Aastra 480i Series SIP Phone (CT version w/ Cordless Handset)

1 Important Notes

- Check the *SIP 3rd Party Validation Website* for current validation status. The *SIP 3rd Part Validation Website* can be viewed at:

  http://testlab.inin.com or http://testlab.vonexus.com

2 Vendor Documentation

Documentation on the Aastra website:

http://www.aastra.com/cps/rde/xchg/SID-3D8CCB73-094B9C52/04/hs.xsl/19466.htm#dl_instructions

3 Validated Software Version

Firmware:	1.4.1.1077
Bootrom:	1.1.0.4

4 Install

Download the Aastra 480i CT files from the 3rd Party Validation website:

http://www.testlab.inin.com or http://www.testlab.vonexus.com

Contained in the zip file will be the validated version of firmware as well as any supplemental configuration files, and sample .cfg files. As of right now there are no sample files provided by Aastra, however there is a sample file in the Administrator guide starting on page 188 that can be pasted into a file as a starting point. The doc includes the overarching, and individual phone sample configs.

5 Configuration

Methods:

- Web interface - There are many advanced options that are exposed in the web interface. Caution should be exercised and the Aastra 480i CT documentation should always be referenced when using the web interface configuration option. Please note that if it is desired to use the .cfg file method then *no configurations should be made via the web interface*. These will overwrite anything that may come from an .cfg file, and the only way to be sure the web options are not overwriting .cfg file options is to reset the phone to factory defaults.
- Manipulation of the .cfg file, then uploading it via TFTP/FTP/HTML. The HTML method was not tested during validation. This was the method used to configure the unit during validation.

Initial Setup:

- Unzip the ZIP file containing the Aastra 480i CT configuration files and firmware.
- Navigate the phone menu to *Network* (selection 9 at the time), and then *Download Protocol* (selection 6). Select the desired protocol and choose done. This will ask for a password, the default being **22222**.
- Then configure the *TFTP/FTP/HTTP Server* information using the subsequent menu options (selections 7, 8, and 9 respectively).
- The phone may then ask to restart. Do not do so until the firmware and configuration steps have been finished. It will not break anything, but will save some time as there will not yet be anything for the phone to pick up software or configuration related.

Download Current Firmware:

- Extract the *phone version*.st file from the ZIP file, or from the *testlab* site. Copy this to the root of the TFTP directory specified in the initial setup section. The phone will copy and install the firmware upon next reboot.

Changing the Configuration:

- Prior to making any changes, the Aastra 480i CT documentation should be consulted for information on configuration parameters, options and functions.
- Changes can be made via the web interface or .cfg file (again, if the .cfg file is to be used, the web interface should not).
- Aastra provides 2 configuration option files for its phones. One is a file called aastra.cfg, which provides a base overview of configurations for all phones, this is loaded first. The second is *MAC Address*.cfg, which is specific to each phone, and will overwrite any duplicate parameter information from the aastra.cfg file.
- Description of more significant parameters. This is not a comprehensive list of all parameters found in the .cfg file. A **bold** face parameter name indicates that it should be changed to represent specific site information.
- Once the configuration has been entered, the phone will need to be restarted to upload the configuration information.

Parameter	Description
DHCP:	Enables DHCP on the phone. *Values*: 1 = enable 0 = disable
web interface enabled:	Enables the phone's web interface. *Values*: 1 = enable 0 = disable
two call support:	This enables support for 2 calls for the same line (this is only available on the 480i model phones).

	Values: 1 = enable 0 = disable
callers list disabled:	Provides a history of calls. *Values*: 1 = enable 0 = disable Note in this case the disable value allows the feature to function.
call forward disabled:	Enables call forwarding on the phone. *Values*: 1 = enable 0 = disable Note in this case the disable value allows the feature to function.
missed calls indicator disabled:	Enables the phone to indicate when a call has been missed. *Values*: 1 = enable 0 = disable Note in this case the disable value allows the feature to function.
redial disabled:	Enables the redial function to be used. *Values*: 1 = enable 0 = disable Note in this case the disable value allows the feature to function.
conference disabled:	Enables the conference function to be used. *Values*: 1 = enable 0 = disable Note in this case the disable value allows the feature to function.
call transfer disabled:	Enables the conference function to be used. *Values*: 1 = enable 0 = disable Note in this case the disable value allows the feature to function.
sip use basic codecs:	Enables the use of the basic codecs. Basic codecs are defined as G.711uLaw, G.711aLaw, and G.729). These can be customized via the Customized Codec Preference List, which is outlined in the Aastra documentation. *Values*: 1 = enable 0 = disable
sip dtmf method:	Defines the method by which DTMF tones will be sent from the phone. *Values*: 0 = RTP 1 = SIP (RFC2833) 2 = Both Note that to completely force RFC2833 only, the sip out-of-bad perameter should be used, which is outlined in the Aastra documentation.
The following SIP line information can be done as a general sip line entry, or to define different values per line, depending on the setup. If there is to be just a general entry, then the header should read: sip *config option: value*, however if it is to be for a given line, it should read: sip *line# config option: value*, with the # being the line number in question.	
sip <line#> auth name:	Value for username if authentication is being applied to the line.
sip <line#> password:	Value for the password if authentication is being

