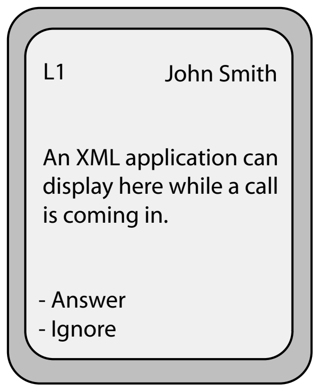
- 1=Answer
- 2=Ignore

***XML Example*:**

```
<SoftKey index="1">
   <Label>Answer</Label>
   <URI>SoftKey:Answer</URI>
</SoftKey>

<SoftKey index="2">
   <Label>Answer</Label>
   <URI>SoftKey:Ignore</URI>
</SoftKey>
```

***XML Screen Example*:**



For 53i:

- An **<Ignore Answer>** line displays on the LCD when the phone receives an incoming call at the same time it receives an XML application.
- XML applications are destroyed if the phone receives a call after the XML has been rendered.

When the **<Ignore Answer>** line displays, you can press the **Right Arrow** key (Answer) to answer the incoming call without disturbing the current XML application. When you answer the call, the <Ignore Answer> line disappears from the LCD. Pressing the **Left Arrow** key ignores the incoming call without disturbing the current XML application.

Implementation (53i)

***XML Example*:**

```
<AastraIPPhoneTextScreen allowAnswer = "yes">
   <Title>Screen Object</Title>
   <Text>The screen object can be implemented similar to the
firmware info screen. Note that white space is preserved in XML so
the display should word-wrap appropriately. Only three lines can
display at a time.</Text>
</AastraIPPhoneTextScreen>
```

***XML Screen Example*:**

XML Softkey for Special Characters (55i, 57i, and 57i CT only)

In Release 2.1, the IP Phone can dynamically receive a Symbol List when it receives the **AastraIPPhoneInputScreen** XML object. You can have a single symbol specified for the softkey, or you can have a list of symbols. When there is only one symbol in the list, the symbol displays with no delay. When there is a list of symbols, you can keep pressing the symbol softkey to cycle through the list of symbols to select the one you want to use.

To display a list of customized symbols to the phone's softkey, the server must include the list of characters in the URI field of the XML softkey script. The URI must be in the format:

```
SymbolList=″<Symbol List content>″
```

The content of the Symbol List must be encapsulated by quotes. You can specify multiple symbols in one URI. For example, the **SymbolList="@#"** specifies the @ and # symbols.

Note: You can have multiple Symbol List softkeys with different lists of symbols. The maximum length of the data in a Symbol List is 230 characters.

There are some special characters that needed to be encoded due to XML limitations. The following table specifies these characters.

Symbol	XML Encoding
single quote (‘)	'
double quote (“)	"
greater-than sign (>)	>
less-than sign (<)	<
ampersand (&)	&

The following is an example XML URI using the characters in the table above:

```
SymbolList="@#&amp;&gt;&lt"
```

The Symbol List content for this URI is @, #, &, >, <.

Implementation

Softkeys:

- 1 = <Single Symbol or Symbol List>

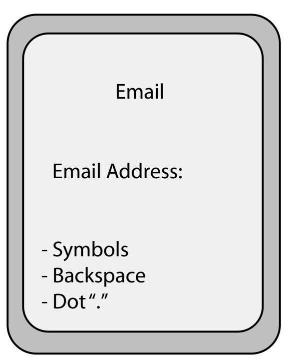
XML Softkey Example:

```
<SoftKey index="1">
   <Label>Symbols</Label>
   <URI>SoftKey:SymbolList="@#=&amp;"</URI>
</SoftKey>
```

XML Object and Softkey Example:

```
<AastraIPPhoneInputScreen type = "IP">
   <Title>Email</Title>
   <SoftKey index="1">
   <Label>Symbols</Label>
   <URI>SoftKey:SymbolList="@"</URI>
   </SoftKey>
   <SoftKey index = "2">
      <Label> Backspace </Label>
      <URI>SoftKey:Exit</URI>
   </Softkey>
   <SoftKey index = "3">
      <Label> Dot </Label>
      <URI>SoftKey:Exit</URI>
   </Softkey>
<Prompt>Email Address:</Prompt>
   <URL>http://myserver.com/myscript.com</URL>
   <Parameter>email</Parameter>
   <Default></Default>
<AastraIPPhoneInputScreen>
```

XML Screen Example:



Sylantro Interoperability Features

Multi-Stage Digit Collection (Billing Codes) Support for Sylantro Servers

This release of the Aastra IP Phones supports Multi-Stage Digit Collection (billing codes) for Sylantro Servers. Sylantro Server features, like mandatory and optional billing codes, requires that the application server notify the phone to collect more digits before completing the call. The IP phone is able to collect digits in two stages to support the billing code feature.

Aastra IP Phone users are prompted to enter the correct billing code when they dial these numbers:

- External numbers.
- Eternal numbers dialed using a Speeddial key.

Billing Codes Implementation Notes

Note the following implementation information:

- IP phone users may enter a 2-9 digit billing code. Billing codes may not start with either 0 (Operator) or 9 (external calls).
- When using Sylantro Click-to-Call, IP phone users select a billing code from a pull-down menu.
- When placing a call, a secondary dial tone alerts IP phone users to enter the billing code. The IP phone UI also displays a “Enter Billing Code” message.
- If an IP phone user redials a number, they do not have to re-enter the billing code. The billing code information is maintained and processed accordingly.
- If an IP phone user enters an invalid billing code, the call fails.

Mandatory versus Optional Billing Codes

This release of the Aastra IP phones supports two types of billing codes: Mandatory and Optional. The Sylantro server configuration determines which type of billing code is used on the IP phones.

- **Mandatory billing codes**: Calls are not connected until the user enters a valid billing code. The user dials the phone number. When prompted for billing codes, user dials the billing code.

  For example, suppose the IP phone user is using billing code 300, and dialing the external number 617-238-5500. The IP user then enters the number using the following format:

  6172385000#300

  Using mandatory billing codes, if the user is configuring a Speeddial number, then they enter the number using the following format:

  <phonenumber>%23<billingcode>

  To use this format with the default dial plan terminator (#), the # sign required by Sylantro as a delimiter should be represented as an escaped character by using the sequence %23. The speed dial format for an external number that includes a mandatory billing code becomes:

  <phonenumber>%23<billing code>

- **Optional billing codes:** The user dials an optional billing code by dialing *50, followed by the billing code digits. When prompted for additional digits, user enters the phone number.

  For example, suppose the IP phone user is using billing code 500, and dialing the external number 617-238-5000. The IP user then enters the number using the following format:

  ***50500#6172385000**

  If the user is dialing configuring a Speeddial number, then they enter the number using the following format:

  ***50<billingcode>#<phonenumber>**

To use this format with the default dial plan terminator (#), the # sign required by Sylantro as a delimiter should be represented as an escaped character by using the sequence %23. The speed dial format for an external number that includes an optional billing code becomes:

***50<billing code>%23<phone number>**

Numbers Not Requiring Billing Codes

Billing codes are not required for the following two types of calls:

- Emergency calls (E911)
- Calls between extensions

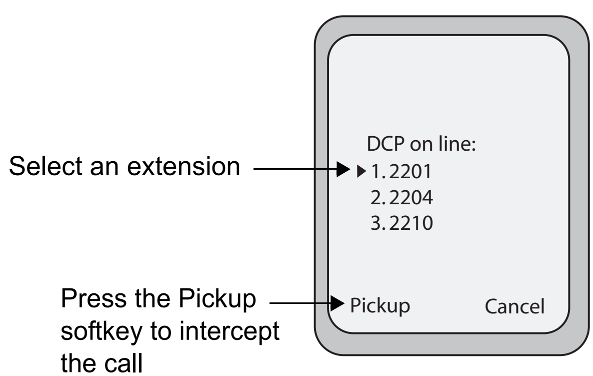
Directed Call Pickup/Group Call Pickup Support for Sylantro Servers

Aastra IP phones now support the Sylantro Directed Call Pickup (DCP) and Group Call Pickup (GCP) features.

