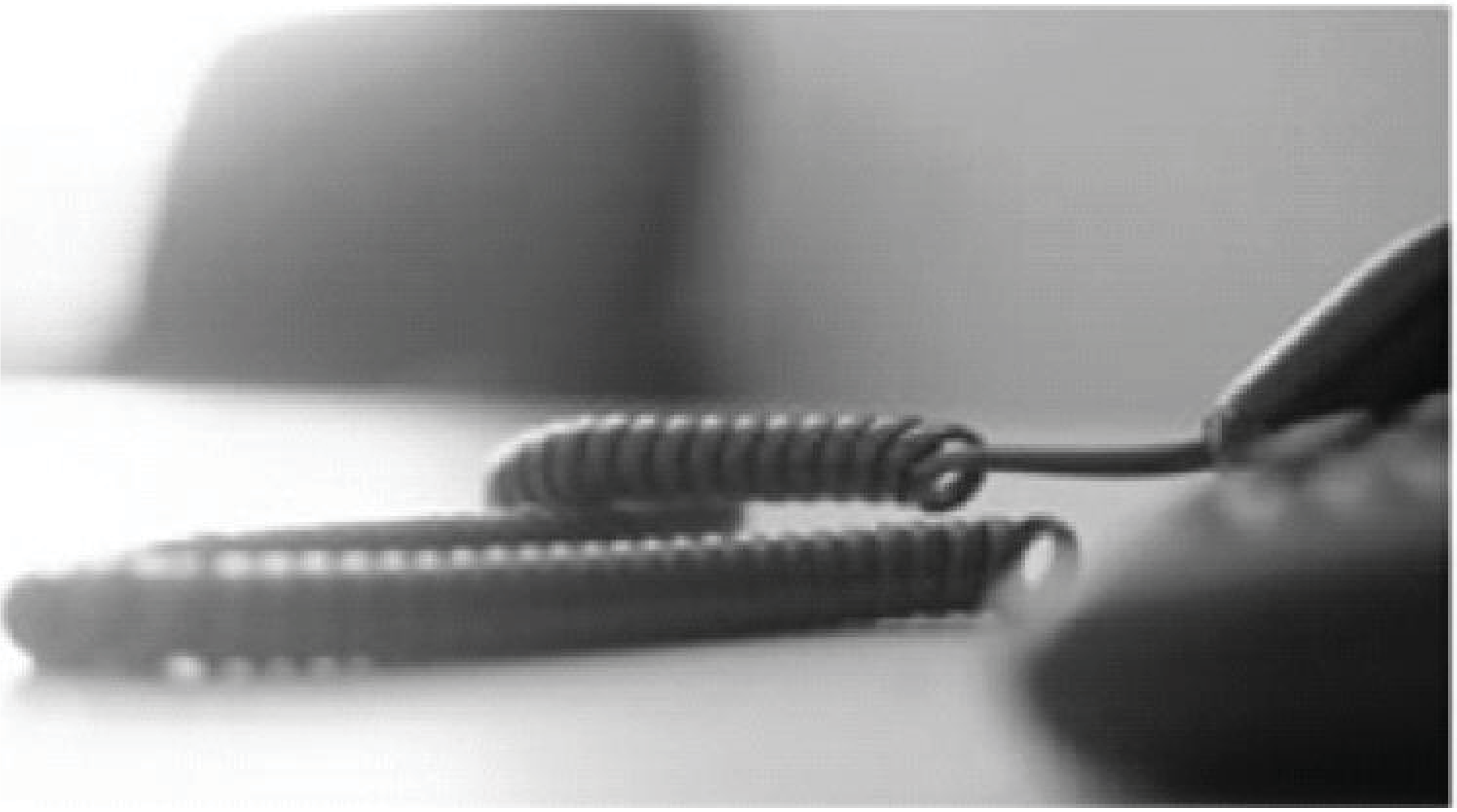
MODEL

9143i, 9480i, 9480i CT and 67xxi Series Phones

SIP IP PHONE

RN-001029-04
Rev 05

Release Note
Release 2.5.3



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SIP IP Phone Models 9143i, 9480i, 9480i CT, and 67xxi Series Phones Release Note 2.5.3

About this Document

This Release Note 2.5.3 provides issues resolved since the release of the 2.5.2 software for the 9143i, 9480i, and 9480i CT SIP IP Phones and the 67xxi Series SIP IP Phones (6730i, 6731i, 6751i, 6753i, 6755i, 6757i, 6757i CT).