	applied to the line.
sip <line#> mode:	Selects mode for line operation. *Values*: 0 = Generic (recommended for basic operation) 3 = BLA (shared/bridged line) The other values do not apply to IC.
sip <line#> user name:	Value for the identification SIP address for IC's dynamic updating of the phone's address.
sip <line#> screen name:	Value for the name displayed on the screen.
sip <line#> proxy ip:	Value (as an IP address, or FQDN… the IP address is recommended) of the proxy to be used (e.g. SIP proxy, IC server, etc…).
sip <line#> backup proxy ip:	Value (as an IP address, or FQDN… the IP address is recommended) of the proxy to be used (e.g. SIP proxy, IC server, etc…) when the primary is not active.
sip <line#> proxy port:	Value of the port to be used to talk to the proxy via SIP, typically 5060.
sip <line#> backup proxy port:	Value of the port to be used to talk to the backup proxy via SIP, typically 5060.
sip <line#> registrar ip:	Value (as an IP address, or FQDN… the IP address is recommended) of the registrar to be used (e.g. SIP proxy, IC server, etc…). Default is 0.0.0.0 which will cause the phone not to attempt to register.
sip <line#> backup registrar ip:	Value (as an IP address, or FQDN… the IP address is recommended) of the registrar to be used (e.g. SIP proxy, IC server, etc…) when the primary is not active. Default is 0.0.0.0 which will cause the phone not to attempt to register.
sip <line#> registrar port:	Value of the port to be used to talk to the registrar via SIP, typically 5060.
sip <line#> backup registrar port:	Value of the port to be used to talk to the backup registrar via SIP, typically 5060.

6 BLA (bridged/shared lines)

The following configuration options are required for using the BLA feature of the Aastra series phones. There is also a description in the Aastra documentation. These options follow the same basic pattern as those above, but will require the line information in the parameter for the BLA line.

Parameter	Description
sip line# proxy ip: **sip line# proxy port:** **sip line# registrar ip:** **sip line# registrar port:** **sip line# mode:**	These must all be configured for the line to be used with BLA. The information is the same as the standard line found above. Note: The mode must be set to 3 to enable BLA.
For the primary phone. **sip line# user name:**	Value of the Identification SIP Address for the primary "sharer" of the line. BLA is done through establishing a line and having others connect off a variant of the user name. For example 1010.
For the secondary phone. **sip line# user name:**	Value of the Identification SIP Address for the secondary "sharee" of the line. To connect the BLA line, a variant of the original name should be used, for example 10100.
For both phones. **sip line# bla number:**	Value for the BLA identification number of the line. This value should be the same on all phones and ties the lines together.

Once the preceding values have been configured and uploaded to the phone, appropriate appearance on/for values will need to be set in IC using the user name values in the respective places for the Identification SIP Address.

7 Soft Keys

The Aastra phone contains many soft key options, which can be set to a variety of functions. Typical uses include voice mail, do not disturb, and speed dial features. These can all be set in the configuration files according to business needs. The Aastra has a complete listing of functions applicable to the soft keys, and instructions on how to implement them in the configuration file(s).

8 Cordless Handset

The CT version of the Aastra 480i series phone has a cordless handset. The handset does not communicate directly to the IC server, but interfaces via the 480i CT base station to work with SIP. As such it has some of the functionality of the base station (such as redial, call lists, mute, etc…), and must be paired with the base station to function properly.

To pair the handset the pairing option must be selected from the handset menu (selection 7 at the time of writing), and the base station (selection 12) at the same time. They will then pair, and the handset will use that base station to communicate via SIP (i.e. if a call is made on the handset, it will occupy one of the lines configured on the base station).

Multiple handsets can be paired to the same base station, just carry out the process above with each.

