



MODEL
480i, 480i CT, 9112i, 9133i

SIP IP PHONE

RN-001033-00

Rev 02

Release Note
Release 1.4.3



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SIP IP Phone Models 480i, 480i CT, 9112i, 9133i Release Note 1.4.3

About this Document

This Release Note 1.4.3 provides issues resolved for the 480i, 480i CT, 9112i, and 9133i SIP IP Phones. It includes the issues resolved between Release 1.4.2 and 1.4.3.

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)
- [Issues Resolved in Release 1.4.3](#)
- [Enhancements in Release 1.4.3](#)
- [Contacting Aastra Telecom Support](#)

General Information

Release Content Information

This document provides release content information on the Aastra 480i, 480i CT, 9112i, and 9133i SIP IP Phone firmware for Release 1.4.3.

Model	Release Name	Release Version	Release Filename	Release Date
9112i	Generic SIP	1.4.3	FC-000058-01-REV14	August 2008
9133i	Generic SIP	1.4.3	FC-000046-01-REV14	August 2008
480i	Generic SIP	1.4.3	FC-000032-01-REV14	August 2008
480i CT	Generic SIP	1.4.3	FC-000040-00-REV14	August 2008

Hardware Supported

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

- 480i
- 480i CT
- 9112i
- 9133i

Bootloader Requirements

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

- 480i - Bootloader 1.1.0.4 or above
- 480i CT - Bootloader 1.1.0.4 or above
- 9112i - Bootloader 1.1.0.10 or above
- 9133i - Bootloader 1.1.0.10 or above

Issues Resolved in Release 1.4.3

This section describes the issues resolved on the 480i, 480i CT, 9112i, and 9133i SIP IP Phones in Release 1.4.3. 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.

Issue Number	Description of Fix
CLN06333	480CT: If you configure Time Server 1 as blank, and then configure either Timer Server 2 or Time Server 3 with FQDN, the phone correctly switches to Time Server 2 and 3 as expected.
CLN06794	480i CT: A new feature was added to the 480i CT phone. When the phone is on a call and a second call comes in on Line 2, if the user hangs up on the first call, then the call that is ringing on L2 is automatically picked up.
CLN07500	The "Onhook Action" URI now initiates when the line on a call is terminated, regardless of whether the local handset, speakerphone, or headset hangs up or the far end handset, speakerphone, or headset hangs up.
CLN07721	Transferring a call or conferencing a call no longer gets interrupted if a call comes into the phone.
CLN07805	Pressing a mapped Conference key now sends DTMF digits within the active call using SIP INFO or RTP DTMF method.
CLN07911	Phones now have enough memory when using the BLF List feature with BroadSoft servers.
CLN10437	All Timezone settings are now correct.
CLN10449	BLA: The phone no longer terminates its own SUBS when a Sylanro server sends SUBS with Expire: 0 and terminates the subscription for BLA by sending NOTIFY in the subscription state.
DEF03800	After restoring factory defaults and rebooting the phone, the phone now uses the applicable DNS.
DEF03805	480i: Timer Server 1, 2, and 3 now get the applicable IP or FQDN.
DEF03917	9112i, 9133i: TUI Phone sip priority header is no longer blank. The TUI for both the 9112i and 9133i now shows "1 SIP" at the screen to enter the SIP Priority for the phones.
DEF04122	Phone no longer requires a power cycle if loading an XML greater than 5000 bytes.
DEF04144	When a user picks up the CT Base handset and the far end is using the CT handset, the audio-path now works as expected, and the speaker LEDs light up as correctly on the CT Base and the CT handset.
DEF06718	When the phone is on an active call and the user presses the Transfer key and dials another extension, and then the callers decides to cancel the transfer, pressing the Cancel key now works as expected and cancels the transfer.
DEF07000	If you change the download protocol parameter to a value other than "tftp" (i.e., http, https, ftp) in the configuration file, and then the phone gets its configuration parameters using mDNS auto-discovery, the HTTP, HTTPS, or FTP setting is correctly applied.
DEF07082	The Copy softkey for Directory and Callers List is now correctly disabled when the Directory and/or Callers List is disabled on the phone.
DEF07427	480i: The NOTIFY for event REFER from the phone is no long missing the Subscription-state header.
DEF07493	480i: The SIP Caller ID feature now works properly. When the "Update Caller-ID" is enabled, the phone no longer displays the information about the previous call.

