



Aastra Models 6700i and 9000i Series SIP IP Phones



SIP 3.2.2 Release Notes

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SIP IP Phone Models 6700i and 9000i Series

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SIP IP Phone Models 6700i and 9000i Series Phones Release Notes 3.2.2

About this document

This document provides new features for the 6700i Series SIP IP Phones (6730i, 6731i, 6739i, 6753i, 6755i, 6757i, 6757i CT) and the 9000i Series SIP IP Phones (9143i, 9480i, 9480i CT) for release 3.2.2.

Note:

This release applies to the phone models mentioned above only.

For more detailed information about 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 *Aastra Models 9000i and 6700i Series SIP IP Phones Administrator Guide* and/or the *Development Guide XML API For Aastra SIP Phones Firmware 3.2.2*.

Release notes topics

Topics in these release notes include:

- [General information](#)
- [New Features in Release 3.2.2](#)
- [Additional Information](#)
- [Issues Resolved in Release 3.2.2](#)
- [Contacting Aastra Telecom Support](#)

General information

Release content information

This document provides release content information on the Aastra 9000i and 6700i Series SIP IP Phone firmware.

| Model | Release name | Release version | Release filename | Release date |
|----------|--------------|-----------------|--------------------|--------------|
| 6730i | Generic SIP | 3.2.2 | FC-001319-01-REV05 | June 2011 |
| 6731i | Generic SIP | 3.2.2 | FC-001318-01-REV05 | June 2011 |
| 6739i | Generic SIP | 3.2.2 | FC-001317-01-REV05 | June 2011 |
| 6753i | Generic SIP | 3.2.2 | FC-001321-01-REV05 | June 2011 |
| 6755i | Generic SIP | 3.2.2 | FC-001322-01-REV05 | June 2011 |
| 6757i | Generic SIP | 3.2.2 | FC-001323-01-REV05 | June 2011 |
| 6757i CT | Generic SIP | 3.2.2 | FC-001324-01-REV05 | June 2011 |
| 9143i | Generic SIP | 3.2.2 | FC-001325-01-REV05 | June 2011 |
| 9480i | Generic SIP | 3.2.2 | FC-001326-01-REV05 | June 2011 |
| 9480i CT | Generic SIP | 3.2.2 | FC-001327-01-REV05 | June 2011 |

Hardware supported

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

| 673xi Models | 675xi Models | 9xxxi Models |
|--------------|--------------|--------------|
| 6730i | 6753i | 9143i |
| 6731i | 6755i | 9480i |
| 6739i | 6757i | 9480i CT |
| | 6757i CT | |

Bootloader requirements

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

| 673xi Models | 675xi Models | 9xxx Models |
|---------------------------------------|---|--|
| 6730i: Bootloader 2.4.0.80 or higher | 6753i: Bootloader 2.0.1.1055 or higher | 9143i: Bootloader 2.2.0.166 or higher |
| 6731i: Bootloader 2.4.0.80 or higher | 6755i: Bootloader 2.0.1.1055 or higher | 9480i: Bootloader 2.2.0.166 or higher |
| 6739i: Bootloader 3.0.0.167 or higher | 6757i: Bootloader 2.0.1.1055 or higher | 9480i CT: Bootloader 2.2.0.166 or higher |
| | 6757i CT: Bootloader 2.0.1.1055 or higher | |

Before you upgrade

Please read before upgrading your phone

Note:

The following information applies to all IP phones EXCEPT the 6739i.

If you have a firmware version on your phone prior to 2.3, please read the following **IMPORTANT** information before upgrading the phones:

- **LLDP is enabled by default**

If LLDP is enabled on your network, the phones may come up with different network settings. For more information about LLDP, see the *Aastra Models 9000i and 6700i Series SIP IP Phones Administrator Guide*.

- **Support for DHCP Options 159 and 160**

If the DHCP server supplies Options 159 and 160, the phones will attempt to contact the configuration server given in these options. For more information about Options 159 and 160, see the *Aastra Models 9000i and 6700i Series SIP IP Phones Administrator Guide*.

- **HTTPS validation**

If you are using HTTPS and the certificates are not valid or are not signed by Veri-sign, Thawte, or GeoTrust, the phones fail to download configuration files. For more information about HTTPS validation, see the *Aastra Models 9000i and 6700i Series SIP IP Phones Administrator Guide*.

- **WatchDog task feature**

If the phone detects a failure (for example, a crash), the phone automatically reboots. For more information about the WatchDog feature, see the *Aastra Models 9000i and 6700i Series SIP IP Phones Administrator Guide*.

Note:

If you factory default a phone with Release 2.3 and above software, when the phone reboots, it attempts to connect to *rcs.aastra.com*. There is no personal information transmitted from the phone and the phone continues to boot up as normal.