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 SIP IP Phone Administrator Guide.

Topics in this release note include:

- General Information
  (release content, hardware supported, bootloader requirements)
- New Features in Release 2.5.3
- Issues Resolved in Release 2.5.3
- Contacting Aastra Telecom Support

General Information

Release Content Information

This document provides release content information on the Aastra 9143i, 9480i, and 9480i CT SIP IP Phone firmware and the 67xxi Series SIP IP Phone firmware.

Model	Release Name	Release Version	Release Filename	Release Date
6730i	Generic SIP	2.5.3	FC-001240-02-Rev07	January 2010
6731i	Generic SIP	2.5.3	FC-001224-03-Rev07	January 2010
6751i	Generic SIP	2.5.3	FC-001126-05-Rev07	January 2010
6753i	Generic SIP	2.5.3	FC-001086-09-Rev07	January 2010
6755i	Generic SIP	2.5.3	FC-001087-10-Rev07	January 2010
6757i	Generic SIP	2.5.3	FC-001088-09-Rev07	January 2010
6757i CT	Generic SIP	2.5.3	FC-001089-07-Rev07	January 2010
9143i	Generic SIP	2.5.3	FC-001092-05-Rev07	January 2010
9480i	Generic SIP	2.5.3	FC-001097-05-Rev07	January 2010
9480i CT	Generic SIP	2.5.3	FC-001101-05-Rev07	January 2010

Hardware Supported

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

- 6730i
- 6731i
- 6751i
- 6753i
- 6755i
- 6757i
- 6757i CT
- 9143i
- 9480i
- 9480i CT

Bootloader Requirements

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

- 6730i - Bootloader 2.4.0.80 or higher
- 6731i - Bootloader 2.4.0.80 or higher
- 6751i - Bootloader 2.0.1.1055 or higher
- 6753i - Bootloader 2.0.1.1055 or higher
- 6755i - Bootloader 2.0.1.1055 or higher
- 6757i - Bootloader 2.0.1.1055 or higher
- 6757i CT - Bootloader 2.0.1.1055 or higher
- 9143i - Bootloader 2.2.0.166 or higher
- 9480i - Bootloader 2.2.0.166 or higher
- 9480i CT - Bootloader 2.2.0.166 or higher

Before you Upgrade

Please Read Before Upgrading Your Phone

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 SIP IP Phone 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 SIP IP Phone Administrator Guide***.
- **HTTPS validation.**
  If you are using HTTPS and the certificates are not valid or are not signed by Verisign, Thawte, or GeoTrust, the phones fail to download configuration files. For more information about HTTPS validation, see the ***Aastra SIP IP Phone 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 SIP IP Phone 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.

New Features in Release 2.5.3

This section provides the new features in SIP IP Phone Release 2.5.3. These new features apply to all of the Aastra IP Phones, unless specifically stated otherwise. Each feature also specifies whether it affects the Administrator, the User, or both.

Feature	Description
Configuration	
New Parameter for Configurable Contact Header Matching (Administrator)	An Administrator can now specify whether or not to use strict SIP contact header matching on the IP Phones using the following new parameter in the configuration files: **sip contact matching**.

Configuration Features

New Parameter for Configurable Contact Header Matching

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

A new parameter, “**sip contact matching**”, now allows the Administrator to specify the method used by the phone to match the Contact Header. This parameter is available via the configuration files only.