Issues Resolved in Release 1.4.3

Issue Number	Description of Fix
DEF07625	9112i: If configuring a SIP outbound proxy in the configuration file, the value for that outbound proxy now displays correctly in the Aastra Web UI in the Outbound Proxy field instead of the Backup Proxy field.
DEF07883	480i, 480i CT: After pressing the "More" softkey to go to a second page on the LCD, the second page now stays on the display.
DEF07906	9133i: The phone no longer intermittently locks up, especially after a user checks voicemail on the phone.
DEF08154	Consulataive transfer through SBC now works properly.
DEF08212	9133i: Phone no longer factory defaults when turning off the power to the phone.
DEF08289	Using the volume control during an active call while using the handset or speakerphone, the LCD now correctly displays the volume setting for the Handset or Speakerphone for only 3 seconds and then disappears as expected.
DEF08596	9112i: The phone no longer has an improper error handling vulnerability that causes a denial of service.
DEF08597	9112i: The Format String vulnerability on the phone no longer causes a denial of service (DoS).
DEF09053	480i - DTMF is now correctly sent after call setup when using a speeddial softkey.
DEF09233	9133i: Phone now downloads firmware from the server on the first attempt without locking up.
DEF09417	9133i: Configured "Pickup" softkey now works as expected.
DEF09460	9133i: Phone now longer experiences sporadic "No Service" states.
DEF10044	9133i: The failover of an NTP time server now works as expected.
DEF10128	480i: Phones no longer lock up due to BLA Subscription termination from server. Memory status information has been added to the Troubleshoot option in the Aastra Web UI to be able to view memory status.
DEF10147	480i, 480i CT: The phone no longer locks up when pressing the delete key to delete an entry from the Caller's List, and then press the UP arrow within a couple of seconds after deleting the entry.
DEF10358	Copyright date is now updated in the firmware.
DEF10523	When 4 incoming calls are placed on hold on a phone, and the user then tries to place a call using a speeddial, the LCD now displays "No Line Available" as expected if no line is available. The Line Mgr now correctly checks to make sure a line is available before placing the call.
DEF10526	The tSipEngine task now works correctly while handling INVITE messages.
DEF10540	9133i: The Backup Registrar feature and Proxy Server feature now failover as expected.
DEF10570	Subsequently pressing the Transfer key and then the Redial key now works correctly as expected.
DEF10655	480i: The Phone no longer gets stuck at "50% Updating config" if corrupted .tuz file is being decrypted. If .tuz files are corrupted, the message "Bad encrypted file" displays to the phone and the phone continues to boot as expected.
DEF10663	9133i: SIP Phone BLA is now able to pickup held calls for all incoming calls when an outgoing call is attempted from the same line appearance.
DEF10674	480i: When a Directory entry with the Character ¿ (U00BF) is displayed in the Spanish language, the phone now allows you to change or delete the entry as expected.
DOC07523	The feature "Incoming call interrupts dialing" has been updated and the correct feature behavior is identified in this release note in the section, " Incoming Call Interrupts Dialing Feature (DOC07523) " on page 5 .

Incoming Call Interrupts Dialing Feature (DOC07523)

You can configure whether or not an incoming call interrupts an outgoing call that is dialing. The “**Incoming Call Interrupts Dialing**” (Web UI) parameter or “**incoming call cancels dialing**” (in config file) parameter controls this feature.

How it Works

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

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



Notes:

1. On the 9112i and 9133i, you must use the down arrow key to ignore the call. To answer the call you must press the line key where the call is coming in.
 2. For all models, if you disable this parameter (0=disable), and the phone receives an incoming call while you are dialing an outgoing call, you can pick up the call and perform transfer or conference as required.
 3. This feature works only if the User selects a line for which to dial out. It is recommended that the Administrator always keeps Live Dialpad ON in order for the User to have to select a line before dialing out.
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Transfer/Conference Call Behavior

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

Intercom Behavior

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

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

An Administrator can configure the “Incoming Call Interrupts Dialing” feature using the configuration file or the Aastra Web UI. A User can enable/disable the “Incoming Call Interrupts Dialing” feature using the Aastra Web UI only.

