Aastra 9133i

Asterisk Interoperability Report
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Asterisk Interoperability Reports describe the certification testing performed by Digium on the speicified product and Asterisk Business Edition. Each Supported feature of the device under test is described as well as how the device was configured to work with Asterisk during testing.

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SIP Device Summary

Make:	Aastra9133i
Firmware:	1.4.0.1048
Tested With:	Asterisk B.E. B.1.1

Product Description

The Aastra 9133i is a versitile SIP phone featuring a high-contrast screen with 7 configurable hardkeys that can set up to act as shortcuts to menu actions, launcher for custom xml applications as well as speed dial extensions with optional SIP presence.

Features Tested and Confirmed Working

- **Call Hold and Retrieve**
- **Call Waiting**
- **Call Transfer and Divert**
- **Other Party Identification (Caller ID)**
- **Conferencing**
- **Call History**
- **Do not Disturb**
- **Message Waiting Identification (Voicemail Alerts)**
- **Call Forwarding**
- **SIP Presence / Busy Lamp Field (Requires Asterisk B.E. Version B.1)**

Asterisk Configuration

For the basic configuration of a SIP device within Asterisk requires the configuration of three configuration files: sip.conf for setting up the SIP device channel (including registration information, channel name, etc.), extensions.conf (for configuring SIP device extension), and voicemail.conf (for configuration of voice-mailbox). The following code snippets were used to configure the Aastra 9133i for interoperability testing.

sip.conf

```
[aas9133i]
type=friend
context=sip-phones
username=aas9133i
secret=blah
host=dynamic
mailbox=9133@default
defaultip=192.168.0.75
dtmfmode=rfc2833
```

voicemail.conf

```
9133 => 5555,Aastra9133i,<email>
```

extensions.conf

Using old=style n+101 extensions:

```
[sip-phones]
...
exten => 9133,1,Dial(SIP/aas9133i,15)
exten => 9133,2,VoiceMail(u9133)
exten => 9133,3,Hangup
exten => 9133,102,VoiceMail(b9133)
exten => 9133,103,Hangup
...
```

Using stdexten macro:

```
[sip-phones]
...
exten => 9133,1,Macro(stdexten,9133,SIP/aas9133i)
...
```

Hints for SIP presence:

```
[buddypres]
...
exten => 9133,hint,SIP/aas9133i
exten => 9133,1,Macro(line,${aas9133i})
```

SIP Device Configuration

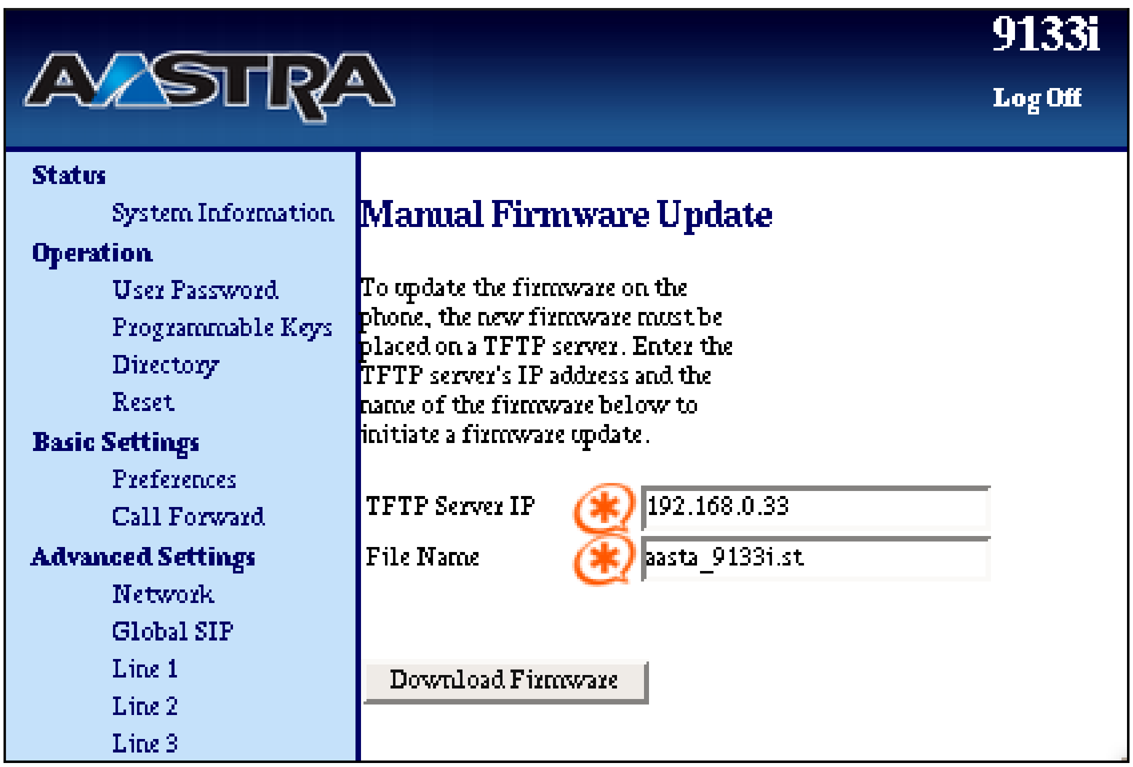
Configuration overview:

The 9133i can be configured in any of these ways:
1.Navigating the web configuration interface
2.Editing phone configuration files on the TFTP server
3.Configuring the phone through the phone's internal menu system

Web Configuration

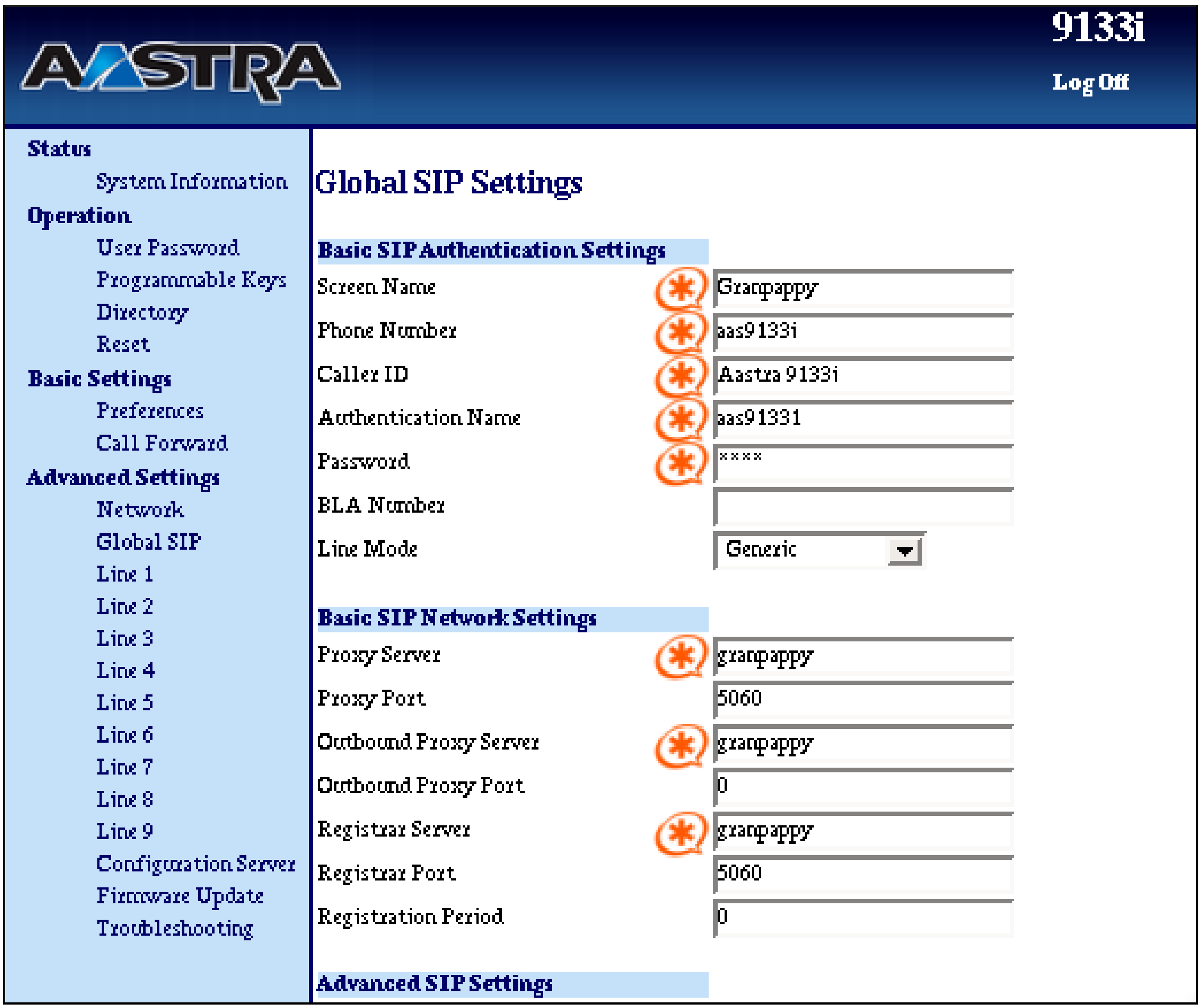
The most vital configurations to the phone can be made easily through the phone's web interface. The 6 sections that must be configured are: Firmware, Global SIP, Network Settings, Configuration Server, SoftKeys and XML, and the individual Line configuration pages.