Configuring the Contact Header Matching

Parameter – *sip contact matching*	**Configuration Files** aastra.cfg, <mac>.cfg
Description	Specifies the method for which the phone uses to match the Contact header in a SIP registration packet.
Format	Integer
Default Value	0
Range	0 (default) URI matching of username, domain name, port, and transport 1 matching of port only 2 matching of username only 3 matching of port and username only
Example	sip contact matching: 1

Issues Resolved in Release 2.5.3

This section describes the issues resolved on the IP Phones in Release 2.5.3. 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.

Release 2.5.3

Issue Number	Description of Fix
SIP	
CLN19852	The phone was randomly freezing during calls and in the idle state. After a re-subscribe only a NOTIFY was sent to the phone - not an OK response. This NOTIFY worked fine but the next NOTIFY that was sent caused a freeze on the phone. This is now corrected.
DEF17372	When using RTP Control Protocol Extended Report (RTCP-XR) in a network, the Network Packet Loss Rate (NLR), Burst/Gap densities & durations were reported incorrectly. This is now corrected.
DEF17514	The IP Phones set the VLAN ID according to the Type-Length-Values (TLV) when receiving a civic location TLV alone. When ELIN is enabled (an additional location TLV with ELIN info), the phone cannot set the VLAN ID according to the network-policy. This issue has been corrected.
DEF18682	A REINVITE from G.722 to G.711 codecs resulted in the “Hi-Q” icon staying on the phone’s screen. This has now been corrected and the “Hi-Q” icon now correctly goes away.
DEF18707	A REINVITE from G.711 to G.722 codes does not display the Hi-Q icon on the Phone UI. This has been corrected and the HI-Q icon now displays as expected.
DEF19258	When the IP Phone uses persistent Transport Layer Security (TLS) with an intermediate certificate that has been signed by the root Certificate Authority (CA), the intermediate CA then issues the server certificate. However, a TLS session cannot be established. After the TLS Handshake, the phone sends a Transmission Control Protocol Finished (TCP FIN). This is now corrected. The phone now registers and connects with any Common Name (CN) used for the proxy and registrar server name.
DEF19416	If TLS is used on the phone, it uses the SIPS URI scheme for all requests such as REGISTER, INVITE, and SUBSCRIBE. However, REFERs are sent using SIP URI scheme. As a result, a call transfer may not work properly when using TLS. This is now corrected. A SIP REFER is now correctly sent using SIPS URI scheme.

Issue Number	Description of Fix
DEF19446	When a phone receives an INVITE, the user accepts the call (ACK) and the phone sends an OK response. If the Call Manager sends a Re-INVITE before the ACK, the phone freezes. This is now corrected.
DEF19454	If the phone is set to use Secure Real-Time Transport Protocol (SRTP) with a setting of "**SRTP Preferred**" mode, and in a non-SRTP session the phone receives a re-INVITE with Session Description Protocol (SDP) containing a crypto key, the phone answers with an OK response containing no crypto key in SRTP. The peer sends SRTP but the phone cannot decrypt it (since it's not in SRTP-mode). Phone sends unencrypted RTP. On a re-INVITE the phone now sets the local SDP. If there is a change involving the crypto key, the change is now included in the OK response.
DEF19455	When a re-INVITE with no SDP comes into a phone with an on-hold call, the phone doesn't respond with its Codec list and crypto key. This is now corrected.
DEF19466	When a 67xxi phone is communicating with another 67xxi phone using SRTP for voice, communication works correctly. However, after approximately 20 minutes of communication, the line goes silent with no audio, even though SRTP packets are still being exchanged. This is now corrected.
DEF19588	When the phone registered to the primary server, static TLS authentication performed as expected (phone used trusted certificate to validate server certificate). When the primary server failed and the phone performed a failover to the backup registrar, the certificate for the backup registrar was signed by the same root CA but the phone refused the certificate with "unknown CA". This has now been corrected and the phone now accepts the applicable certificate(s) from the backup registrar.
ENH17750	Previously, the IP Phones did not parse Session Description Protocol (SDP) Annex B attribute when using a G.729AB codec. This caused an interoperability issue. The phone now fully supports the G.729 Annex A and Annex B.
ENH17935	A phone enabled with Link-Layer Discovery Protocol (LLDP) does not set the VLAN ID when receiving an LLDP frame containing Type-Length-Values (TLV) which sets the data and VLAN ID from the switch. The IP Phones now accept LLDP frames and the phones set the VLAN ID correctly.
ENH19694	The default Network Time Protocol value on the IP Phones is now set to "**aastra.pool.ntp.org**." This resolves some issues with certificate validation.
DEF20397	Network Packet Loss Rate (NLR) values provided by phone were missing fractional values and rounding was not always done correctly. This is now corrected. NLR values now contain fractions if they exist and rounding is done correctly.
Usability	
DEF18026	If the "**Switch UI Focus to Ringing Line**" parameter is disabled, and two calls come into the phone at the same time (Line 1 and Line 2), when the User picks up the call on Line 1, a ringing is heard in the handset from Line 2. This is now corrected. A User no longer hears ringing in the handset when picking up the first Line.