Reference

For more information about configuring and enhancements to the “Incoming Call Interrupts Dialing” feature, see the following IP Phone documentation:

- 2.1.1 Release Notes, Part Number 41-001029-00, REV04

Enhancements in Release 1.4.3

This section describes enhancements made to the 480i, 480i CT, 9112i, and 9133i SIP IP Phones in Release 1.4.3. The following table provides the enhancement number and a brief description of the enhancement.

Enhancement Number	Description of Enhancement
CLN09141	The IP phones now support manual setup of SIP traffic source port. For configuration information, see description below.

Configurable SIP Source Port

A System Administrator can now configure the SIP source port on the IP Phone. Previously, the IP phone used default values (**5060** for UDP/TCP). The new parameter for configuring the SIP source port is:

- **sip local port**

You can configure the SIP source port using the configuration files or the Aastra Web UI. **After configuring this parameter, you must reboot the phone.**

If Network Address Translation (NAT) is disabled, the port number also shows in the VIA and Contact SIP headers.

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




Note: By default, the IP phones use symmetric UDP signaling for outgoing UDP SIP messages. When symmetric UDP is enabled, the IP phone generates and listens for UDP messages using port 5060. If symmetric UDP signaling is disabled, the phone sends from random ports but it listens on the configured SIP local port.

Configuring SIP Source Port Using the Configuration Files

Parameter – <i>sip local port</i>	Aastra Web UI: Advanced Settings->Global SIP-> Advanced SIP Settings Configuration Files aastra.cfg, <mac>.cfg
Description	Specifies the local source port (UDP/TCP) from which the the phone sends SIP messages
Format	Numeric
Default Value	5060
Range	Greater than 1024 and less than 65535 Notes: 1. It is recommended that you avoid the conflict RTP port range in case of a UDP transport. 2. By default, the IP phones use symmetric UDP signaling for outgoing UDP SIP messages. When symmetric UDP is enabled, the IP phone generates and listens for UDP messages using port 5060.If symmetric UDP signaling is disabled, the phone sends from random ports but it listens on the configured SIP local port.
Example	sip local port: 5060

Configuring SIP Source Port Using the Aastra Web UI

Use the following procedure to configure the SIP source port using the Aastra Web UI..

 Aastra Web UI	
1	<p>Click on Advanced Settings->Global SIP->Advanced SIP Settings..</p> <div data-bbox="326 443 1170 1247" style="border: 1px solid black; padding: 5px;"> <p>Advanced SIP Settings</p> <p>Explicit MWI Subscription <input type="checkbox"/> Enabled</p> <p>Explicit MWI Subscription Period <input type="text" value="86400"/></p> <p>Missed Call Summary Subscription <input type="checkbox"/> Enabled</p> <p>Missed Call Summary Subscription Period <input type="text" value="86400"/></p> <p>Send MAC Address in REGISTER Message <input type="checkbox"/> Enabled</p> <p>Send Line Number in REGISTER Message <input type="checkbox"/> Enabled</p> <p>Session Timer <input type="text" value="0"/></p> <p>T1 Timer <input type="text" value="0"/></p> <p>T2 Timer <input type="text" value="0"/></p> <p>Transaction Timer <input type="text" value="4000"/></p> <p>Transport Protocol <input type="text" value="UDP"/> <input type="button" value="v"/></p> <p>Local SIP Port <input type="text" value="5090"/></p> <p>Registration Failed Retry Timer <input type="text" value="1800"/></p> <p>Registration Timeout Retry Timer <input type="text" value="120"/></p> <p>Registration Renewal Timer <input type="text" value="15"/></p> <p>BLF Subscription Period <input type="text" value="3600"/></p> <p>ACD Subscription Period <input type="text" value="3600"/></p> <p>Blacklist Duration <input type="text" value="300"/></p> <p>Whitelist Proxy <input type="checkbox"/> Enabled</p> </div> <p>“Local SIP Port” Parameter</p>
2	<p>The “Local SIP Port” field has a default value of 5060. Change this value if required to a value greater than 1024 and less than 65535.</p> <p>Note: It is recommended that you avoid the conflict RTP port range in case of a UDP transport.</p>
3	<p>Click <input type="button" value="Save Settings"/> to save your changes.</p>

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 480i, 480i CT, 9112i, 9133i

1.4.3 Release Notes

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