Firmware



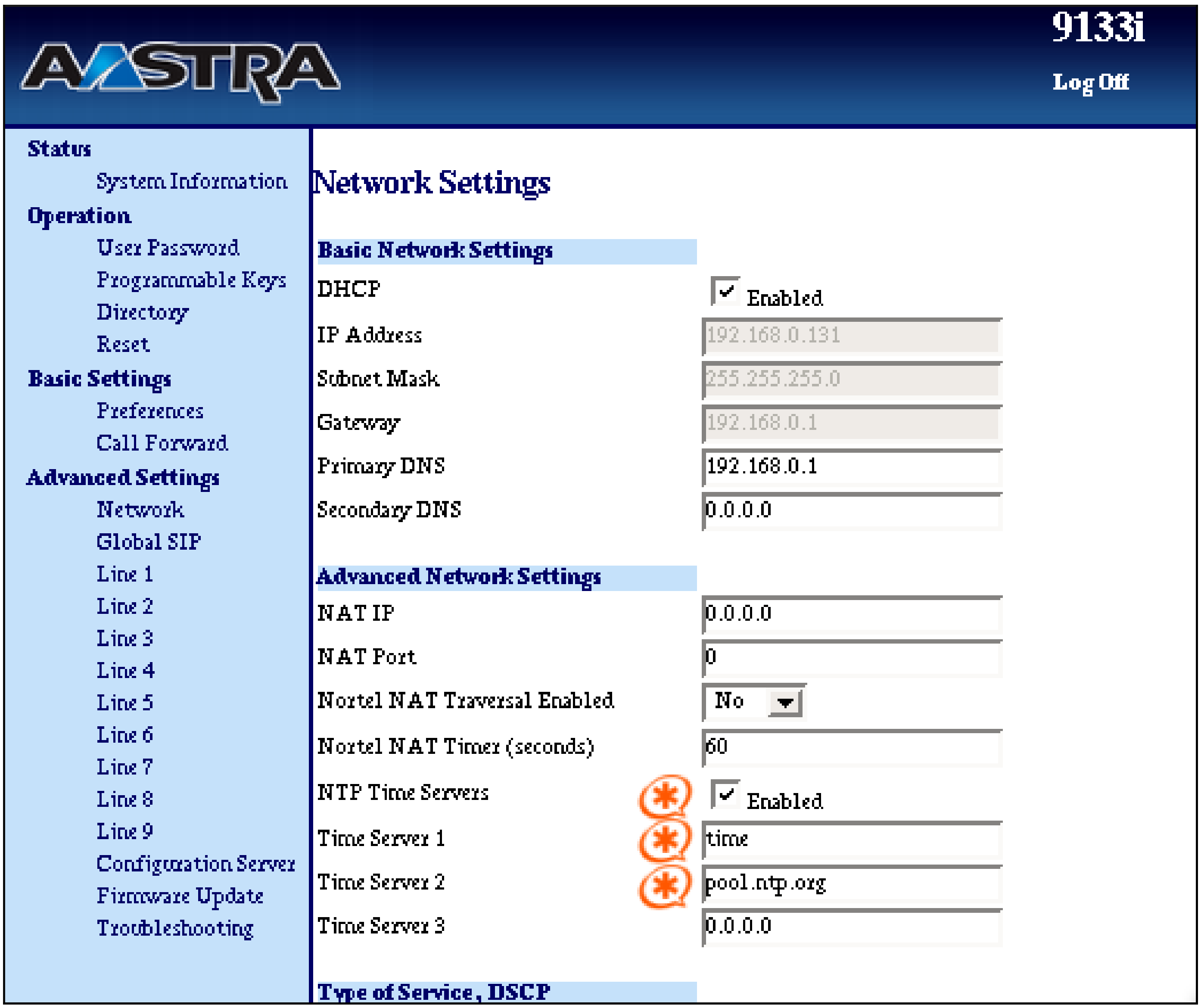
Global SIP

Settings in this section are not strictly necessary since they can be set in the individual Line settings page for each “line” the phone registers to and uses; however, since in many cases the authentic user name and password, Caller ID information, etc. remain the same on all Lines, it is helpful to complete this configuration page.