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

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

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



Configuring DCP/GCP Using Configuration Files

You use the following parameter to configure DCP and GCP in the configuration files:

- `dcp`

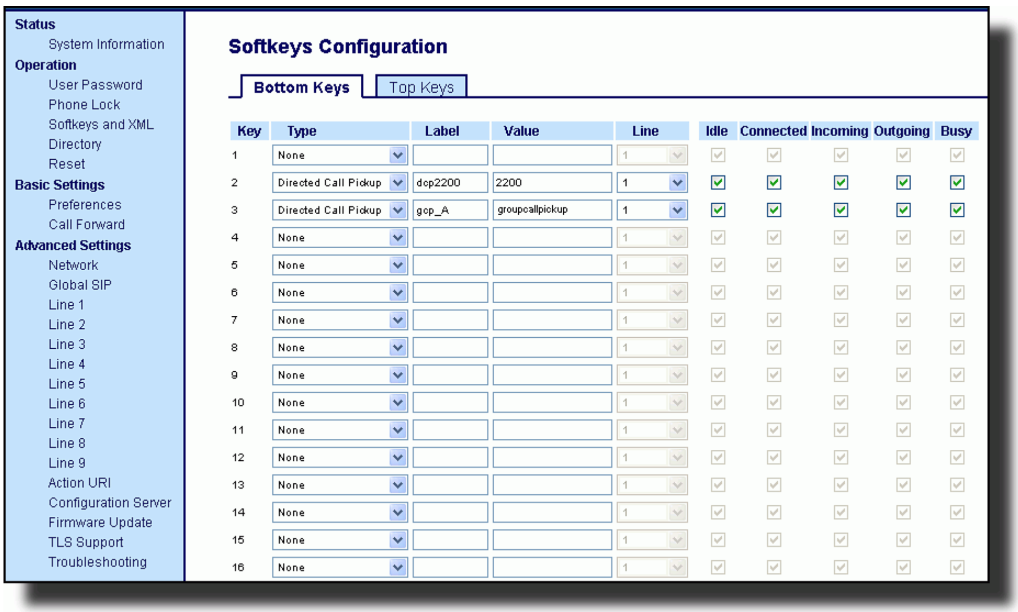
Parameter – *dcp* *Directed Call Pickup* (in Web UI)	**Aastra Web UI** **Configuration Files**	Operation->Softkeys and XML->Type Operation->Programmable Keys->Type Operation->Expansion Module N->Type aastra.cfg, <mac>.cfg
Description	Configures a new softkey that functions as a Directed Call Pickup (DCP) or Group Call Pickup (GCP) key.	
Format	Text	
Default Value	N/A	
Range	N/A	
Examples	**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	

Configuring Directed Call Pickup (DCP) Using the Aastra Web UI

Use the following procedure to configure Directed Call Pickup using the Aastra Web UI.

Aastra Web UI

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



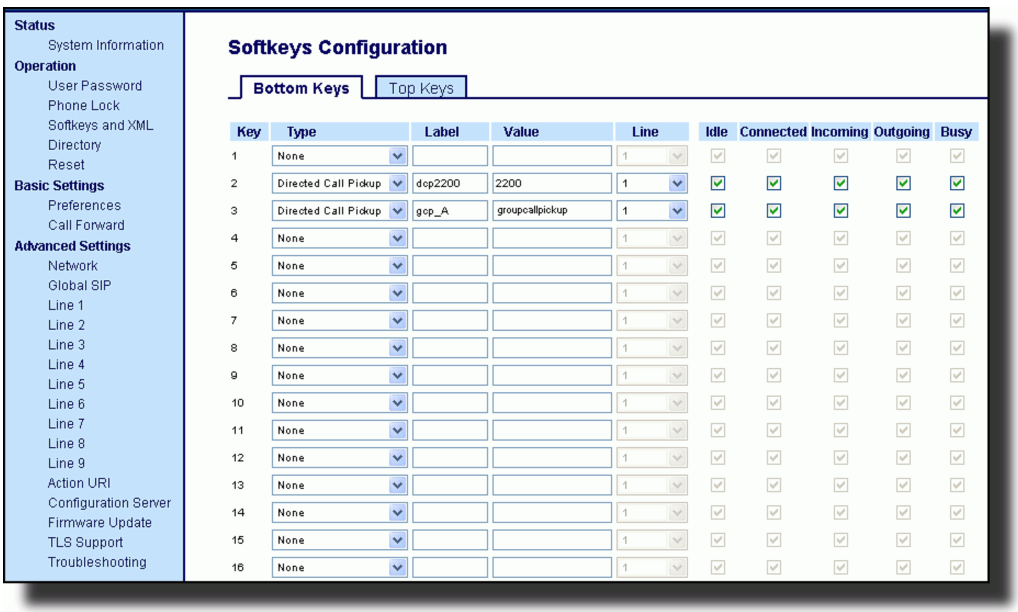
2	Select a key for which to configure Directed Call Pickup.
3	In the "**Type**" field, select **Directed Call Pickup**.
4	In the **"Label"** field, specify a name for this Directed Call Pickup softkey. For example: **DCP2200**
5	In the **"Value"** field, specify the extension you want to intercept when 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

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**.



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.

Autodial (Hotline and Warmline) Feature

This release of the Aastra IP phones includes an "Autodial" feature. 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: 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: Please read the following important information before configuring Autodial on your IP phone:

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

Configuring AutoDial Support Using Configuration Files

You use the following parameters to configure Autodial support in the configuration files:

Global Configuration

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

Per-Line Configuration

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

AutoDial Global Configuration

Parameter – *sip autodial number* *Autodial Number* (in Web UI)	**Aastra Web UI** **Configuration Files**	Advanced Settings>Global SIP> Autodial Settings 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** **Configuration Files**	Advanced Settings>Global SIP> Autodial Settings aastra.cfg, <mac>.cfg
Description	Globally specifies the time, in seconds, that the phone waits to dial a preconfigured number after the handset is lifted from the IP phone cradle. If this parameter is set to 0 (hotline), the phone immediately dials a preconfigured number when you lift the handset. If this parameter is set to a value greater than 0, the phone waits the specified number of seconds before dialing the preconfigured number (warmline) when you lift the handset. Default is 0 (hotline).	
Format	Integer	
Default Value	0	
Range	0 to 120	
Examples	sip autodial timeout: 30	

AutoDial Per-Line Configuration

Parameter – *sip lineN autodial number* *Autodial Number* (in Web UI)	**Aastra Web UI** **Configuration Files**	Advanced Settings>LineN>Autodial Settings aastra.cfg, <mac>.cfg
Description	On a per-line basis, this parameter specifies the SIP phone number that the IP phone autodials when the handset is lifted from the phone cradle. Valid values can be: -1 — (Default) The phone uses the global autodial setting for this line. Blank — (Empty field) Disables autodial on this line. Valid SIP Number — Dials the SIP number specified for this line.	
Format	Integer	
Default Value	-1	
Range	Any valid SIP number.	
Examples	sip line1 autodial number: 8500	

Parameter – *sip lineN autodial timeout* *AutoDial Timeout* (in Web UI)	**Aastra Web UI** Advanced Settings>LineN>Autodial Settings **Configuration Files** aastra.cfg, <mac>.cfg
Description	On a per-line basis, this parameter specifies the time, in seconds, that the phone waits to dial a preconfigured number after the handset is lifted from the IP phone cradle. If this parameter is set to 0 (hotline), the phone immediately dials a preconfigured number when you lift the handset. If this parameter is set to a value greater than 0, the phone waits the specified number of seconds before dialing the preconfigured number (warmline) when you lift the handset. Default is 0 (hotline).
Format	Integer
Default Value	0
Range	0 to 120
Examples	sip line1 autodial timeout: 30

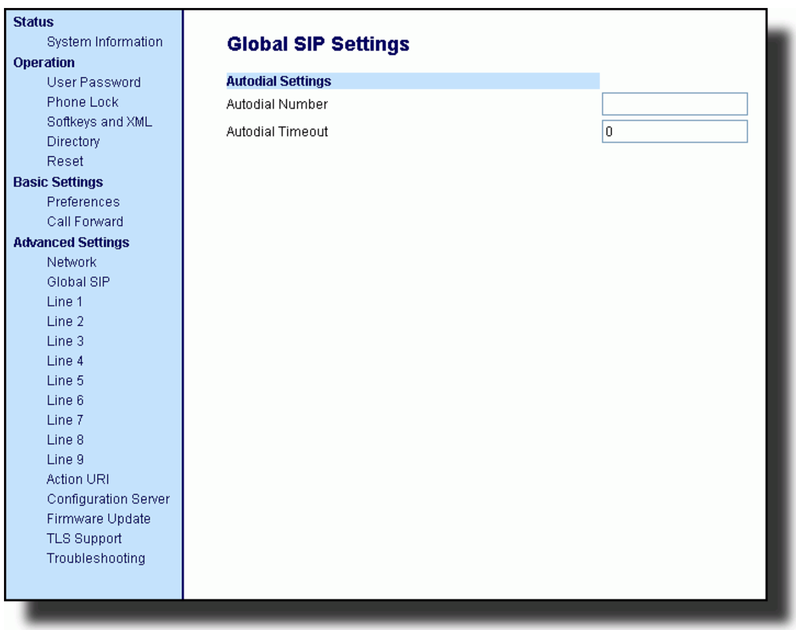
Configuring Autodial Using the Aastra Web UI