Warning:

Applicable to IP Phone Models 6730i and 6731i only:

The default negotiation setting for the Ethernet ports on the phones is “auto-negotiation”. If you have changed this default setting to another value (i.e., Half Duplex or Full Duplex), you **MUST** set the negotiation value back to the default value of “auto-negotiation” before upgrading to Release 3.2. Failure to do so may cause the phone to fail to connect to the network. In addition, downgrade will not be possible.

New Features in Release 3.2.2

This section provides the new features in SIP IP Phone Release 3.2.2. These new features apply to all of the Aastra IP Phones, unless specifically stated otherwise.

The following table summarizes each new feature and provides a link to more information within this Release Note. Each feature also specifies whether it affects the Administrator, the User, or the XML Developer.

This table also provides the documentation location of features that have already been documented in Aastra's documentation suite. Refer to those documents for more information about the applicable feature.

| Feature | Description |
|---|--|
| Configuration Features | |
| Model Specific Configuration File (For Administrators) | In addition to the aastra.cfg and <mac>.cfg files, there is now a model specific configuration file that is requested after the aastra.cfg file is downloaded. The file is called <model>.cfg. *New for all phones |
| SIP Features | |
| Keep-Alive Feature for Persistent TLS Connections (For Administrators) | Now in 3.2.2, the keep-alive feature has been added for persistent TLS connections only. Administrators can configure this keep-alive feature using the new parameter called "sip persistent tls keep alive". When this feature is configured, the phone will send keep-alive pings to the proxy server at configured intervals. *New for all phones |
| Custom Ad Hoc Conference (For Administrators) | Now Administrators can use the "custom adhoc conference" parameter to enable the phone to interoperate with Genband Call Manager for ad hoc conferences. This parameter is disabled by default. *New for all phones |
| User Interface Features | |
| Shifting of Softkey Positions for Busy States (For Administrators) | User programmable softkeys 1 and 2 on the 6755i, 6757i, 6757iCT, 9480i, and 9480iCT phones will now appear on page 1 in position 4 and 5 during the outgoing, ringing, connected, and hold states. *New for 6757i, 6757i, 57iCT, 9480i, & 9480iCT only |
| Local Directory Loose Number Matching (For Administrators) | Administrators can enter a parameter " directory digits match " to determine the number of digits (from the incoming call) that the phone will use to perform a lookup in the local directory to match the number to a contact. *New for all phones |
| Background Image on Idle Screen (For Administrators) | The 6739i phone can now display a background image that will be shown on the idle screen only. *New for 6739i only |

| Feature | Description |
|--|--|
| Usability Features | |
| <p>Filter Out Incoming DTMF Events (For Administrators)</p> | <p>Administrator can now enter a parameter in the configuration files to suppress incoming DTMF playback. This new parameter called "suppress incoming dtmf playback" will suppress the playback of both SIP INFO and RFC2833 incoming DTMF tones. *New for all phones</p> |
| <p>Send DTMF for Remapping Conference or Redial Key (For Administrators)</p> | <p>Administrator can now enter two parameters to remap the "Conf" and "Redial" keys to send the stored number as DTMF using the phone configured DTMF method. *New for all phones</p> |
| Network Features | |
| <p>LLDP-MED Time-out Period (For Administrators)</p> | <p>Currently, Aastra IP Phones have a 5 second time-out for listening to LLDP-MED responses when the phone is booting up. Now administrators can configure this time-out interval using the new "lldp startinterval" parameter. *New for all phones</p> |

Additional Information

Configuration Features

Model Specific Configuration File

In addition to the `aastra.cfg` and `<mac>.cfg` files, there is now a model specific configuration file that is requested after the `aastra.cfg` file is downloaded. The file is called `<model>.cfg`, where "model" should be the same string that is used for the model name (e.g. `6730i.cfg`, `6731i.cfg`, `6739i.cfg`, `6753i.cfg`, `6755i.cfg`, `6757i.cfg`, `6757iCT.cfg`, `9143i.cfg`, `9480i.cfg`, `9480iCT.cfg`).

When the phone starts up, it request configuration files in the following order:

1. `aastra.cfg`
2. `<model>.cfg`
3. `<mac>.cfg`

If the same parameter appears in more once in the above configuration files, the last parameter/value read will be used (i.e., the following precedence rules will apply):

- Settings in the `<model>.cfg` file can overwrite `aastra.cfg` settings
- Settings in the `<mac>.cfg` file can overwrite `<model>.cfg` settings

The same download protocol and URL is used for the `<model>.cfg` file.