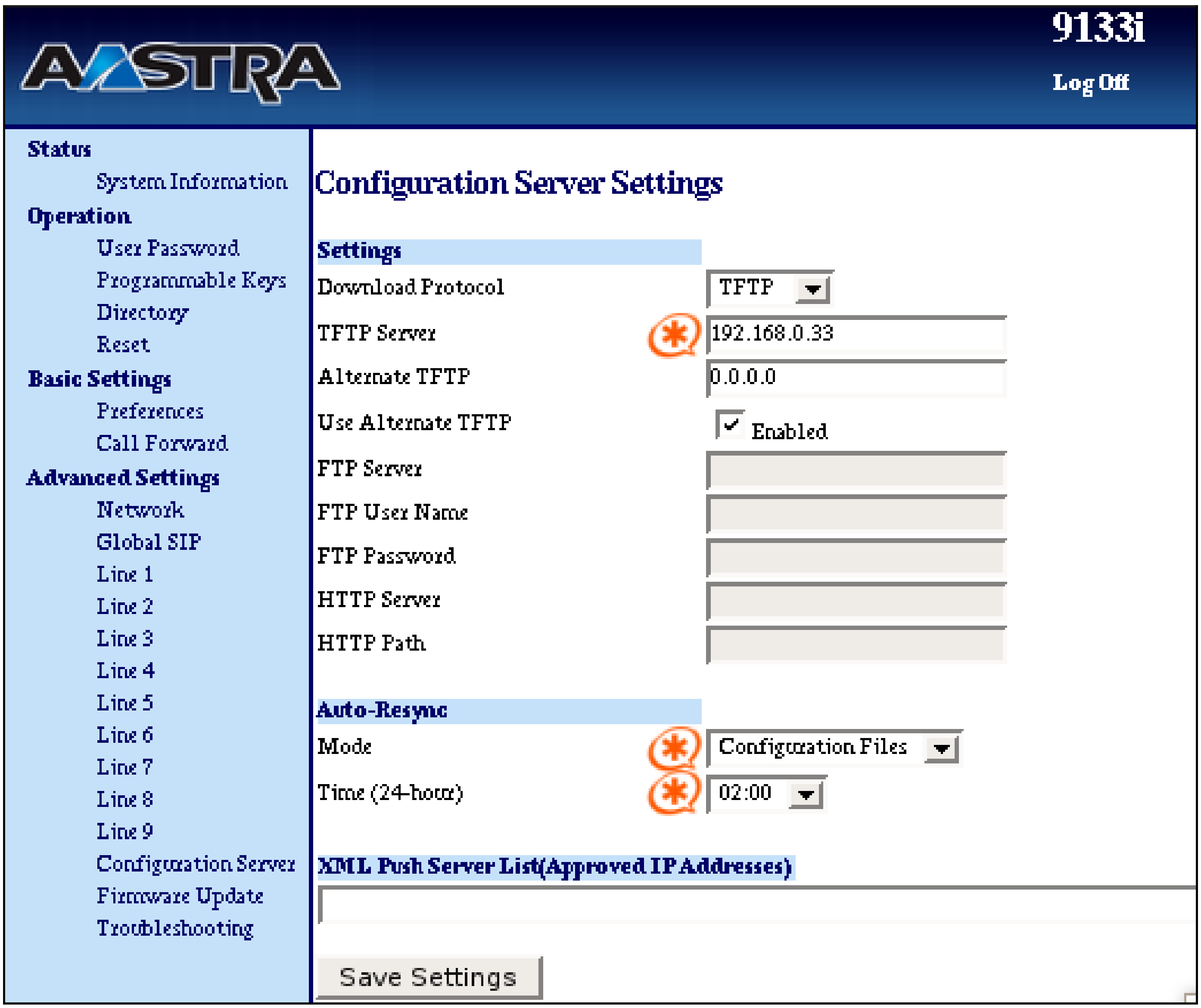
Network Settings

The only configuration necessary to make in this section is enabling NTP and specifying the NTP server to connect to. Unless of course one needs to specify a static IP, DNS information, NAT, etc.