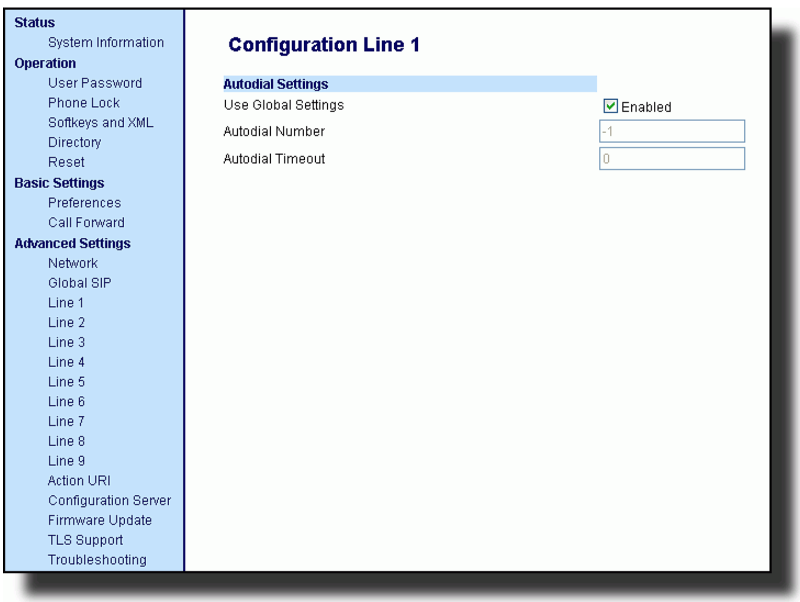
Use the following procedure to configure the Autodial feature on an IP phone using the Aastra Web UI.

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. For example: **8500**

Aastra Web UI	
3	In the **“Autodial Timeout”** field, specify a value 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**
4	Click Save Settings to save your changes.
Per-Line Configuration	
1	Click on **Advanced Settings->Line <1 - 9>->Autodial Settings.** 
2	Do one of the following actions: • To allow this line to use the global autodial settings, click on the **Use Global Settings** parameter to enable it, then click Save Settings to save your changes. • To specify a different autodial configuration for this specific line, disable the **Use Global Settings** parameter. Then proceed to step 3.

	Aastra Web UI
3	In the "**Autodial Number**" field, specify the SIP number for this line that the IP phone dials whenever the IP phone is off-hook as follows: • If set to -1, then the global autodial settings for this IP phone to this line. • If set to empty (blank), then disable Autodial on this line. • If set to a valid SIP number, dial the SIP number specified for this line. For example: **8500**
4	In the **"Autodial Timeout"** field, specify a value 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**
5	Click Save Settings to save your changes.

Centralized Conferencing for Sylantro Servers

Release 2.1 of the Aastra IP phones includes 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 centralized conferences, see your Phone-specific User Guide.

Configuring Centralized Conferencing Using the Configuration Files

You use the following parameter to configure centralized conferencing in the configuration files:

Global Parameter

- `sip centralized conf`

Per-Line Parameter

- `sip lineN centralized conf`

<table>
<tr><td>Parameter –
sip centralized conf

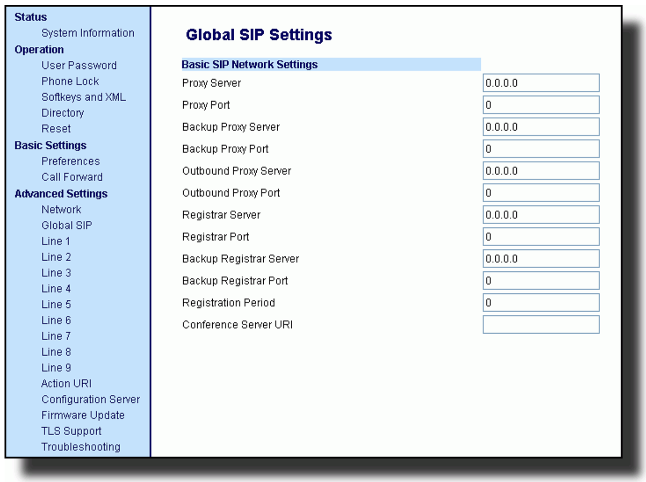
Conference Server URI
(in Web UI)</td><td>Aastra Web UI Advanced->Global SIP Settings-> Basic SIP Network Settings
Configuration Files aastra.cfg, <mac>.cfg</td></tr>
<tr><td>Description</td><td>Globally enables or disables SIP centralized conferencing for an IP phone as follows:
• To disable centralized 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></td></tr>
<tr><td>Format</td><td>String</td></tr>
<tr><td>Default Value</td><td>Blank</td></tr>
<tr><td>Example</td><td>sip centralized conf: conf</td></tr>
</table>

Parameter – *sip lineN centralized conf* *Conference Server URI* (in Web UI)	**Aastra Web UI** **Configuration Files**	Advanced->Line <1 thru 9>-> Basic SIP Network Settings 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	

Configuring Centralized Conferencing Using the Aastra Web UI

Use the following procedure to configure centralized conferencing on an IP phone.

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

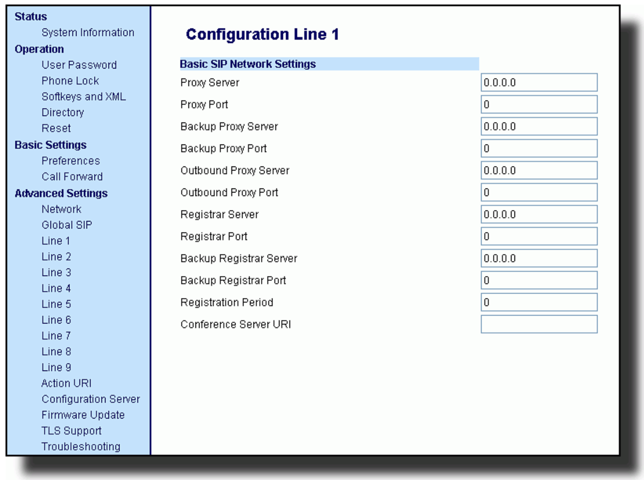


</td></tr>
<tr><td>2</td><td>In the “Conference Server URI” field, do one of the following actions:
• To disable centralized conferencing on the IP phone, leave this field empty (blank).
• To enable SIP centralized conferencing on the IP phone, do one of the following actions:
— If you have specified 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></td></tr>
</table>

Aastra Web UI	
3	Click **Save Settings** to save your changes.
Per-Line Configuration	
1	Click on **Advanced Settings->Line <#>->Basic SIP Network Settings** 
2	In the “**Conference Server URI**” field, do one of the following actions: • To disable centralized conferencing on this line, leave this field empty (blank). • To enable SIP centralized conferencing on this line, do one of the following actions: — If you have specified a proxy server/registrar server, then to reach the media server via the proxy server, set this field to one of the following: **conf** (Sylantro server), or **Conference** (Broadsoft server) By setting this field to **conf** or **Conference**, you specify conf@<proxy_server _address>:<proxy_port>. — To reach the media server using a different address/port than that specified by the proxy, set this field to the following: **conf@<media_server _address>: <media_port>**
3	Click **Save Settings** to save your changes.

Automatic Call Distribution (ACD) Support for Sylantro Servers

Release 2.1 of the IP phones includes Automatic Call Distribution (ACD) support 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 first must 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 on page 66).

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

Note: Aastra recommends configuring no more than a single ACD softkey or programmable key per IP phone.

ACD LED Table

The LED located next to the ACD softkey flashes or remains solid to indicate the current status of the IP phone. In addition, an icon on the IP Phone UI reminds IP phone users of their current status.

The following table describes the meaning of the ACD LEDs and icons for each model IP phone.

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

Using the 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.

Configuring ACD Softkeys Using Configuration Files (55i, 57i, 57i CT)

You use the following parameters to configure ACD support in the configuration files for 55i, 57i, 57i CT:

- **`softkeyN type`**
- **`softkeyN label`**
- **`softkeyN line`**
- **`softkeyN states`**

A sample configuration file is shown below:

```
softkey1 type: acd
softkey1 label: sales
softkey1 line: 1
softkey1 states: idle
```

In addition to the above parameters, you can also use the following parameters to configure ACD support in the configuration files for 57i, 57i CT:

- **`topsoftkeyN type`**
- **`topsoftkeyN label`**
- **`topsoftkeyN line`**
- **`topsoftkeyN states`**

A sample configuration file is shown below:

```
topsoftkey1 type: acd
topsoftkey1 label: sales
topsoftkey1 line: 1
topsoftkey1 states: idle
```

For complete parameter descriptions, refer to the following tables.