As with the `aastra.cfg` and `<mac>.cfg`, the `<model>.cfg` will be written to the `server.cfg` on the phone. The `server.cfg` contents will be a concatenation of the `aastra.cfg`, `<model>.cfg`, and `<mac>.cfg`, respectively.

Note:

The `<model>.cfg` file is only requested if the `aastra.cfg` file has been downloaded successfully. If the `<model>.cfg` file does not exist on the server and/or has timed out on the download, the phone will proceed to request download of the `<mac>.cfg` file.

Administrators can also encrypt the `<model>.cfg` file using an Astra-supplied encryption tool called "*anacrypt.exe*" to create a `<model>.tuz` file.

SIP Features

Keep-Alive Feature for Persistent TLS Connections

Now in 3.2.2, the keep-alive feature has been added for persistent TLS connections only. Administrators can configure this keep-alive feature using the new parameter called “**sip persistent tls keep alive**”. When this feature is configured, the phone will send keep-alive pings to the proxy server at configured intervals.

The keep-alive feature for persistent TLS connections performs the following functionalities:

1. After a persistent TLS connection is established or re-established, activate the keep-alive, which will send CRLF to peer periodically.
2. The phone will retry the connection automatically when a persistent TLS connection is down.
3. When a persistent TLS connection is re-established (primary is up or primary is down and backup is up), refresh registration of the accounts associated with the connection.
4. When a persistent TLS connection to primary is down, switch to backup if connection to backup is working.

Configuring Persistent TLS Keep-Alive

Use the following parameter to configure the persistent TLS keep-alive feature:

| | |
|---|---|
| Parameter – sip persistent tls keep alive | Configuration Files aastra.cfg, <mac>.cfg |
| Description | When enabled, the configured value indicates frequency (in seconds) that phone will send the keep alive messages. |
| Format | Integer |
| Default Value | 0 (disabled) |
| Range | 0-3600 |
| Example | sip persistent tls keep alive: 10 |

Custom Ad Hoc Conference

Previously, the phone will wait for server response prior to completing the ad hoc and centralized conference feature. This results in an interoperability issue with certain call managers, such as Genband. Now the phone no longer waits for server response before referring the call to the conference host when the "**custom adhoc conference**" parameter is enabled.

Configuring Custom Ad Hoc Conference

Use the following parameters to configure the custom ad hoc conference:

| | |
|---|---|
| Parameter – custom adhoc conference | Configuration Files aastra.cfg, <mac>.cfg |
| Description | Enables/disables the phone to interoperate with Genband Call Manager for ad hoc conference. |
| Format | Boolean |
| Default Value | 0 |
| Range | 0-1 0 (disable) 1 (enable) |
| Example | custom adhoc conference: 1 |

User Interface Features

Shifting of Softkey Positions for Busy States

Administrators can shift the softkey position for busy states using the **“collapsed context user softkey screen”** parameter. If this parameter is configured, user programmable softkeys 1 and 2 on the 6755i, 6757i, 6757iCT, 9480i, and 9480iCT phones will appear on page 1 in position 4 and 5 during the following states:

- outgoing
- ringing
- connected
- hold

If there are more than two softkeys, the phone will still show the “More” key and then show softkey 3 on the next page in position 1.

Note:

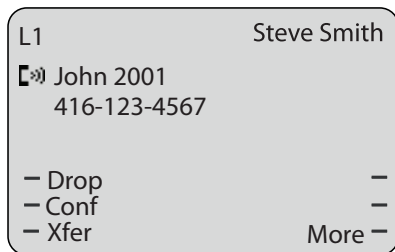
The phone defined context-sensitive softkeys (including empty/unassigned keys) will remain in the same location. As a result, some screens may only show 1 additional softkey (e.g. softkey 1).

For Example:

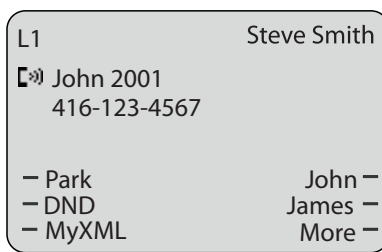
During a call with this parameter disabled (Example 1), softkeys 1 (Park) and 2 (DND) do not appear on page 1. The user has to press the More key to access them on page 2.

During a call with the **“collapsed context user softkey screen”** parameter enabled (Example 2), Park and DND appear on page 1 in position 4 and 5.

Example 1: During a call with the “**collapsed context user softkey screen**” parameter disabled (default).

