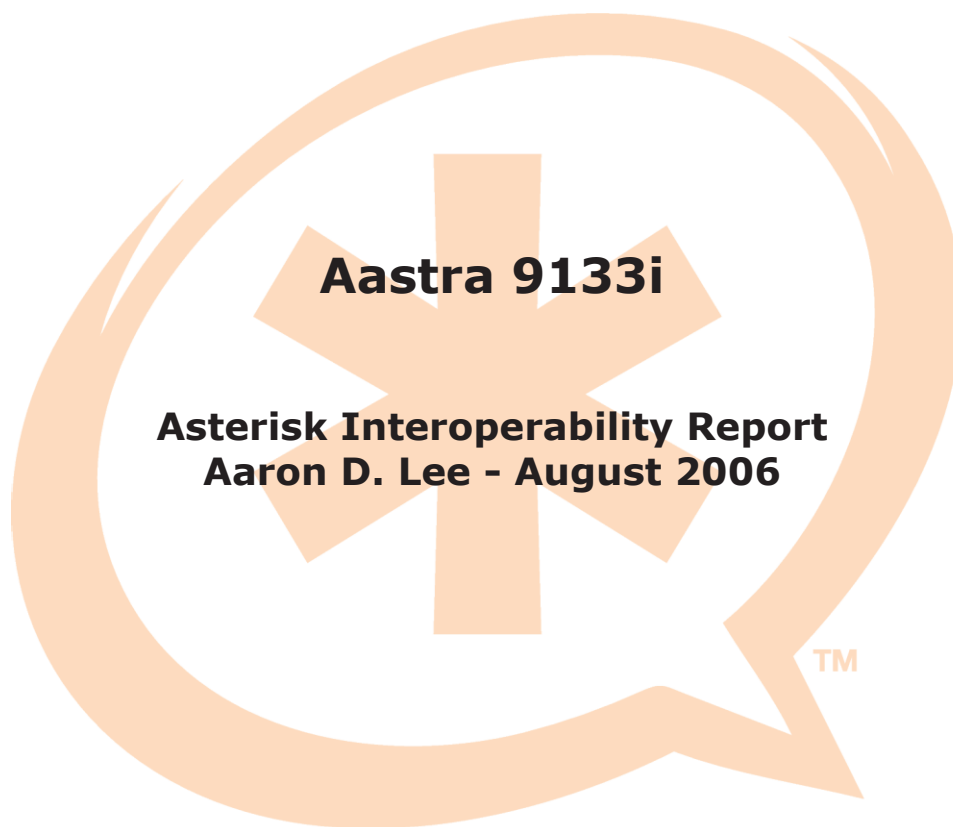


Aastra 9133i

Form: Asterisk Interoperability Report



Aastra 9133i

Asterisk Interoperability Report

Asterisk Interoperability Reports describe the certification testing performed by Digium on the specified 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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Aastra 9133i

Form: Asterisk Interoperability Report

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 versatile 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 | voicemail.conf |
|--|--|
| <pre>[aas9133i] type=friend context=sip-phones username=aas9133i secret=blah host=dynamic mailbox=9133@default defaultip=192.168.0.75 dtmfmode=rfc2833</pre> | <pre>9133 => 5555,Aastra9133i,<email></pre> |
| extensions.conf | |
| Using old=style n+101 extensions: | |
| <pre>[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 ...</pre> | |
| Using stdexten macro: | |
| <pre>[sip-phones] ... exten => 9133,1,Macro(stdexten,9133,SIP/aas9133i) ...</pre> | |
| Hints for SIP presence: | |
| <pre>[buddypress] ... exten => 9133,hint,SIP/aas9133i exten => 9133,1,Macro(line,\${aas9133i})</pre> | |

SIP Device Configuration

Configuration overview:

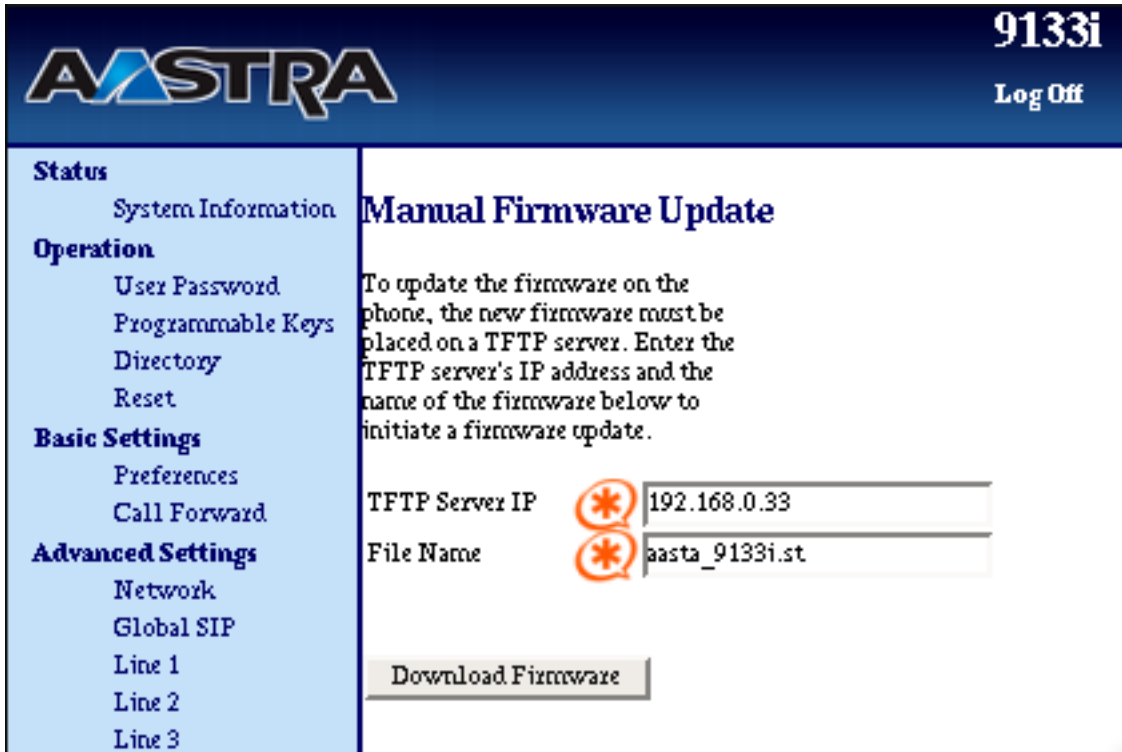
The 9133i can be configured in any of these ways:

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.