Parameters – *softkeyN type* *Auto Call Distribution* (in Web UI)	**Aastra Web UI** Operation->Softkeys and XML **Configuration Files** aastra.cfg, <mac>.cfg
Description	Defines the type of softkey that is configured for this key. To configure an ACD softkey, set this parameter to "**acd"**
Format	Text
Default Value	none
Default Range	See the IP Phone Administrator's Guide.
Example	softkey1 type: acd

Parameters – *softkeyN label* *Auto Call Distribution* (in Web UI)	**Aastra Web UI** Operation->Softkeys and XML **Configuration Files** aastra.cfg, <mac>.cfg
Description	Text label that displays on the IP phone for this softkey. Typically an easily recognizable name identifying the queue to which this softkey subscribes. For example: **Sales**
Format	Text
Default Value	Blank
Range	See the IP Phone Administrator's Guide.
Example	softkey1 label: Sales

Parameters – *softkeyN line* *Auto Call Distribution* (in Web UI)	**Aastra Web UI** Operation->Softkeys and XML **Configuration Files** aastra.cfg, <mac>.cfg
Description	This is the line associated with the softkey you are configuring. For example: **1**
Format	Integer
Default Value	1
Range	1 through 9
Example	softkey1 line: 1

Parameters – *softkeyN states* *Auto Call Distribution* (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. To configure an ACD softkey, set the state to "**idle**."
Format	Text
Default Value	**For softkey type - ACD:** idle
Range	See IP Phone Administrator's Guide.
Example	softkey1 states: idle

Parameters – *topsoftkeyN type* *Auto Call Distribution* (in Web UI)	**Aastra Web UI** Operation->Softkeys and XML **Configuration Files** aastra.cfg, <mac>.cfg
Description	Defines the type of top softkey that is configured for this key. To configure an ACD softkey, set this parameter to "**acd"**
Format	Text
Default Value	none
Default Range	See the IP Phone Administrator's Guide.
Example	topsoftkey1 type: acd

Parameters – *topsoftkeyN label* *Auto Call Distribution* (in Web UI)	**Aastra Web UI** Operation->Softkeys and XML **Configuration Files** aastra.cfg, <mac>.cfg
Description	Text label that displays on the IP phone for this top softkey. Typically an easily recognizable name identifying the queue to which this softkey subscribes. For example: **Sales**
Format	Text
Default Value	Blank
Range	See the IP Phone Administrator's Guide.
Example	topsoftkey1 label: Sales

Parameters – *topsoftkeyN line* *Auto Call Distribution* (in Web UI)	**Aastra Web UI** Operation->Softkeys and XML **Configuration Files** aastra.cfg, <mac>.cfg
Description	This is the line associated with the top softkey you are configuring. For example: **1**
Format	Integer
Default Value	1
Range	1 through 9
Example	topsoftkey1 line: 1

Parameters – *topsoftkeyN states* *Auto Call Distribution* (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 top softkey is pressed. To configure an ACD softkey, set the state to "**idle**."
Format	Text
Default Value	**For softkey type - ACD:** idle
Range	See IP Phone Administrator's Guide.
Example	topsoftkey1 states: idle

Configuring ACD Programmable Keys Using Configuration Files (53i, 55i)

You use the following parameter to configure ACD support in the configuration files:

- **`prgkeyN type`**
- **`prgkeyN line`**

Parameter – *prgkeyN type* *Auto Call Distribution* (in Web UI)	**Aastra Web UI** **Configuration Files**	Operation->Programmable Keys aastra.cfg, <mac>.cfg
Description	Defines the type of programmable key for this key. To configure an ACD programmable key, set this parameter to "**acd**."	
Format	Text	
Default Value	N/A	
Range	See the IP Phone Administrator's Guide.	
Example	prgkey4 type: acd	

Parameter – *prgkeyN line* *Auto Call Distribution* (in Web UI)	**Aastra Web UI** **Configuration Files**	Operation->Programmable Keys aastra.cfg, <mac>.cfg
Description	This is the line associated with the programmable key you are configuring.	
Format	Integer	
Default Value	1	
Range	1 through 9	
Example	prgkey4 line: 1	

Configuring ACD Expansion Keys Using Configuration Files (53i, 55i, 57i, and 57i CT)

You use the following parameter to configure ACD support in the configuration files:

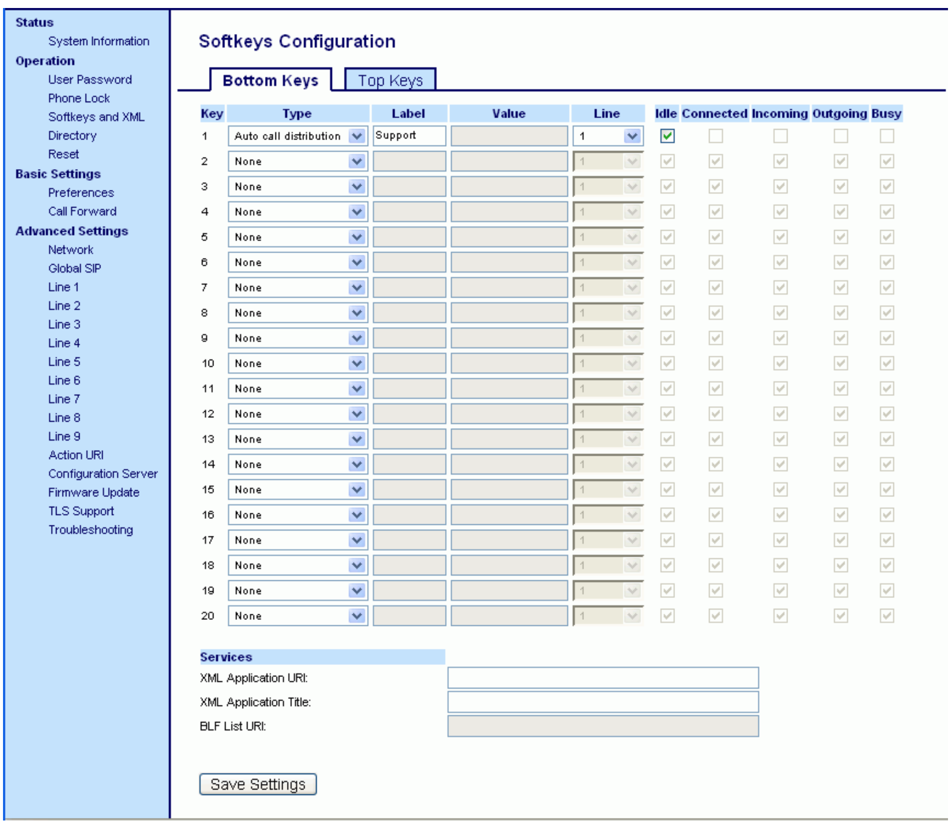
- **`expmodX keyN type`**
- **`expmodX keyN line`**

Parameter – *expmodX keyN type* *Type* (in Web UI)	**Aastra Web UI** Operation->Expansion Module N **Configuration Files** aastra.cfg, <mac>.cfg
Description	Defines the key type on the expansion module. Applicable expansion modules include the 536M and 560M. To configure an ACD expansion module key, set this parameter to "**acd**."
Format	Text
Default Value	none
Range	See IP Phone Administrator's Guide.
Example	expmod1 key1 type: acd

Parameter – *expmodX keyN line* *Line* (in Web UI)	**Aastra Web UI** Operation->Expansion Module N **Configuration Files** aastra.cfg, <mac>.cfg
Description	This is the line associated with the expansion module key you are configuring. Applicable expansion modules include the 536M and 560M.
Format	Integer
Default Value	1
Range	1 through 9
Example	expmod1 key1 line: 1

Configuring ACD Softkeys/Programmable Keys/Expansion Module Keys 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 57i IP phone as an example.

<table>
<tr><th colspan="2">Aastra Web UI</th></tr>
<tr><td>1</td><td>Click on Operation->Softkeys and XML, or Operation->Programmable Keys, or Operation->Expansion Module <N> depending on the model phone or expansion module you are configuring.

