MODEL

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

SIP IP PHONE

RN-001033-00
Rev 01

Release Notes

Release 1.4.2

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Contents

SIP IP Phone Models 480i, 480i CT, 9112i, 9133i Release Notes 1.4.2

About this Document

This document provides information specific to the SIP IP Phone release 1.4.2. It includes the following information:

- General Information (release content, hardware supported, bootloader requirements, and upgrade notes)
- Changes in 1.4.2, Build 3000
- Changes in 1.4.2, Build 1081
- Known Anomalies in Release 1.4.2
- 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.

Model	Release Name	Release Version	Release Filename	Release Date
480i	Generic SIP	1.4.2	FC-000032-01-13	May 2007
480i CT	Generic SIP	1.4.2	FC-000040-00-13	May 2007
9133i	Generic SIP	1.4.2	FC-000046-01-13	May 2007
9112i	Generic SIP	1.4.2	FC-000058-01-13	May 2007

Hardware Supported

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

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

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

Changes in 1.4.2, Build 3000

This section describes the feature enhancements, and issues resolved, in Release 1.4.2, build 3000.

Enhancements to "Incoming Call Interrupts Dialing" Feature

The "**Incoming Call Interrupts Dialing**" feature has been enhanced to work more efficiently.

In previous releases, if the "**Incoming Call Interrupts Dialing**" feature was disabled, and you were dialing out on your phone, and you received an incoming call at the same time, the incoming call would go to an available line and the LED would blink to let you know where the call was placed by the phone. The Caller ID would display on the LCD and the number you were dialing disappears. If you wanted to continue dialing out, you would have to press the Line key for which you were originally dialing out on. If "Incoming Call Interrupts Dialing" is enabled, the incoming call interrupts your dialing sequence and displays to the phone's LCD for you to answer.

In Release 1.4.2, Build 3000, the behavior for "Incoming Call Interrupts Dialing" has been modified. Now when this feature is disabled, the incoming call goes to an available line, the LED blinks, but the LCD screen still displays the number you were dialing. When this feature is enabled, it performs the same as in previous releases.

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”** 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”** 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.

Issues Resolved in 1.4.2, Build 3000

This section describes the issues resolved in Release 1.4.2, build 3000. The following table provides the issue number, and a brief description of each fix.

Issue Number	Description of Fix
CLN07721	The “Incoming Call Interrupts Dialing” feature has been enhanced to work more efficiently. For example, if the phone receives an incoming call while you are dialing an outgoing call, and “Incoming Call Interrupts Dialing” is disabled, the incoming call goes to an available line, and the LED blinks. However, the phone UI continues to display the number you are dialing. (Incoming calls do not take precedence over outgoing calls.)
CLN07805	If you place a call on hold, and then press a Conference softkey, the call now remains on hold; the phone does not send dtmf digits over this line.
DEF07059	When you press the Mute button on the IP phone, the phone UI now displays “Mute,” and the LED flashes, notifying you that the mute feature is now active. Prior to this release, the phone UI would display “Connected.”

Issue Number	Description of Fix
ENH07628	If the subscription service for BLF is terminated (for example, if a server goes down and back up), the IP phone now attempts to automatically resubscribe. You no longer have to reboot the phone in order to initiate BLF subscription. Before this fix, if the server went down, and the subscription service for BLF was terminated, you had to reboot the IP phone in order to resubscribe. BLF was lost until the IP phone successfully rebooted.
CLN07911	Fixed the problem of the IP phone occasionally crashing if the BLF list was configured while parsing incoming BLF list NOTIFY messages.
ENH07628	Prior to this release, if for some reason the dial plan was misconfigured, the IP phone would still attempt to use it when rebooting. As a result, the phone could not reboot successfully. With this release, the IP phone ignores a misconfigured dial plan, and instead reboots using the default dial plan.

Changes in 1.4.2, Build 1081

This section describes the changes made, and issues resolved, in Release 1.4.2, build 1081.

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

Issue Number	Description of Fix
CLN05048	XML: AastraIPPhoneTextMenu object now supports turning off automatic numbering of items. You turn off automatic numbering of items by adding the attribute “style=none” to the root tag of the AastraIPPhoneTextMenu object. For example: **<AastraIPPhoneTextMenu style=”none”>** **<Title>Example</Title>** **<MenuItem>** **<Prompt>Item 1</Prompt>** **<URI>10.50.10.49</URI>** **</MenuItem>** **</AastraIPPhoneTextMenu>**
CLN05071	Added support for virtual web servers, by adding the “Host” header to all XML GET requests from the phone.
CLN05088	Redial list now remembers pause key presses.
CLN05089	Redial list is now accessible during active calls.
DEF04183	Sidetone is now only enabled during calls.
DEF04448	SIP: The contact URI is now used in subsequent transfer requests.