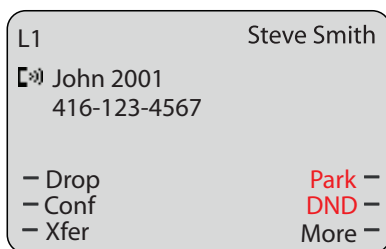


Page 1

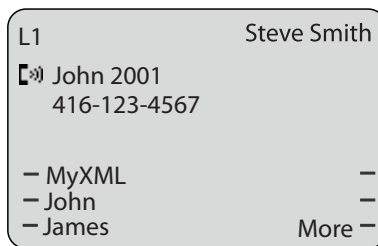


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Example 2: During a call with the “**collapsed context user softkey screen**” parameter enabled.



Page 1



Page 2

Configuring Collapsed Context User Softkey Screen

Use the following parameters to configure the collapsed context user softkey screen:

| | |
|---|---|
| Parameter — <i>collapsed context user softkey screen</i> | Configuration Files aastra.cfg, <mac>.cfg |
| Description | This will display the configured softkey1 and softkey2 on page 1 in position 4 and 5. |
| Format | Boolean |
| Default Value | 0 |
| Range | 0-1 0 (Disabled) 1 (Enabled) |
| Example | collapsed context user softkey screen:1 |

Local Directory Loose Number Matching

If the incoming calling number does not match exactly to the local directory, the lookup will fail and the phone will not display the stored directory name. Now, the phone can be configured to match only the specified number of digits against the local directory.

Administrators can enter a parameter “**directory digits match**” to determine the number of digits (from the incoming call) that the phone will use to perform a lookup in the local directory to match the number to a contact. The phone will take the last *n* digits of the incoming call to perform the lookup.

For example, if a local directory contains the following contact:

555123456 Spain Office,

and the “directory digits match” parameter is configured (directory digits match: 9), the phone will take the last 9 digits of the incoming number (+34555123456) and match it to the contact in the directory and display the following on the phone:

Spain Office

+34555123456

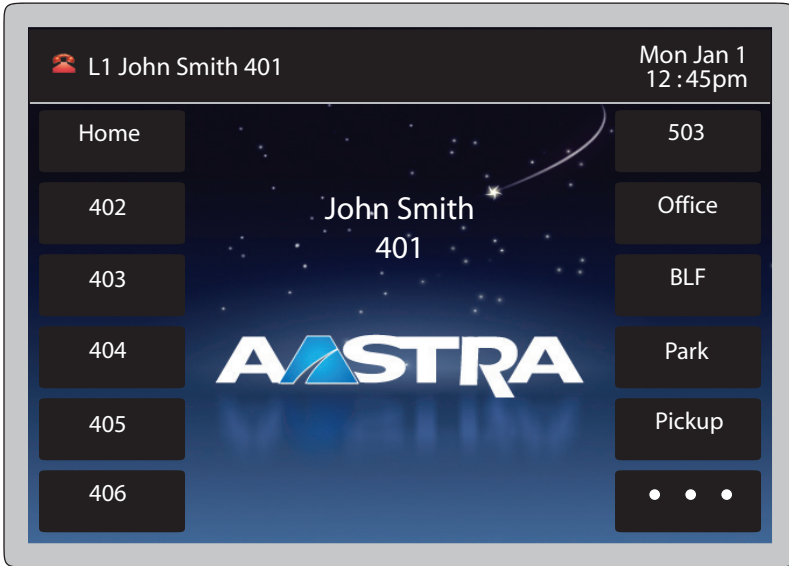
Configuring Local Directory Loose Number Matching

Use the following parameters to configure the local directory loose number matching:

| | |
|--|--|
| Parameter- <i>directory digits match</i> | Configuration Files aastra.cfg, <mac>.cfg |
| Description | The phone will take the last <i>n</i> digits of the incoming call to perform the lookup. |
| Format | Integer |
| Default Value | 0 (No Exact Matching) |
| Range | 0-10 |
| Example | directory digits match: 9 |

Background Image on Idle Screen

Customers can now brand the idle screen of the 6739i with their own company logo or image. The 6739i phone can now display a background image in the bottom layer of the idle screen. This image can be set by the new “**background image**” parameter. Current text (i.e. screen name, extension, and date) and softkeys on the idle screen are on top of the background image. The background image can be downloaded from your configuration server using either tftp, ftp, http, and https protocols.



Note:

The background image should not have white areas where the overlaying white text is located.

Idle Screen Image Requirements:

- 640 x 480 pixels
- There should be no frame around the image
- 24-bit color depth recommended
- Both .png and .jpg files are supported

Configuring Background Image

Use the following parameters to configure the background image:

| | |
|--|---|
| Parameter– background image | Configuration Files aastra.cfg, <mac>.cfg |
| Description | Set up background image for idle screen on 6739i phone. |
| Format | String (up to 256 characters) |
| Default Value | NA |
| Range | ftp://server/image.png ftp://server/image.jpg http://server/image.png https://server/image/jpg |
| Example | background image: http://10.30.100.233/pics/image.png |

Usability Features

Filter Out Incoming DTMF Events

In Release 3.2.2, an Administrator can now enter a parameter in the configuration files to suppress incoming DTMF playback. This new parameter called “**suppress incoming dtmf playback**” will suppress the playback of both SIP INFO and RFC2833 incoming DTMF tones. The locally generated DTMF tones will still be played.