</td></tr>
<tr><td>2</td><td>In the “Type” field, select Auto Call Distribution.</td></tr>
</table>

	Aastra Web UI
3	In the **“Label”** field, specify a name for this ACD softkey. The Label helps identify which queue you are subscribing to when you login. (This field does not apply to the 53i.) For example: **Support**
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 ACD Auto-Available Timer Using Configuration Files

You use the following parameters to configure an ACD auto-configuration timer in the configuration files:

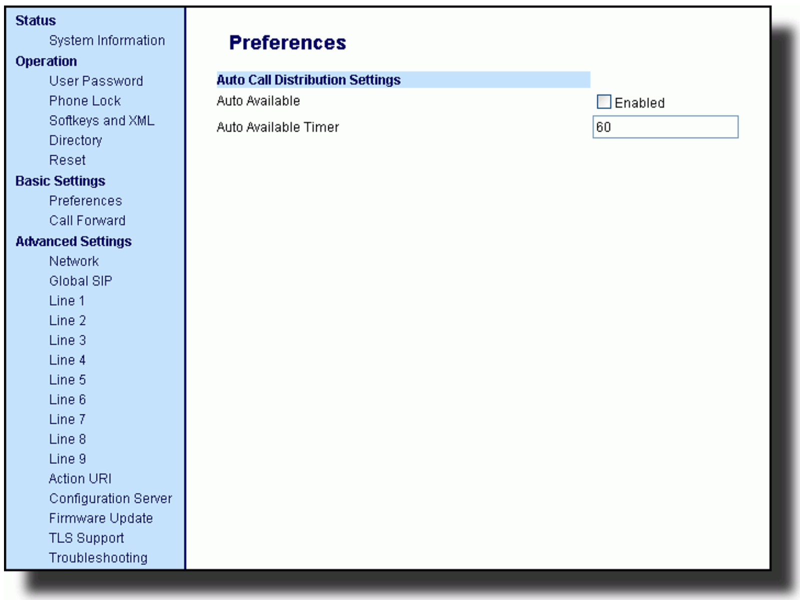
- **`acd auto available`**
- **`acd auto available timer`**

Parameter – *acd auto available* *Auto Available* (in Web UI)	**Aastra Web UI** Basic Settings->Preferences-> Auto Call Distribution Settings **Configuration Files** aastra.cfg, <mac>.cfg
Description	Enables the ACD timer to automatically reset the IP phone status to "available." Set to 0 = Off, Set to 1 = On
Format	Binary
Default Value	0 (disable)
Range	0 (disable) 1 (enable)
Example	acd auto available: 1

Parameters – *acd auto available timer* *Auto Available Timer* (in Web UI)	**Aastra Web UI** Basic Settings->Preferences-> Auto Call Distribution Settings **Configuration Files** 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 (<mac.cfg>)	acd auto available timer: 60

Configuring ACD Auto-Available Timer Using the 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.” For example: **60**
4	Click Save Settings to save your changes.

Using the ACD Feature on your IP Phone

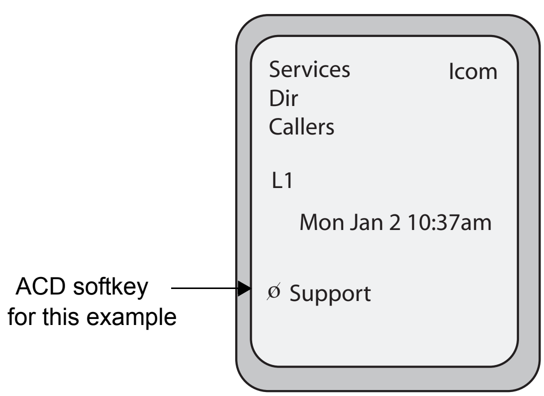
The ACD feature allows you to login to a phone queue in order to receive distributed calls on your IP phone. To login to a phone queue, your system administrator must preconfigure an ACD softkey or programmable key on your Aastra IP phone.

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

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

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

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



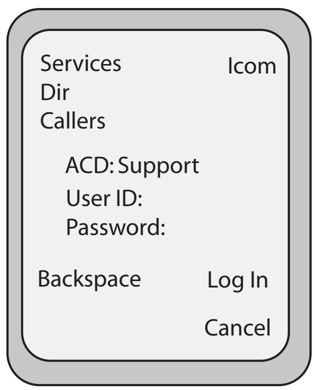
This icon changes when you log on to the phone queue and are available to take calls. The icon changes again when you are busy with an active call. The table below describes the meaning of the LED, and each icon, as they may appear on your IP phone:

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

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

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

Aastra IP Phone UI

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 (53i)

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

Aastra IP Phone UI

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

Incoming/Outgoing Intercom with Auto-Answer and Barge-In for 53i IP Phone

In Release 2.1, the 53i IP phone now supports incoming/outgoing Intercom calls with auto-answer and barge-in features.

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

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.

Allow Barge In

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

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

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

Configuring Intercom, Auto-Answer, and Barge-In via the Configuration Files

Use the following parameters to configure Intercom, Auto-Answer, and Barge-In via the configuration files:

Outgoing Intercom

- **`sip intercom type`**
- **`sip intercom line`**

Incoming Intercom

- **`sip allow auto-answer`**
- **`sip intercom mute mic`**
- **`sip play warning tone`**
- **`sip intercom allow barge in`**

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 line* *Line* (in Web UI)	**Aastra Web UI** Basic Settings->Preferences->Outgoing Intercom Settings **Configuration Files** aastra.cfg, <mac>.cfg
Description	Specifies the line for which the IP phone uses the configuration from, when making the Intercom call. The IP phone uses the first available line for physically making the call but uses the configuration from the line you set for this parameter. **Note:** The "*sip intercom type*" parameter must be set with the **Server-Side** option to enable the "*sip intercom line*" parameter.
Format	Integer
Default Value	1
Range	Line 1 through 9
Example	sip intercom line: 1

Incoming Intercom Settings

Parameter – *sip allow auto answer* *Auto-Answer* (in Web UI)	**Aastra Web UI** **Configuration Files**	Basic Settings->Preferences->Incoming Intercom Settings aastra.cfg, <mac>.cfg
Description	Enables or disables the IP phone to allow automatic answering for an Intercom call. If auto-answer is enabled on the IP phone, the phone plays a tone to alert the user before answering the intercom call. If auto-answer is disabled, the phone rejects the incoming intercom call and sends a busy signal to the caller.	
Format	Boolean	
Default Value	1 (true)	
Range	0 (false - do not allow auto-answer) 1 (true - allow auto-answer)	
Example	sip allow auto answer: 0	

Parameter – *sip intercom mute mic* *Microphone Mute* (in Web UI)	**Aastra Web UI** **Configuration Files**	Basic Settings->Preferences->Incoming Intercom Settings aastra.cfg, <mac>.cfg
Description	Enables or disables the microphone on the IP phone for Intercom calls made by the originating caller.	
Format	Integer	
Default Value	1 (true)	
Range	0 (false - microphone is not muted) 1 (true - microphone is muted)	
Example	sip intercom mute mic: 1	

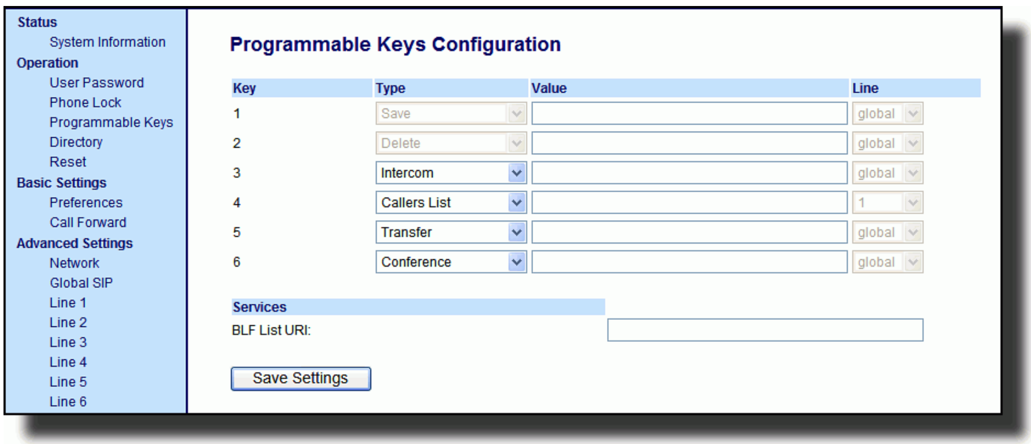
Parameter – *sip play warning tone* *Play Warning Tone* (in Web UI)	**Aastra Web UI** Basic Settings->Preferences->Incoming Intercom Settings **Configuration Files** aastra.cfg, <mac>.cfg
Description	Enables or disables a warning tone to play when the phone receives an incoming intercom call on an active line.
Format	Integer
Default Value	1 (true)
Range	0 (false - warning tone will not play) 1 (true - warning tone will play)
Example	sip play warning tone: 0

Parameter – *sip intercom allow barge in* *Allow Barge In* (in Web UI)	**Aastra Web UI:** Basic Settings->Preferences->Incoming Intercom Settings **Configuration Files** aastra.cfg, <mac>.cfg
Description	Enable or disables how the phone handles incoming intercom calls while the phone is on an active call. When you enable this parameter (1 = enable), which is the default value, an incoming intercom call takes precedence over any active call, by placing the active call on hold and automatically answering the intercom call. When you disable this parameter (0 = disable), and there is an active call, the phone treats an incoming intercom call like a normal call and plays the call warning tone. **Note:** After enabling or disabling this feature, it takes affect on the phone immediately.
Format	Boolean
Default Value	1 (true)
Range	0 (false) 1 (true)
Example	sip intercom allow barge in: 0

Configuring Intercom, Auto-Answer, and Barge-In via the Aastra Web UI

Use the following procedure to configure Intercom, Auto-Answer, and Barge-In using the Aastra Web UI.