<table>
<tr><th>Issue Number</th><th>Description of Fix</th></tr>
<tr><td>DEF04481</td><td>Added an additional VLAN mode which enables the phone to be on a VLAN. All untagged packets are sent to the passthrough port.

To enable this mode, you enable tagging on the phone port as normal, but set the passthrough VLAN ID to 4095,.

For example, the following configures the phone to be on VLAN 3 but the passthrough port is configured as untagged.

tagging enabled: 1
VLAN id: 3
VLAN id port 1: 4095</td></tr>
<tr><td>DEF04929</td><td>When there is an active call, the speeddial keys now send DTMF digits through the active voice path. To dial out, you now have to first put the active call on hold and press the speeddial key.</td></tr>
<tr><td>DEF05224</td><td>The “Action URI Registered” is now executed on the first successful registration of each unique line configured on the phone.</td></tr>
<tr><td>DEF05722</td><td>Increased the length of the stuttered dial tone used to indicate message waiting.</td></tr>
<tr><td>DEF05964</td><td>Usernames containing dots (“.”) are now supported.</td></tr>
<tr><td>DEF06238</td><td>VLAN: The SIP and RTP DSCP default values are now set according to RFC4504.

<table><tr><th>Parameter</th><th>Old Default</th><th>New Default</th></tr><tr><td>tos sip</td><td>26</td><td>24</td></tr><tr><td>tos rtp</td><td>46</td><td>32</td></tr></table></td></tr>
</table>

<table>
<tr><th>Issue Number</th><th>Description of Fix</th></tr>
<tr><td>ENH05218</td><td>A User and Administrator can now configure whether or not an incoming call interrupts an outgoing call that is dialing. A new parameter has been added (“incoming call interrupts dialing”) to control this feature.

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. This was the existing default behaviour in the previous 1.4.1 release and is also the behaviour of this parameter in 1.4.2 if it is enabled.

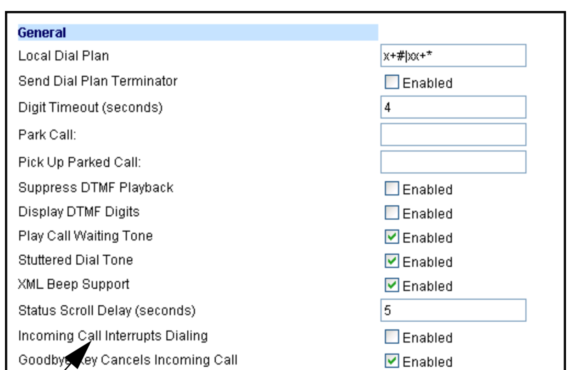
IMPORTANT NOTE: If you disable this parameter, the behaviour of the phone has changed in 1.4.2. In 1.4.2, this feature performs as follows if disabled:

When you disable the “incoming call interrupts dialing” parameter (0 = disable), which is the default, the phone does not interrupt the outgoing call during dialing and instead rings the incoming call on another free line (or sends busy signal if all remaining lines are busy). You have a choice to ignore the incoming call, or answer the incoming call on another line. 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.

On the 480i, 480i CT, and 9133:
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.

On the 9112i:
If you disable this parameter (0=disable), and the phone receives an incoming call while you are dialing an outgoing call, you can pickup the call but you cannot transfer or conference that call.

An Administrator can set this parameter using the configuration files (incoming call interrupts dialing) or the Aastra Web UI (Incoming Call Interrupts Dialing) at Basic Settings->Preferences->General). A User can set this parameter using only the Aastra Web UI.



Incoming Call Interrupts Dialing (in Web UI) (Default is disabled.)</td></tr>
</table>

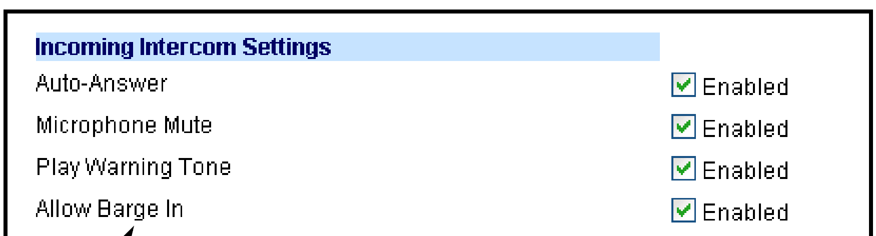
<table>
<tr><th>Issue Number</th><th>Description of Fix</th></tr>
<tr><td>ENH05637</td><td>The IP phones no longer support the alpha character in the password.