Configuring Suppress Incoming DTMF Playback

Use the following parameters to suppress incoming DTMF playback:

| | |
|--|--|
| Parameter – <i>suppress incoming dtmf playback</i> | Configuration Files aastra.cfg, <mac>.cfg |
| Description | Suppress playback of both SIP INFO and RFC2833 DTMF tones. |
| Format | Boolean |
| Default Value | 0 |
| Range | 0-1 0 (Disabled) 1 (Enabled) |
| Example | suppress incoming dtmf playback:1 |

Send DTMF for Remapping Conference or Redial Key

Previously, the “Conf” and “Redial” keys could be mapped to a speed dial to generate a call to a custom number when the phone was idle. During an active call, pressing the “Redial” or “Conf” keys would put the current call on hold and then dial the custom number. Now the “Conf” and “Redial” key remappings have the same behavior as the “Speed Dial” key when the phone is idle. During an active call the phone will send the custom number as DTMF using the phone configured DTMF method (inbound vs out-of-band RFC2833 vs SIP INFO).

This feature can be configured using the new “**map redial as dtmf**” and “**map conf as dtmf**” parameters.

When a user presses the “Redial” key, the mapped number will be sent out as DTMF during an active call if the current “**map redial key to**” parameter is configured to a number and the “**map redial as dtmf**” parameter is set to “1”.

When a user presses the “Conf” key, the mapped number will be sent out as DTMF during an active call if the current “**map conf key to**” parameter is configured to a number and “**map conf as dtmf**” parameter is set to “1”

Configuring for Sending of DTMF with Remapped Redial and Conf Keys

Use the following parameters to configure the remapping of the “Redial” and “Conf” keys:

| | |
|--|---|
| Parameter– <i>map redial as dtmf</i> | Configuration Files aastra.cfg, <mac>.cfg |
| Description | Allows the phone to send the stored number as DTMF using the phone configured DTMF method when the “Redial” key is pressed. |
| Format | Boolean |
| Default Value | 0 |
| Range | 0-1 0 (Disabled) 1 (Enabled) |
| Example | map redial as dtmf:1 |

| | |
|--|---|
| Parameter– <i>map conf as dtmf</i> | Configuration Files aastra.cfg, <mac>.cfg |
| Description | Allows the phone to send the stored number as DTMF using the phone configured DTMF method when the “Conf” key is pressed. |
| Format | Boolean |
| Default Value | 0 |
| Range | 0-1 0 (Disabled) 1 (Enabled) |
| Example | map conf as dtmf: 1 |

| | |
|---|--|
| Parameter– <i>map conf key to</i> Map Conf Key To (In Web UI) | Aastra Web UI Configuration Files Basic Settings->Preferences->Key Mapping aastra.cfg, <mac>.cfg |
| Description | Sets the Conf key as a Speed Dial key if a value is entered for this parameter. If you leave this parameter blank, the Conf key returns to its original functionality. Note: If you configure the Conf key for speed dialing on the 6757i CT BaseStation, the Conf key on the 6757i CT handset retains its original functionality. The Conf key on the handset is not configured for speed dial. |
| Format | Integer |
| Default Value | NA |
| Range | NA |
| Example | map conf key to: 5551267 |

| | |
|---|--|
| Parameter– <i>map redial key to</i> Map Redial Key To (In Web UI) | Aastra Web UI Configuration Files Basic Settings ->Preferences->Key Mapping aastra.cfg, <mac>.cfg |
| Description | Sets the Redial key as a Speed Dial key if a value is entered for this parameter. If you leave this parameter blank, the Redial key returns to its original functionality. Note: If you configure the Redial key for speed dialing on the 6757i CTBase Station, the Redial key on the 6757i CT handset retains its original functionality. The Redial key on the handset is not configured for speed dial. |
| Format | Integer |
| Default Value | NA |
| Range | NA |
| Example | map redial key to: 5551234 |

Network Features

LLDP-MED Time-out Period

Currently, Aastra IP Phones have a 5 second time-out for listening to LLDP-MED responses when the phone is booting up. If LLDP-MED responses are received after this initial listening period, the phone will ignore the response. Now administrators can configure this time-out interval using the new "**lldp startinterval**" parameter. This parameter is only valid during the phone bootup process and it will control the LLDP-MED time-out interval where the phone sends LLDP-MED advertisements and listens for the LLDP-MED responses from the switch before proceeding to the DHCP stage. The default value of this parameter is 5 seconds.