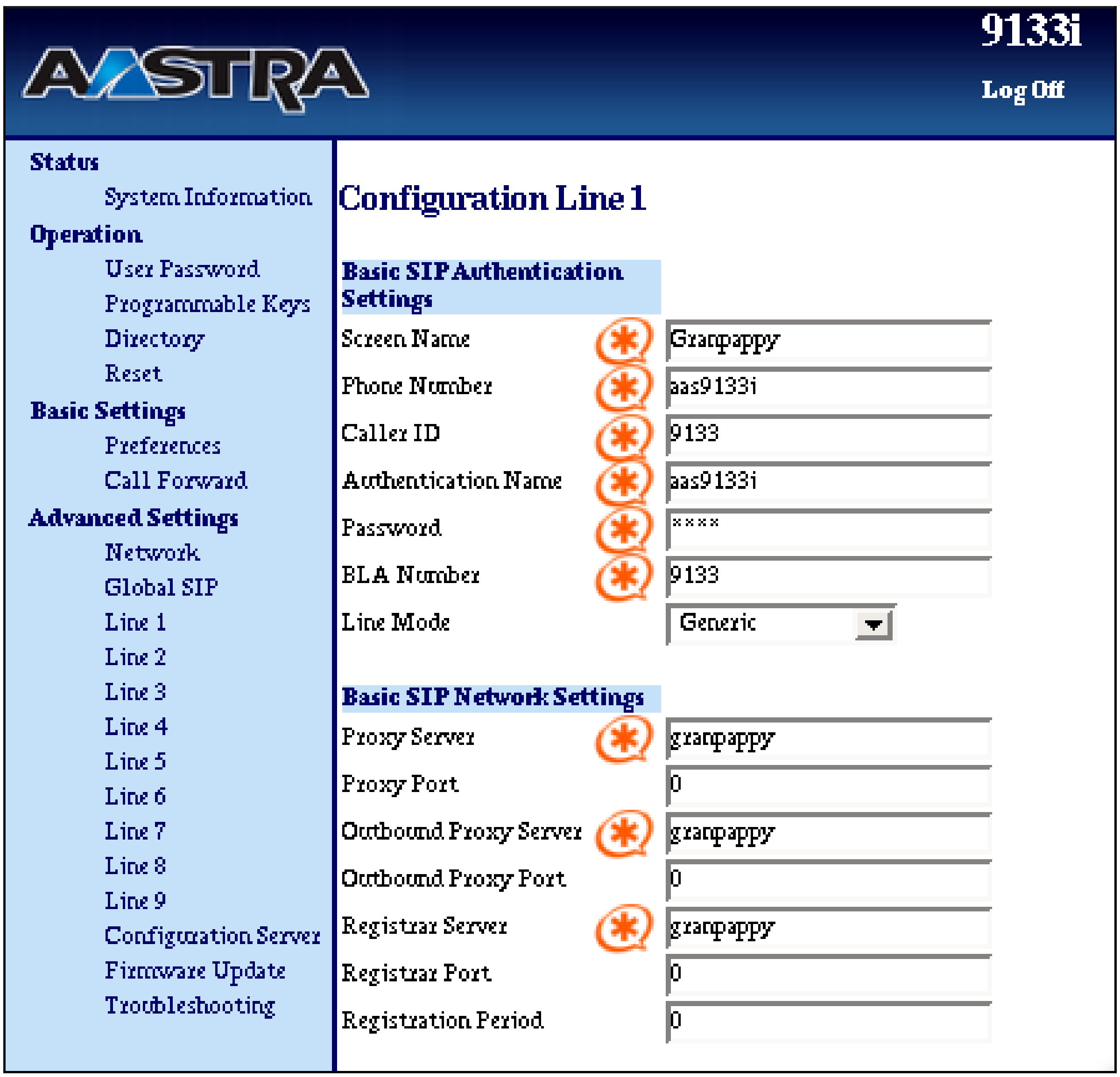
Configuration Settings

This page allows one to configure the TFTP (or FTP) server.



Line Settings

These pages (Line 1 – Line 9) are used to configure registration and network settings for individual Lines.



Softkeys and XML

This page is used for configuring the SoftKeys (used for holding extensions, lines, etc.) and for configuring the phone to utilize XML applications.

Phone Configuration Settings

There are two configuration files one can edit in order to configure the Aastra 9133i. First is the aastra.cfg file, which contains the settings for all aastra phones utilizing that TFTP server, the second is the <mac>.cfg (where <mac> is the phone's mac address) which contains phone specific settings. Below are samples of said configuration files.

aastra.cfg

```
# Aastra Telecom Inc.
# Common settings for all Aastra IP Phones.  Any parameter
listed in this file can be
# overwritten by the same parameter in <mac>.cfg file.

#Time Server Settings
#time server disabled: 0
#time server1: pool.ntp.org

#Sip Settings
sip proxy ip: 0.0.0.0
sip proxy port: 5060
sip registrar ip: 0.0.0.0
sip registrar port: 5060
sip outbound proxy:
sip outbound proxy port:

sip registration period: 3600
sip registration retry timer: 30

sip use basic codecs: 1
sip line1 vmail:

#sip intercom type: 2
#sip intercom prefix code: *55

web interface enabled: 1

#Daily Resync of cfg and firmware files
#auto resync mode: 3
#auto resync time: 23:30

#Server-based directory download
#directory 1: mydirectory.csv
```

<mac-address>.cfg

```
# Aastra Telecom Inc.
# This file contains specific settings for the phone with the
MAC
# address for which this file was named.  Settings which have
already
# appeared in aastra.cfg will be overridden by those in this
file.


#line info
#line 1
sip line1 auth name: aas9133i
sip line1 password: blah
sip line1 mode: 0
sip line1 user name: aas9133i
sip line1 display name: Aastra9133i
sip line1 screen name: Aastra9133i

# Softkey Settings
softkey1 type: speeddial
softkey1 label: Voicemail
softkey1 value: 8500
softkey1 line:
softkey1 states: idle

softkey2 type: dnd
softkey2 label: DND
softkey2 value:
softkey2 line:
softkey2 states: idle

softkey3 type: blf
softkey3 label: Snom 360
softkey3 value: 3600
softkey3 line: 1
```

Test Reports

The following test reports give an overview of the tests performed, as well as their objectives and expected and actual results.