Issue Number	Description of Fix
DEF18140	If there are two Users with different numbers that have the same Calling Line ID in Broadsoft, the signalling for BLF comes in only on the first BLF key. This is now corrected. The signalling now comes in on both Users BLF keys as expected.
DEF18630	In some call scenarios, the TLS/SRTP key icon displayed when it shouldn't have (RTP used) or didn't display when it should have. (SRTP used). This was inconsistent behavior for these scenarios. This has been corrected and the TLS/SRTP key icon now displays (or doesn't display) as expected.
DEF18733	Previously on the phones, when in a menu list, the User could press the number of the item in the list and then press the action key to go to the item's menu. Now after pressing the number of the item in the list, the phone goes directly to the item's menu.
DEF20093	When pressing the ACD key, the display showed "**Please log in**" in English, regardless of the language set on the phone. This is now corrected. The prompt "**Please log in**" now displays in the language currently set on the phone.
DEF20366	When the phone is set to use SRTP and the mode is set to "**SRTP Only**", this is causing one-way voice path for active calls. This is now corrected.
XML	
DEF17457	Previously, when you created a Textmenu with the '**wraplist**' tag set to **yes**, and used it to display a succession of 2-line items and 1-line item, when you scrolled from a 2-line item to a 1-line item the second line of the previous item still displayed. This is now corrected.
DEF17763	When entering values for XML objects, some non-ASCII UTF-8 characters (that use 2 bytes) were not correctly displayed. This is now corrected.
DEF19494	When XML variables, such as $$SIPAUTHNAME$$, are passed to the phone using the AastraIpPhoneExecute object, the phone did not accept the variables. This is now corrected.
ENH18492	On 3-line LCD phones, if you create an AastraIPPhoneInput field of type "string", there was no way to input a "+" character. You can now use the "0" key to enter a "+" for an XML object's string.
ENH18956	Previously, in an XML tag, the application would wrap-around from the last entry in a menu list to the first entry, and likewise from the first entry to the last entry. This wrapping has now been disabled to allow a User to stop at the first or last entry in a menu list.
ENH19092	Previously in XML TextMenu screens on the phone, pressing a number associated with the menu item did not launch that menu item. Now, when you press the number for a menu item on an XML screen, the phone launches that menu item.

Issue Number	Description of Fix
ENH19141	Currently, in an XML application, the Right navigation arrow (Done) on the phone ca be used for the NextURI, and the Left navigation arrow (Back) can be used for the PrevURI. Now the phones can also use the Up and Down navigation arrows to push URIs and override the default behavior. These navigation keys can be used with the objects: TextScreen, FormattedTextScreen and ImageScreen. **Note:** The navigation keys take their default actions (up, down, left, and right) unless a URI is specified.
ENH20321	Previously, when an XML application was loading, the display showed “Loading page.” Now the displays shows “Please wait....” when an XML application loads to the phone.

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
Models 9143i, 9480i, 9480i CT, and 67xxi Series

2.5.3 Release Notes

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