Configuring LLDP-MED Time-out Period

Use the following parameters to configure the LLDP-MED time-out period:

| | |
|--|--|
| Parameter- <i>lldp startinterval</i> | Configuration Files aastra.cfg, <mac>.cfg |
| Description | Control the LLDP start interval. |
| Format | Integer |
| Default Value | 5 (seconds) |
| Range | 0- 65535 |
| Example | lldp startinterval: 7 |

Issues Resolved in Release 3.2.2

This section describes the issues resolved on the IP Phones in Release 3.2.2

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.

Issues Resolved

| Issue Number | Description of Fix |
|----------------------|--|
| Configuration | |
| DEF22868 | When the <mac>.tuz2 file is not successfully decrypted due to an unmatched key or a corrupted file, there is no warning message displayed. Now the warning message "Bad encrypted cfg" is displayed when the <mac>.tuz2 file is not successfully decrypted. |
| SIP | |
| DEF16955 | When the call is using Secure Real-time Transport Protocol (SRTP), the phone didn't send PUBLISH packets or Real-time Transport Control Protocol Extended Reports (RTCP-XR). Now the phones send PUBLISH packets and RTCP-XR reports for SRTP calls. |
| DEF23199 | The phone did not complete the call when a user dialed the destination using an IP address. Now users can dial the destination using an IP address. |
| DEF23233 | There was no secondary dial tone until the second digit was entered when the live dial pad was on. Now the secondary dial tone is heard when the user dials the first digit. |
| DEF22974 | When the outbound proxy is configured on the phone and if the primary proxy fails, the phone will restart after sending a SUBSCRIBE for as-feature-sync. Now the as-feature-sync message is successfully sent out when the backup outbound proxy is used and the phone does not restart. |
| DEF23257 | When a Speed Dial key was configured to send DTMF according to RFC2833, the DTMF "end" messages were incorrectly sent. Now each string of the RTP messages with DTMF content end with three "end" messages and the time between the RTP messages (i.e. the inter-digit time) is 40ms. |
| DEF23258 | 6739i: When a Speed Dial key is configured to send DTMF, according to RFC2833, the "end" messages do not respect the time stamp. The time stamp between the 2 digits increments correctly (720-800), however the first event had the event duration as 204 instead of '0', then incremented to 48, 640, 880 (end packet), which shifted the end packet out of range. Now the first event starts at '0', and then increments to 240, 480, 640 (end packet). |

| Issue Number | Description of Fix |
|----------------|---|
| DEF23639 | When a phone running the 2.5 or 2.6 firmware is upgraded to 3.2, the phone may fail to register in some unique scenarios if the SIP 401 challenge is received with the router WAN IP instead of the phone local IP address. Also, when an outbound proxy was configured, the symmetrical signalling was not working and a random port was used in the UDP source port header. Now 2.5 and 2.6 phones can upgrade and register successfully to 3.2.x. |
| DEF23671 | When the Automatic Call Distribution (ACD) softkey was configured on the phone, the user tried to enable the available and unavailable keys but instead was directed to the speed dial page. When the user presses the available and unavailable keys the phone should indicate that the call center state is "Available/Unavailable". This is now working as expected. |
| DEF23772 | The Switch UI Focus to Ringing Line feature was not working as expected. When the phone received a diverted phone call on L2 while L1 was in a Connected state, a redirection message appeared on the screen and the bottom softkeys disappeared (labels were gone and the buttons did not work). Now the phone UI retains focus on L1 after a second call diversion and the bottom softkeys are visible and work as expected. |
| CLN24113 | The focused line (active call) instead of the ringing line (a new call with distinctive ringing) was used to get the call waiting tone type, so the distinctive ringing on call waiting did not play correctly. Now the phone uses the ringing line to get the call waiting tone to play, instead of the focused line. |
| CLN24173 | When a user has set the message diversion feature (e.g. Lunch), this message is not shown on the calling party's display, nor is the option for the caller to contact the Operator. Now the message diversion information is successfully provided to the terminal via a NOTIFY XML message. |
| DEF24121 | 6739i: The phone didn't clear the status display "Lunch" when the user pressed the Goodbye hardkey to remove it. Now when the user follows the procedure to remove the "Lunch" status, the phone will successfully remove it, changing the status to "Lunch Off". |
| DEF22802/24118 | When the user selected Conference and then dialed the number, the call was not completed once they pressed Dial. Now users are able to complete the first conference call successfully. |
| CLN24178 | 6739i: The phone did not announce its 1000Mbps speed capabilities in the LLDP frame and the user was not able to select this option when manually setting the port speed in the Web or phone UI. Now users can set the port speed to 1000Mbps in the Web and phone UI and it is available in the LLDP frame. |
| DEF24048 | When the phone is configured to support Transport Layer Security (TLS) and interoperates with MX-ONE, free seating login (e.g. a call center phone with no permanent user, so each user must login to the phone in order to use it) fails and a restart was needed to successfully register. Now free seating login is seamlessly supported. |
| DEF23760 | When the phone received 183 with Session Description Protocol (SDP) right after receiving 180 ringing (without SDP), and the to-tag in 183 was different than 180, the phone didn't send RTP. This issue impacted the Early Media Mute Mic feature as the user couldn't talk when early media was received in that situation. Now the phone sends and receives RTP using information in the latest SDP received in the latest 18x message with SDP, and the Early Media Mute Mic feature works as expected. |