Hold and Retrieve	
Test Objective:	Verify that a call can be placed on hold, other calls can be made, and the original call can be retrieved.
Procedure:	Place a call to the 9133i and place the calling party on hold. Then from the 9133i call out to another party, disconnect newest call and retrieve the call on hold.
Expected Results:	The call will be placed on hold and can be retrieved whenever.
Actual Results:	As expected.
Status:	**Pass**

Call Waiting	
Test Objective:	Verify that call waiting is functional, allowing a new call to be answered by placing existing conversing party on hold.
Procedure:	Place a call to the 9133i and answer it, with another device call the 9133i. Place the first calling party on hold the answer the new call. Hangup (or place on hold) and resume the conversation with the first calling party.
Expected Results:	The original caller will be on hold until new caller is disconnected or put on hold itself.
Actual Results:	As expected.
Status:	**Pass**

Transfer and Divert	
Test Objective:	Verify transferring calls works using the transfer button on the SIP phone.
Procedure:	Place a call to the 9133i during the conversation press "Xfer" dial the number of the party to which you will be transferring the call, then after connection is established with said party, press "Xfer" once more to complete the transfer.
Expected Results:	The call will be successfully transferred via the attended transfer method.
Actual Results:	As expected.
Status:	**Pass**

Other Party Identification	
Test Objective:	Verify the phone displays the proper caller ID information.
Procedure:	Place a call to the 9133i and verify caller ID information is displayed correctly.
Expected Results:	Caller ID information should be displayed upon receiving a call.
Actual Results:	As expected.
Status:	**Pass**

Conferencing	
Test Objective:	Verify that conferences can be initiated using the Conf option within the phone itself.
Procedure:	Place a call to the first conference member then press "Conf" then dial the second member for the conference then press "Conf" once more to bridge all members.
Expected Results:	The conference should be initiated using the "Conf" button option.
Actual Results:	As expected.
Status:	**Pass**

Call History	
Test Objective:	Verify that an accurate call history is recorded and displayed from within the phone.
Procedure:	Place a few answered as well as missed calls to the phone and then press "Services" then select "Callers Lists" browse through received and missed calls, verifying they reflect the call history properly.
Expected Results:	The call history should be recorded and displayed in the "Callers Lists" menu.
Actual Results:	As expected.
Status:	**Pass**

Do Not Disturb	
Test Objective:	Verify if "Do not Disturb" mode is turned on calls to the 9133i will be sent directly to voicemail.
Procedure:	After registration, press the "Do Not Disturb" button (which must be configured in the Softkey menu in the web administration page or in the phone configuration file) and from another device place a call to the 9133i.
Expected Results:	The call placed to the 9133i will jump directly to voicemail.
Actual Results:	As expected.
Status:	**Pass**

Waiting Message Indication	
Test Objective:	Verify Asterisk phone receives WMI from Asterisk and displays this information.
Procedure:	Call the 9133i and leave a message on it's voicemail, verify that after a short while the phone recieves the WMI. *Note: a speed dial Softkey set for the voicemail extension can be configured in the Softkey menu in the web administration page.
Expected Results:	After a voicemail is placed, Asterisk will send WMI to phone, and the information will be displayed onscreen.
Actual Results:	As expected.
Status:	**Pass**

Forwarding	
Test Objective:	Verify if specified calls can be forwarded to a specified extension.
Procedure:	Press the "Options" button and select option number 8 by either pressing "8" or scrolling down and pressing "Show". Then configure a forwarding extension and set it to forward either All, Busy, NoAns, BusyNoAns, or Off. With it configured to all any call to the 9133i will instantly be forwarded to the configured extension.
Expected Results:	The calls to the 9133i should be forwarded to whatever extension is specified, using the forwarding condition as specified.
Actual Results:	As expected.
Status:	**Pass**

SIP Presence / Busy Lamp Field (BLF)	
Test Objective:	Verify if BLF softkeys are configured (and if Asterisk is correctly configured) the BLF extensions will have their status (on-hook or off-hook) displayed on-screen.
Procedure:	Configure the softkeys menu in the web administration page so that there some BLF extensions specified. (This is done in the same way that "speeddial" extensions are setup only "BLF" is specified rather than "speeddial"
Expected Results:	Small phone icons will appear next to the specified extensions. When any of these devices is busy, the phone icon will appear as if it is off-hook.
Actual Results:	As expected.
Status:	**Pass**