Non-numeric passwords are ignored in release 1.4.2 and all subsequent releases. The default password is used instead.</td></tr>
<tr><td>ENH05953</td><td>Release 1.4.2 includes the support of new changes in the North American Daylight Savings Time starting March 11, 2007.</td></tr>
<tr><td>ENH06153</td><td>A User and Administrator can now configure a “barge-in” feature on the phone.

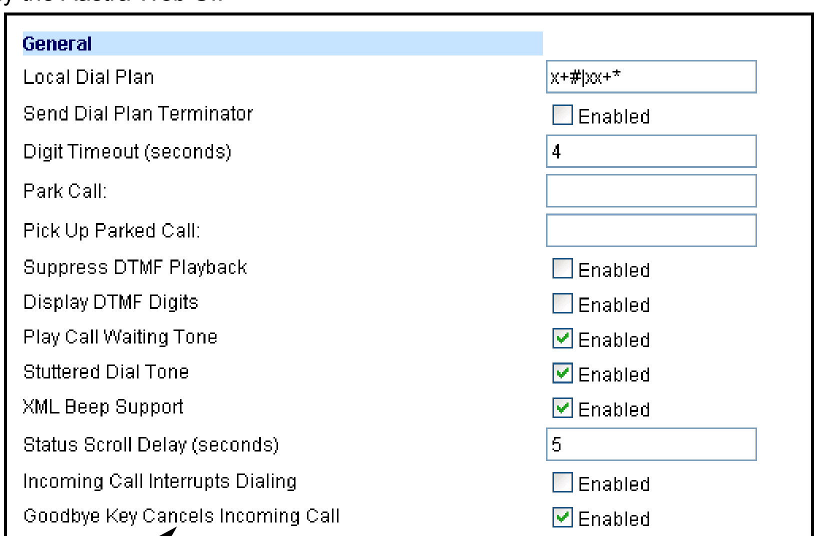
A new parameter has been added (“sip intercom allow barge in”) which controls how the phone handles an incoming intercom call when an active call is in progress.

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.

An Administrator can set this parameter using the configuration files (sip intercom allow barge in) or the Aastra Web UI (Allow Barge In at Basic Settings->Preferences->Incoming Intercom Settings). A User can set this parameter using only the Aastra Web UI.



Allow Barge In (in Web UI) (Default is enabled.)</td></tr>
<tr><td>ENH06301</td><td>REFER-TO header now uses uppercase characters in escape codes.</td></tr>
</table>

<table>
<tr><th>Issue Number</th><th>Description of Fix</th></tr>
<tr><td>ENH06372</td><td>
A User and Administrator can now configure the Goodbye key to drop active calls or ignore incoming calls.
A new parameter has been added (“goodbye cancels incoming call") which controls the behavior of the goodbye key when the phone is in an active call and a second call is presented to the phone.
When you enable this parameter (1 = enable), which is the default, the Goodbye key rejects the incoming call. When you disable this parameter (0 = disable), the Goodbye key hangs up the existing call.
If you enable this parameter, and the phone receives another call when an active call is already present, the phone displays softkey 1 as “answer” and softkey 2 as “ignore” (on the 480i and 480i CT; the 9112i and 9133i displays “answer” and “ignore” using the DOWN arrow key).
An Administrator can set this parameter using the configuration files (goodbye cancels incoming call) or the Aastra Web UI (Goodbye Key Cancels Incoming Call at Basic Setting->Preferences->General). A User can set this parameter using only the Aastra Web UI.


Goodbye Key Cancels Incoming Call (in Web UI) (Default is enabled.)
</td></tr>
<tr><td>ENH06544</td><td>9112i and 9133i: These model phones now have extended SPRE support.</td></tr>
</table>