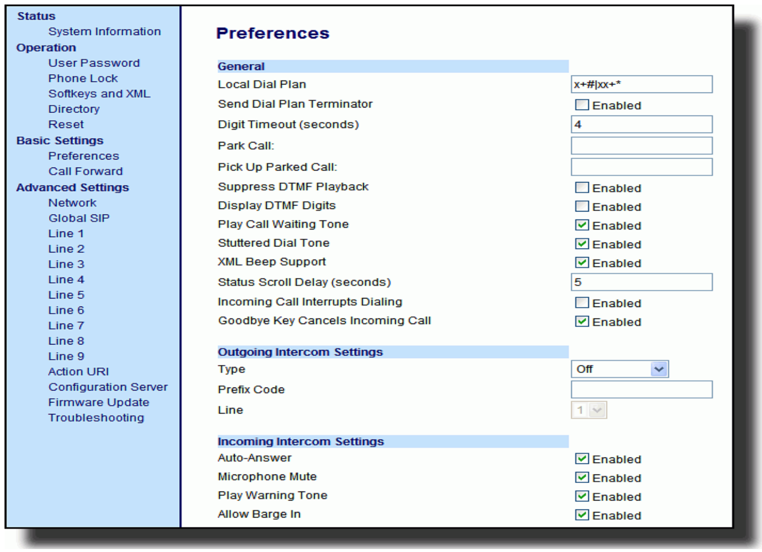
<table>
<tr><th colspan="2">Aastra Web UI</th></tr>
<tr><td>1</td><td>Click on Operation->Programmable Keys.



</td></tr>
<tr><td>2</td><td>In the “Type” field, select the Intercom option for a specific key.

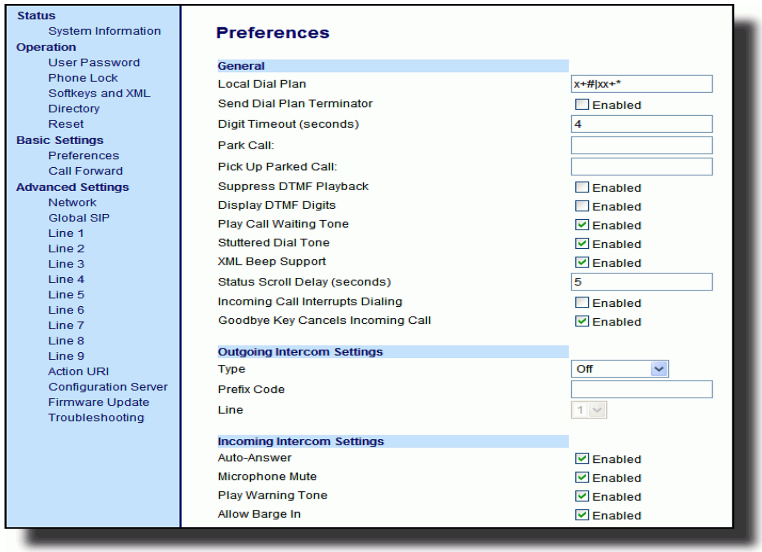
Note: Keys 3, 4, 5, and 6 are programmable only. Keys 1 and 2 are hardcoded and cannot be changed.</td></tr>
<tr><td>3</td><td>Click 

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

Aastra Web UI	
Outgoing intercom settings:	
1	Click on **Basic Settings->Preferences->Outgoing Intercom Settings**. 
2	Select an Intercom type for outgoing Intercom calls from the **Type** list box. Valid values are **Phone-Side**, **Server-Side**, **Off**. Default is **Off**.
3	If Server-Side is selected, enter a prefix to add to the phone number in the **"Prefix Code"** field. **Note:** For Sylantro servers, enter *96.
4	If Phone-Side or Server-Side is selected, select a line from the **Line** list box for which you want the IP phone to use as its configuration on the Intercom call. **Note:** The IP phone uses the configuration from the line you select from this list box. The call itself is made using the first available line at the time of the call.
5	Click Save Settings to save your changes.

Aastra Web UI

<table>
<tr><td colspan="2">Incoming intercom settings:</td></tr>
<tr><td>1</td><td>Click on Basic Settings->Preferences->Incoming Intercom Settings.



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

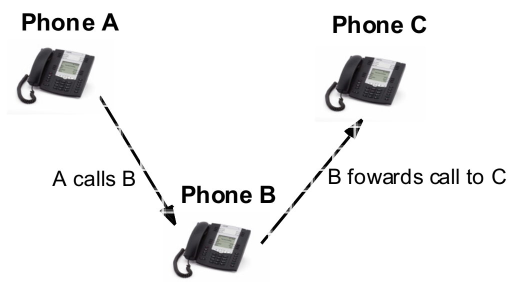
Note: If the Auto-Answer field is not checked (disabled), the phone rejects the incoming intercom call and sends a busy signal to the caller.</td></tr>
<tr><td>3</td><td>The "Microphone Mute" field is enabled by default. The microphone is muted on the IP phone for Intercom calls made by the originating caller. To disable this field, uncheck the box.</td></tr>
<tr><td>4</td><td>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.</td></tr>
<tr><td>5</td><td>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.</td></tr>
<tr><td>6</td><td>Click 

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

Missed Call Summary Subscription

A new feature in Release 2.1 on the IP phones allows missed calls that have been redirected by the server, to be incremented in the missed calls indicator on the phone it was initially directed to. This feature is called the Missed Call Summary Subscription and can be set with a timer that allows the phone to use the feature for a period of time before the timer expires. For this feature to work, you must configure voicemail on the phone that the call was initially directed to.

For example, phones A, B, and C are connected to the server. You configure the server to direct calls coming into phone B (which has voicemail configured) to be forwarded to phone C. When phone A calls phone B, the server forwards the call to phone C. With the new feature in 2.1, phone B receives notification from the server that the call was forwarded and the missed calls indicator is incremented on phone B.



Missed calls indicator increments on phone B.
Note:Voicemail must be configured on phone B.

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

Configuring Missed Call Summary Subscription using the Configuration Files

You can configure the Missed Call Summary Subscription feature on a global or per-line basis. 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.

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`**

Global Parameters

Parameter – *sip 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 **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** Advanced Settings->Global SIP->Advanced SIP Settings **Configuration Files** aastra.cfg, <mac>.cfg
Description	Enables or disables the amount of time, in seconds, that the phone uses the Missed Calls Summary Subscription feature. This parameter is always enabled with a default value of 86400 seconds. When the phone reaches the limit set for this parameter, it sends the subscription again. To disable this parameter, leave the field blank or set the field to zero (0).
Format	Integer
Default Value	86400
Range	0 to 99999999
Example	sip missed call summary subscription period: 70000

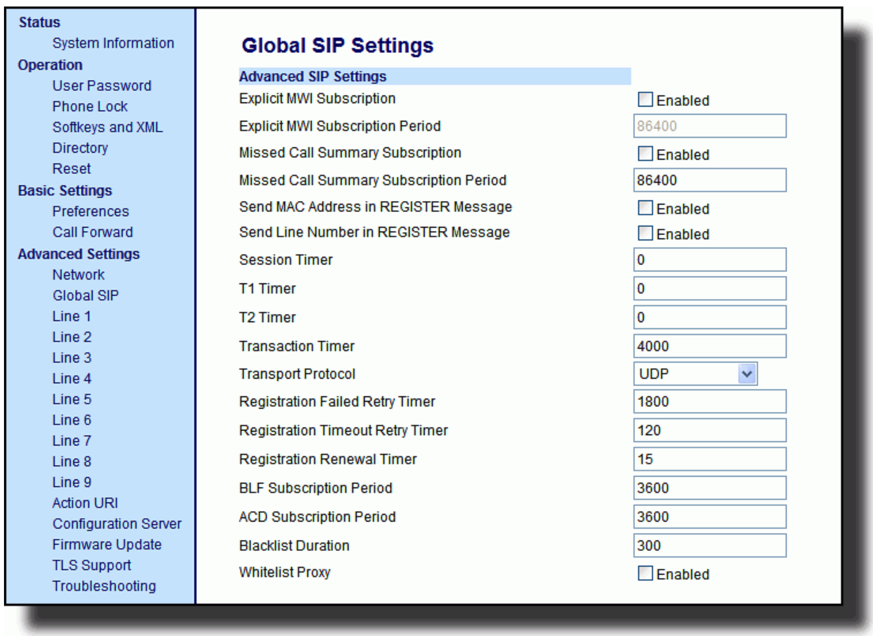
Per-Line Parameter

Parameter – *sip lineN missed call summary subscription* *Missed Call Summary Subscription* (in Web UI)	**Aastra Web UI** 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