| Issue Number | Description of Fix |
|--------------|--|
| CLN24282 | When the phone was configured to send RTCP-XR to any valid <user-name>@<server> address, the phone did not send the PUBLISH message with the RTCP-XR statistics. Now the phone sends the PUBLISH message with the correct RTCP-XR stats. |
| CLN24296 | When the http digest force login parameter is enabled and HTTP is set as the download server and protocol in the aastra.cfg file, the login screen will ask the user for a username and password. When the admin typed in the wrong user ID and pressed Submit, the phone displayed a failed login message screen. When the admin pressed OK to exit the screen, the phone rebooted. Now the phone will simply exit the screen when the user presses OK to exit the failed login message screen. |
| CLN24317 | When the directed call pickup feature was enabled on the phone, the user wasn't able to pickup a monitored extension by pressing on the BLF key that was monitoring the extension. The issue was the directed call pickup didn't add the dialog from the NOTIFY message in the request URI. Now users are able to intercept a call from the BLF key using directed call pickup. |
| DEF21486 | Phone was not sending MOS scores for calls that were made using G.722 codec. This issue is fixed now. |

User interface

| | |
|----------------|---|
| DEF22650 | After locking the Speed Dial soft key using the configuration file, the phone UI did not display the "Key is locked" message on the Options > Preferences > Speed Dial edit screen. Now when the Speed Dial softkey is locked, you will see the "Key is locked" message on the phone UI when you try to edit the key. |
| DEF22710 | 6739i: When the http " http digest force login " parameter was configured, on the phone UI the phone didn't provide the correct authentication password, however it displayed a login softkey after bootup. When the user pressed the login softkey the phone did not allow the user to enter a username and password. Now the login softkey prompts the user for the username and password. |
| DEF22720 | When establishing Paging Multicast RTP group from the phone, the Paging key had to be pressed twice if the handset was hung up. Now the user only has to press the Paging key once. |
| DEF23340/23347 | When the input language is set on either the phone or Web UI to, for example, Nordic, the phone UI shows another language (e.g. Portuguese). Now when a input language is selected, it properly displays on the phone UI. |
| DEF23616/21789 | When the screen language was set to Russian, the phone UI displayed junk characters. Also in the Preferences > Language > Screen Language menu on 3-line LCD phones, the Russian language option was displayed incorrectly. Now Russian characters appear correctly on the phone UI. |
| DEF21688 | When the phone boots up with a screen language other than English, the phone displayed the "Downloading Language Packs" message in English, instead of in the selected screen language. Now the message is displayed in the appropriate screen language. |
| DEF22817 | In the Options > Preferences > Time and Date menu, the "time format" option is not translated when the screen language is not English. This has been corrected. |