Issues Resolved in 1.4.2, Build 1081

This section describes the issues resolved in Release 1.4.2, build 1081. 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
CLN04782	XML: "alert" status message type now displays for the correct amount of time.
CLN04786	After failover to backup/primary proxy, the subscription no longer terminates after 3 minutes.
CLN04787	When failing over to the backup proxy, the NTFY from the server now changes the line state.
CLN04793	480i CT: Answer softkey now displays when an XML screen is interrupted by an incoming Intercom call from the handset.
CLN05043	Turned off caching of WebUI page due to problems with pages not displaying correctly all of the time.
DEF05019	XML: “Cancel” and “Done” softkeys now show up in correct places in the AastraIPPhoneInputScreen object.
CLN05072	9133i/9112i: Fixed possible crash when unconfigured programmable keys are pressed.
CLN05079	XML: Loading an XML page now cancels volume change ringer.
CLN05189	480iCT: Fixed possible crash when first call to base after a reset was an intercom call from the handset.
CLN05223	XML: Fixed intermittent crash when parsing AastraIPPhoneDirectory objects.
CLN05229	Broadsoft SCA: Incoming calls are now presented on the correct SCA appearance.
CLN05287	Sylantro BLA: Fixed rare crash caused by a race condition when seizing a line.
CLN05325	Sidetone no longer disabled after returning from Call Hold.
CLN05390	List of mDNS servers now do not display .local portion.
CLN06230	Fixed intermittent crash when rebooting the phone via XML or via the autosync feature.
DEF02980	Asterisk: Fixed crash when an INVITE message is received after the SIP dialog has been closed.
DEF03715	Incoming Intercom calls no longer force existing calls on hold.

Issue Number	Description of Fix
DEF03717	Incoming Intercom calls no longer go to headset when an existing speakerphone call is on hold.
DEF04448	URIs are now used in subsequent REFER-TO addresses.
DEF04717	9133i: The Message Waiting Indicator (MWI) light now continues to flash after receiving a call.
DEF04970	Hanging up the handset now correctly executes the “Action URI Onhook”.
DEF05047	Fixed crash if user incorrectly pressed park key when the phone is idle.
DEF05056	9133i: Phone with a call on Hold no longer receives Intercom call with the speaker turned off.
DEF05098	480i CT: Intercom call no longer plays over an existing call.
DEF05145	Line softkeys now initiate dialing of typed number when pressed.
DEF05182	XML: destroyOnExit=yes now works with the Refresh header.
DEF05269	480iCT: When a user is on the Base phone and another user is on it’s cordless handset, and two calls come into the phone, putting one call on hold no longers causes one-way audio on the other line.
DEF05293	It is no longer possible to corrupt the time server information from the Options menu, which caused the phone to lockup during the next boot cycle.
DEF05472	Fixed a crash when processing a NOTIFY message which terminated the subscription but did not have the option reason field.
DEF05570	480iCT: When the phone is idle, pressing the left arrow button no longer causes the LCD to display blank and the phone no longer locks up.
DEF05873	ININ BLA: There is no longer a 2 second delay when picking up a held call.
DEF05890	9133i: The programmable key LED is now turned off if DND is disabled during an active call.
DEF05978	Use the correct Contact header in 200 OK responses to NOTIFY messages.
DEF06101	Fixed registration problems when phone is configured to use both UDP and TCP. The phone now sends a single Contact in the REGISTER message, but without a transport parameter.
DEF06118	Call-Info answer-after parameter and picking up the handset no longer opens up a new line.
DEF06127	Fixed a crash on startup caused by long dial plans with prepend rules.
DEF06411	Phones no longer lose BLF subscriptions when Asterisk sends an unrecognized NOTIFY message.

Issue Number	Description of Fix
DEF06417	The "**sip update callerid**" parameter in the configuration files now works correctly on all phone models.
DOC06909	In Appendix A of the 1.4.1 Administrator Guide Revision 10, the following corrections were made and can be found in the 1.4.1 Administrator Guide Revision 11: **Configuration Parameter** — **Correction** log ip — log server ip log port — log server port call forward disabled — Example read as "callers list disabled". Now reads "**call forward disabled**". vlan id — VLAN id vlan id port — VLAN id port

Known Anomalies in Release 1.4.2

This section describes the known anomalies in release 1.4.2.

Note: Unless specifically indicated, these known anomalies apply to all phone models.

Issue Number	Description
DEF04535	480i and 480i CT: The TO field displays your own name during an originated call. When you place a call, and the other side answers, your phone screen displays your own line's name in the "TO" field.

DEF04899	In XML APIs, password is not hidden when it is marked as not editable. When you create an "**AastraIPPhoneInputScreen**" input object that has password set to "yes" and editable set to "no", the password shows up on the display but cannot be edited.
	
	Example:
	
	`<AastraIPPhoneInputScreen type = "string" password="yes" editable="no">`
	`<Title> Input </Title>`
	`<Prompt> Enter something </Prompt>`
	`<URL> http://10.50.10.61/screen.xml </URL>`
	`<Parameter> param </Parameter>`
	`<Default>Hello</Default>`
	`</AastraIPPhoneInputScreen>`

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

1.4.2 Release Notes

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