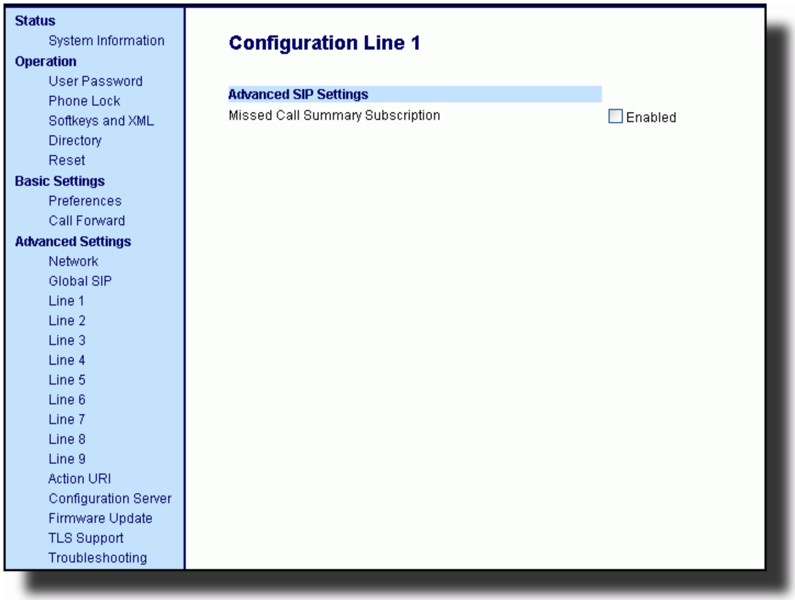
Configuring Missed Call Summary Subscription using the Aastra Web UI

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.
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.

Aastra Web UI

Per-Line Configuration	
1	Click on **Advanced Settings->Line <N>->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.
3	Click **Save Settings** to save your changes.

Message Waiting Indicator on Single or All Lines

Release 2.1 now allows a User or Administrator to 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 or Administrator can configure the MWI on single or all lines using the configuration files or the Aastra Web UI.

Configuring MWI using the Configuration Files

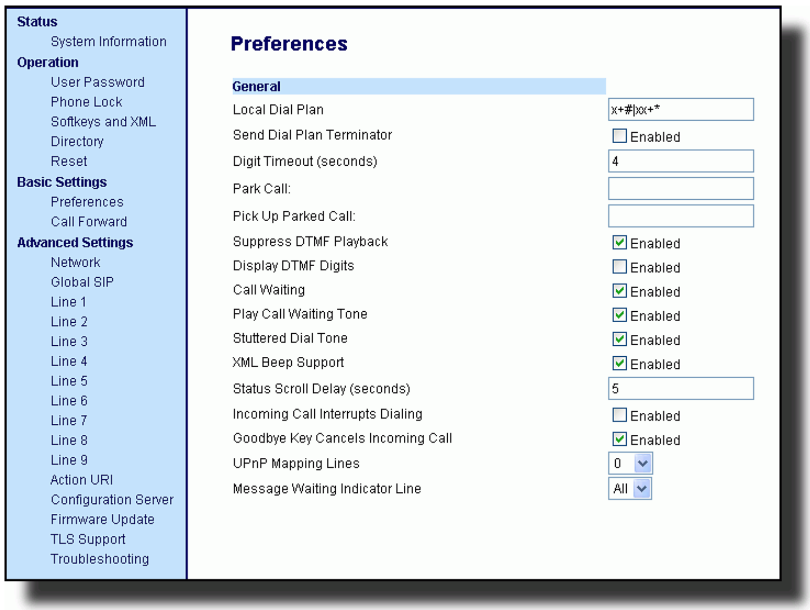
Use the following parameter to configure MWI for a single line or for all lines on the phone using the configuration files.

- **`mwi led line`**

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 to 9	
Example	mwi led line: 3	

Configuring MWI using the Aastra Web UI

Use the following procedure to configure the MWI for a single line or for all lines on the phone using the Aastra Web UI.

	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.

Support For “Delay” before Auto-Answer

Aastra IP Phones now 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 the Auto-Answer feature on the IP phone, as follows:

- Using the Aastra Web UI, see *Basic Settings->Preferences-> Incoming Intercom Settings*. Set the “**Auto-Answer**” parameter to **Enabled**.
- Using the configuration files, set **sip allow auto answer** to **1**.

See the *SIP IP Phone Administrator Guide* for more information on configuring Auto-Answer using either the Web UI, or the configuration files.

SIP Asserted Identity for Sylantro Servers

This release includes support for a private extension to the SIP, Asserted Identity within Trusted Networks (as defined in RFC 3325), inside the User Agent Server (UA) in the Aastra IP phones.

This feature allows a network of trusted SIP servers to assert the identity of authenticated users, and verify that phone messages originate from a Trusted Identity. Upon receiving a message from a caller in the Trust Network, the IP phone reads the contents of the P-Asserted-Identity (PAI) header field and displays it on the phone UI. This field contains a more accurate description of the caller identity (extension/phone number) than is contained in the SIP message.

When an IP phone receives an incoming call, the IP phone does the following actions:

- Checks to see if the incoming call is from a registered proxy server.
- If the call is forwarded via a registered proxy server, then the message has already been verified and authenticated by the server. The caller is part of the Trust Network. The IP phone UI displays the caller information contained in the PAI header.
- If the call is not forwarded via a registered proxy server - and therefore is not a “Trusted Entity” - the IP phone UI does not display any trust information contained in the PAI header.

"Whitelist" Proxy Support for Sylantro Servers

To protect your IP phone network, you can configure a "whitelist" proxy feature that screens incoming call requests received by Aastra 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 Support Using Configuration Files

You use the following parameter to configure the whitelist proxy feature in the configuration files:

- **`sip whitelist`**

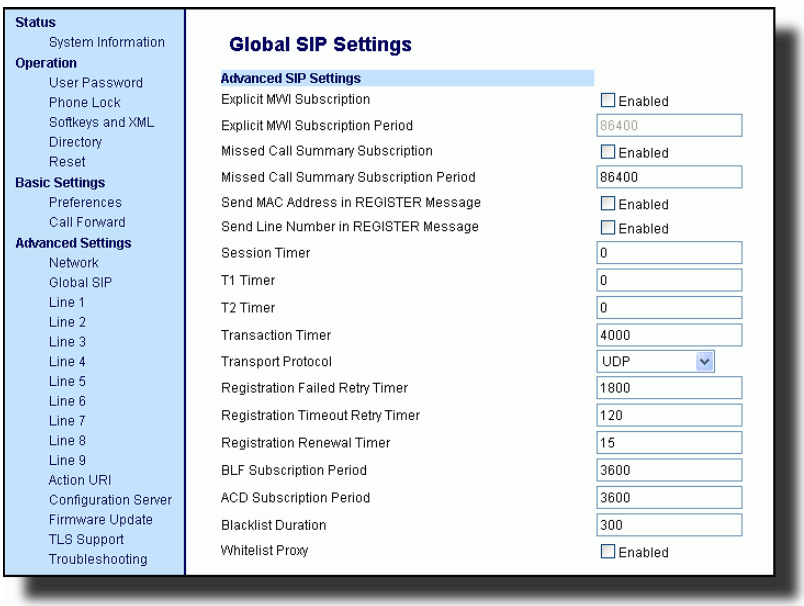
Parameter – *sip whitelist* *Whitelist Proxy* (Aastra Web UI)	**Aastra Web UI** **Configuration Files**	Advanced Settings->Global SIP-> Advanced SIP Settings aastra.cfg, <mac>.cfg
Description	This parameter enables/disables the whitelist proxy feature, as follows: • Set to 0 to disable the feature. • Set to 1 to enable the feature. When enabled, the IP phone rejects call requests received from an untrusted proxy server.	
Format	Boolean	
Default Value	0 (disabled)	
Range	0 (disabled) 1 (enabled)	
Example	sip whitelist: 1	

Configuring Whitelist Proxy Support 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. This feature allows you to enable or disable the whitelist proxy support on the IP Phone.