| Issue Number | Description of Fix |
|--------------|---|
| DEF22823 | The Missed Calls message displayed on the phone was not translated when the screen language was changed. This has now been corrected. |
| DEF23598 | In the Options > Advanced > Reset menu, the text was not translated into the current screen language on the phone. This has now been corrected. |
| DEF23625 | The phone and DND softkey LCD lights are OFF when a user changes the phone line using the “->” or “<-” navigation buttons. For example, when the phone is in the idle state and a user defines a DND softkey and enables it, the LED light is ON for L1. If the user presses the “->” navigation button to L2, the DND softkey LED should still be ON. This behavior now works as expected. |
| ENH18625 | 6739i: When a user pressed the Redial key twice, the redial screen went back to the idle screen instead of dialing the last number. Now when a user presses the Redial key, the redial list is displayed on the screen. When they press the Redial key again, the phone will dial the last number. |
| CLN24130 | The phone displayed the caller ID from the previous call when the incoming call didn't have any name information. Now the caller ID works as expected if the caller has a username or not. |
| DEF21655 | When a user presses several times on a 3-line LCD phone hardkey to scroll through the characters assigned in the menu, only lowercase characters appeared. Now both uppercase and lowercase characters are appear on the screen. |
| DEF23642 | When a user changed the Speed Dial softkey to a Call Forward softkey on the Web UI, the softkey label was not updated (it kept the speed dial label instead of having the new call forward label). The softkey label now updates correctly when the user changes the softkey. |
| DEF24143 | When the 123 Input mode (numeric mode) is selected, if a digit was pressed repetitively in a short sequence, only the first key stroke took effect. For example in the Admin Menu > Config Server > TFTP > Primary TFTP, when “123” input mode was selected and 111 was entered, only the first “1” was shown on the screen. This issue was also seen when entering in an extension in the Speed Dial Edit screen. Now the “123” input mode is working as expected. |
| CLN24310 | The caller's list Quit softkey should simply exit the Callers List screen, however it opened the Speed Dial Edit screen. Now the Quit softkey exits the Callers List screen successfully. |
| XML | |
| DEF22545 | When 'beep' is enabled on the phones, a beep sound wasn't heard when the XML page is displayed. Now when the XML page is displayed, users can hear a beep sound. |
| DEF22907 | When displaying an XML AastralPPhoneTextMenu object, the right navigation key didn't work to select menu items. Now the right navigation key works correctly. |
| DEF23025 | When the users presses on the Redial hardkey or the XML TextMenu softkey, hooks-off the handset, and presses the Call softkey, audio is heard from the speaker instead of from the handset. Now audio is properly heard from the handset. |

| Issue Number | Description of Fix |
|--------------|--|
| DEF23028 | When some XML UI objects were show (i.e. TextMenu, ImageMenu), the Line and Speaker keys are not functional if there was no "Dial" tag involved. Now the Line and Speaker keys are functional while the phone shows any type of XML UI object. |
| DEF23082 | 6739i: Latin-2 characters were not properly displayed on a softkey label. Now the default font used for softkey labels fully supports Latin-2 characters. |
| DEF23129 | When the phone sends Register, and the server responds 2000K with XML body for local mode status, the phone failed to display "localmode" on the UI. Now the LocalMode status feature is working properly. |
| DEF23139 | When the AstralPPhoneInputScreen object is sent to the phone, the cursor is on the Name field and the type of character is capitals (ABC). After the first character, the character type doesn't change automatically to lowercase (abc). Now the character type automatically changes to lowercase once the first character is entered. |
| DEF24026 | The ChangeMode softkey label does not switch automatically to lowercase (abc) when the mode actually switches from uppercase to lowercase after the first character has been entered. The ChangeMode softkey label now automatically switches to lowercase once the first character is entered. |
| DEF23232 | 6739i: The custom Exit softkey did not work in the XML AstralPPhoneInputScreen UI object with multiple inputs. Now the custom Exit softkey works as expected. |
| DEF23512 | When a 3-line LCD phone was playing a WAV file, the characters '>>' were not displayed at the end of the 2nd line on the phone display. Now when the AstralPPhoneExecute object (with Wav.Play command and a valid WAV file) is assigned to a key, and the users presses the key, the WAV file starts streaming and the characters display properly. |
| DEF24029 | When the phone is playing a WAV file, the characters '>>' were incorrectly displayed at the end of the first line on the phone UI and the WAV file name was shown on the second line. Now the WAV file name is on the first line and the characters '>>' are on the second line. |
| DEF23618 | 6739i: When a TextMenu with, for example, 20 elements is displayed with unitScroll set to 'no', if selected item was item 6 (the second item on the second page), pressing the up arrow displayed items 2-5 whereas it should have displayed items 1-4. When the user pressed the down arrow, the UI displayed items 6-9, whereas it should have displayed items 5-8. The up and down arrows now behave like a "page up/down" arrow, showing the first/last non-visible elements on the list. When the user presses the up or down arrow, they will go to the previous or next page, showing the correct items. |

| Issue Number | Description of Fix |
|--------------|--|
| DEF23859 | When the user pressed the Scroll Up key in an XML text menu with 5 entries, the 5th item was not shown. Now when the user presses the Scroll Up key all entries are visible. |
| DEF20987 | 6739i: When the user added a new BLF key via the PBX management interface, the PBX sent a SIP NOTIFY message containing an XML Configuration Object, in order to add the new BLF/List key. The phone, however, did not un-subscribe from the current dialog subscription and did not start a new subscription. This issue has been corrected. Now the phone will un-subscribes from the current dialog subscription and will start a new subscription showing the BLF/List key properly. |

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 contact Aastra Telecom Support via one of these methods:

North America

- Toll Free 1-800-574-1611
- Direct +1-469-365-3639
- Online at <http://www.aastratelecom.com/support>, click on Contact Technical Support

Outside North America

Please contact your regional Aastra Technical Support.



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