3 Click Save Settings to save your changes.

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
```

Broadsoft Interoperability Features

"Hold" Feature Enhancement for Broadsoft Servers

In this release of the Aastra IP phones, the "Hold" feature has been enhanced as described in RFC3264. The Hold feature allows IP phone users to put an active call on "hold," then retrieve the call later. This enhancement is intended for Aastra IP phones operating with servers that support RFC3264 (for example, Broadsoft). If a server does not support RFC3264, then the IP phones do not use RFC3264 functionality.

Centralized Conferencing for Broadsoft Servers

For information about this feature, see "Centralized Conferencing for Sylantro Servers" on page 60.

Support for the SIP "UPDATE" Message

This release of the Aastra IP phones supports the SIP UPDATE message, as specified in RFC3311.

Support for SIP Server Blacklist

The server blacklist 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, but 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. The default setting is 300 seconds. If you set the duration to 0 seconds, then you disable the blacklist feature.

Configuring a SIP Server Blacklist Using the Configuration Files

You use the following parameter to configure a SIP server blacklist in the configuration files:

- **`sip blacklist duration`**

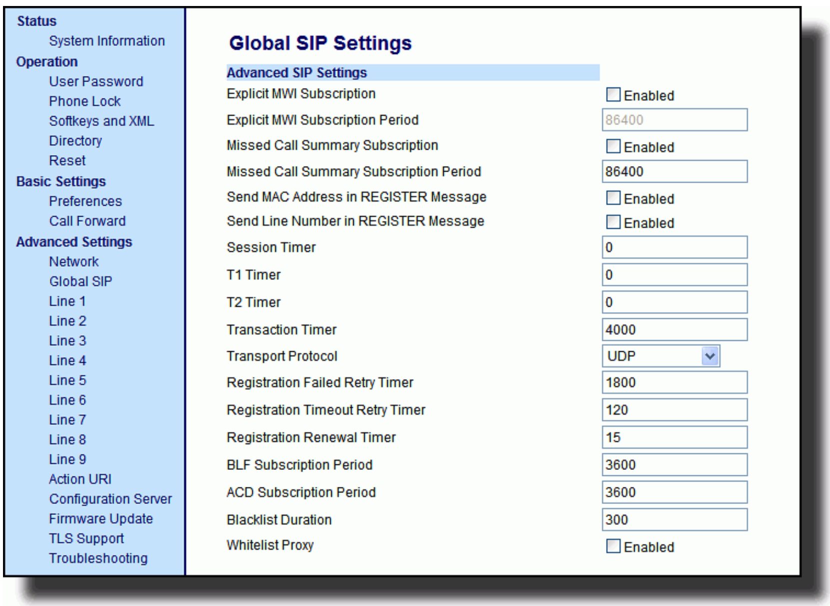
Parameter – *sip blacklist duration* *Blacklist Duration (Aastra Web UI)*	**Aastra Web UI** Advanced Settings->Global SIP Settings-> Advanced SIP Settings **Configuration Files** aastra.cfg, <mac>.cfg
Description	Specifies the length of time, in seconds, that a failed server remains on the server blacklist. 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

Configuring a Server Blacklist Using the Aastra Web UI

You use the following procedure to configure a server blacklist using the Aastra Web UI.

Aastra Web UI

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



2 In the "**Blacklist Duration**" field, specify the length of time, in seconds, that a failed server remains on the server blacklist.

The IP phone avoids sending a SIP message to a failed server (if another server is available) for this amount of time.

For example: **600**

3 Click Save Settings to save your changes.

Other Interoperability Features

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.

Symmetric UDP Signaling Support

By default, Aastra 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.

This release allows you to 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 Configuration Files

You use the following parameter to configure symmetric UDP signaling in the configuration files:

sip symmetric udp signaling

Parameter – *sip symmetric udp signaling*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	Allows you to enable or disable the phone to use port 5060 to send SIP UDP messages. The value "**1**" (which is the default) enables the phone to use port 5060. The value "**0**" (zero) disables the phone from using port 5060 and allows the phone to choose a random port to send SIP UDP messages.
Format	Boolean
Default Value	1 (enabled)
Range	0 (disabled) 1 (enabled)
Example	sip symmetric udp signaling: 0

Ability to Remove UserAgent and Server SIP Headers

Currently, the phone always configures the SIP UserAgent/Server headers to contain:

Aastra <PhoneModel>/<FirmwareVersion>

A new feature has been added to Release 2.1 that allows an Administrator to 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.

An Administrator 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.

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

Issues Resolved on Series 5i Phones in Release 2.1, Build 2145

This section describes the issues resolved on the Series 5i IP phones in release 2.1. The following table provides the issue number and a brief description of each fix.

Note: Unless specifically indicated, these resolved issues apply to all phone models.

Issue Number	Description of Fix
User Interface	
CLN06471	Dial tone for outgoing intercom calls is no longer cancelled after the inter-digit timeout expires.
CLN06707	Contrast setting is now applied at start of the phone booting.
CLN06721	Added a check when the NAT address was being set, to stop it being set to "…" which was corrupting the VIA header in SIP messages.
DEF04273	57i CT: Unanswered outbound call - CANCEL is now sent correctly after expiry of m_ringtimer.
DEF05136	"Directory" and "Callers" keys LED now light up when you enter these screens.
DEF05170	Softkeys are now accessible when you are on a call and another call comes in.
DEF06418	Phone no longer needs to be rebooted when a FQDN time server is configured via the web interface.
DEF06813	Added "status scroll delay" and "xml beep support" parameters configuration to the web interface.
DEF06866	Changing SIP DSCP value from the options menu now prompts for reboot.
DEF06958	57i CT: Fixed an intermittent crash on the 57i phone when making a call from the base to a cordless handset after you had previously removed the pairing for a 2nd cordless handset.
DEF07032	In the Aastra Web UI, the intercom "prefix code" field is now disabled for phone-side outgoing Intercom settings.
ENH07231	To speed up the operation of speed dial keys, DTMF Playback is now disabled by default.

Issue Number	Description of Fix
XML	
DEF05187 DEF05188 DEF05192 DEF05237 DEF05239	Speeddial now works correctly when an offhook Action URI is configured.
DEF05192	With offhook Action URI configured, phone doesn't dial the number in <Dial> tag.
DEF05237	Mute key now works when an XML object is displayed on the phone.
DEF05239	Offhook Action URI could trigger multiple times during a single call. Now it is only called once for each call.
DEF05719	53i: Default XML action keys are now displayed after changing the volume while on a call.
DEF06763	XML ImageScreen treats new lines as bitmap data.
DEF06810	Fixed a crash when time input format was used with custom softkeys.
DEF07088	Onhook Action URI is now fired when far-end hangs-up for a speakerphone or headset call.
DEF07120	Answer and Ignore keys are now displayed when using a formatted text screen.
BroadSoft Interop	
DEF05137	SCA: Phone no longer (incorrectly) seizes a second line when the first seize failed.
DEF05144	SCA: Simultaneous seize no longer turns of the led on another phone.
DEF05264	MCA with 3 calls, LED on line 2 no longer goes off when line 3 is ringing.
DEF05265	Broadsoft TCP: The transport=tcp in the contact header/req URI is now included in outgoing INV.
Sylantro Interop	
DEF03787	Sylantro: Third Party Registration now works correctly when using BLA.
DEF05654	BLA: Fixed a crash when the BLA line is deleted on the server.
DEF05669	57i Broadsoft SCA: Incoming call not longer results in SipEngine crash.
DEF05841	Broadsoft: Phones now support DNS caching with Broadsoft servers.
Call Control	
CLN05301	When there are two active calls on the CT (one on the base and one of the cordless handset) putting one of these calls on hold no longer mutes the other.
CLN05332	Sidetone is now only turned on during the call, once the call is finished it is now disabled.

Issue Number	Description of Fix
CLN06182	Mute key now works when the phone is locked. This enables the user to "answer" and intercom call, which automatically mutes the voice path.
DEF06632	57iCT: No longer automatically answers a second call when the user hangs up on the first call.
SIP	
CLN06064	Username containing a "." are now supported.
CLN06444	Phone now correctly follows the setting of the "sip cancel after blind transfer" parameter.
CLN06502	Phone now treats "600: Busy Everywhere" responses for an outgoing call as a busy indication, rather than a failed call.
DEF04481	VLAN tagged ARP packets now get to Port 1 as expected.
DEF04982	Early media call now works correctly; 200 OK no longer ignored.
DEF05265	When TCP is selected for the transport, "transport=tcp" is now added to the contact header/req URI of SIP messages.
DEF06630	Referred-By header is now added to the INVITE message from Tranferee.
DEF07268	Fixed a rare phone lockup caused by excessive SIP messages caused by configuring large numbers of BLF/List keys.
ENH04263	Support for server redundancy now works correctly.
ENH07192	Phone now unsubscribes to BLF and BLF/List services when they are removed via the Web interface.

Contacting Aastra Telecom Support

If you've read this release note, and consulted the Troubleshooting section of your phone model's manual and still have problems, please send inquiries via email to support@aastra.com.

Generic SIP IP Phone
Model 5i Series

2.1 Release Notes

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