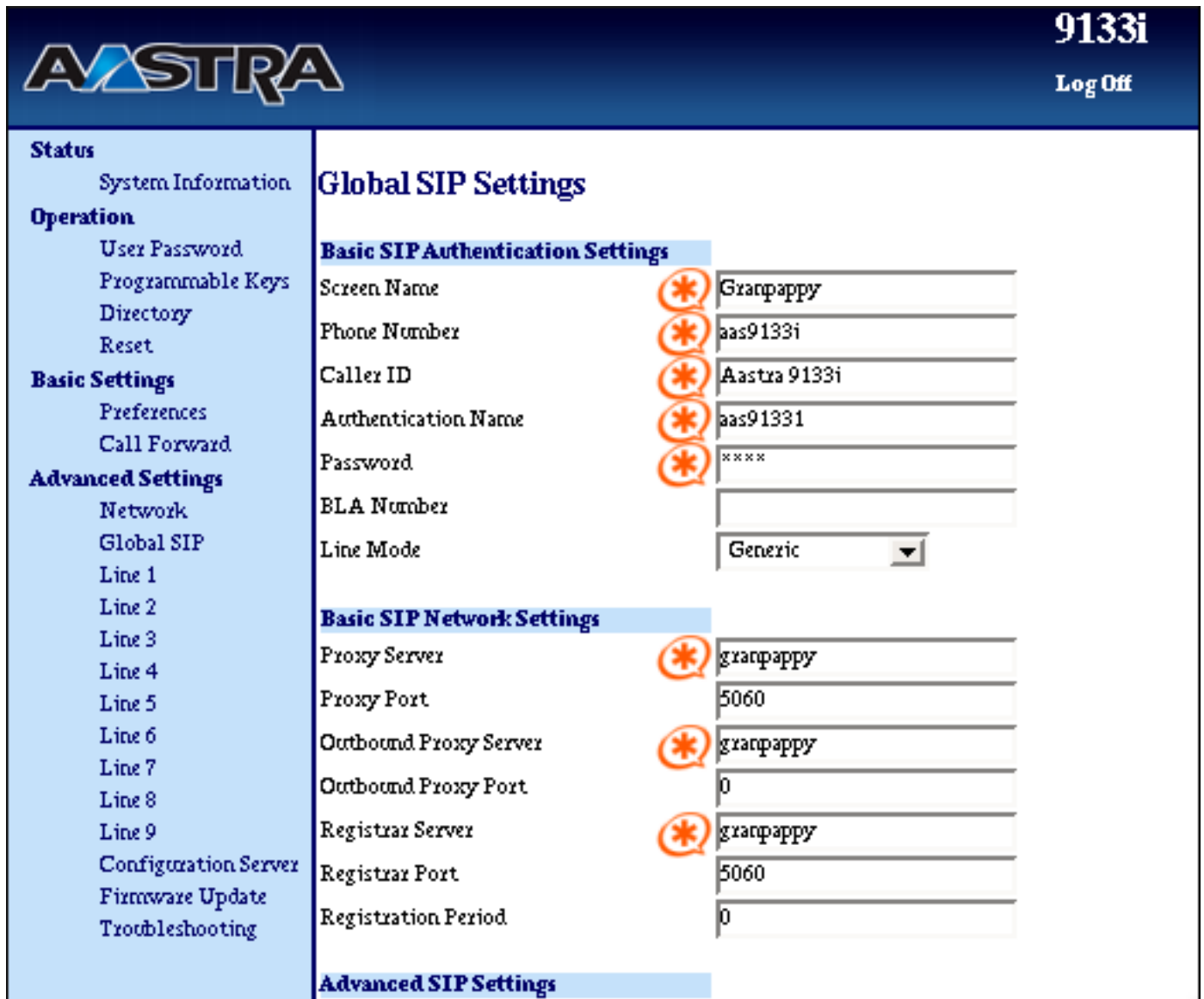
Firmware



The screenshot shows the Aastra 9133i web configuration interface. The top header includes the Aastra logo and the text "9133i" with a "Log Off" link. A left-hand navigation menu is visible, with sections for Status, Operation, Basic Settings, and Advanced Settings. The main content area is titled "Manual Firmware Update" and contains the following text: "To update the firmware on the phone, the new firmware must be placed on a TFTP server. Enter the TFTP server's IP address and the name of the firmware below to initiate a firmware update." Below this text are two input fields: "TFTP Server IP" with the value "192.168.0.33" and "File Name" with the value "aaasta_9133i.st". A "Download Firmware" button is located at the bottom of the form.

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.




The screenshot shows the Aastra 9133i web interface. The top header includes the Aastra logo on the left, the model number '9133i' on the right, and a 'Log Off' link. A left-hand navigation menu is visible, with categories like Status, Operation, Basic Settings, and Advanced Settings. The main content area is titled 'Global SIP Settings' and is divided into three sections: Basic SIP Authentication Settings, Basic SIP Network Settings, and Advanced SIP Settings. Each setting field has a red asterisk icon to its left, indicating a required field. The 'Basic SIP Authentication Settings' section includes fields for Screen Name (granpappy), Phone Number (aas9133i), Caller ID (Aastra 9133i), Authentication Name (aas91331), Password (masked with ****), BLA Number, and Line Mode (set to Generic). The 'Basic SIP Network Settings' section includes Proxy Server (granpappy), Proxy Port (5060), Outbound Proxy Server (granpappy), Outbound Proxy Port (0), Registrar Server (granpappy), Registrar Port (5060), and Registration Period (0).

| Section | Setting | Value |
|-----------------------------------|-----------------------|--------------|
| Basic SIP Authentication Settings | Screen Name | granpappy |
| | Phone Number | aas9133i |
| | Caller ID | Aastra 9133i |
| | Authentication Name | aas91331 |
| | Password | **** |
| | BLA Number | |
| | Line Mode | Generic |
| Basic SIP Network Settings | Proxy Server | granpappy |
| | Proxy Port | 5060 |
| | Outbound Proxy Server | granpappy |
| | Outbound Proxy Port | 0 |
| | Registrar Server | granpappy |
| | Registrar Port | 5060 |
| | Registration Period | 0 |
| Advanced SIP Settings | | |

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.



9133i

Log Off

| | |
|---|--|
| <p>Status</p> <ul style="list-style-type: none"> System Information <p>Operation</p> <ul style="list-style-type: none"> User Password Programmable Keys Directory Reset <p>Basic Settings</p> <ul style="list-style-type: none"> Preferences Call Forward <p>Advanced Settings</p> <ul style="list-style-type: none"> Network Global SIP Line 1 Line 2 Line 3 Line 4 Line 5 Line 6 Line 7 Line 8 Line 9 Configuration Server Firmware Update Troubleshooting | <h3 style="margin-top: 0;">Network Settings</h3> <div style="background-color: #e6f2ff; padding: 2px; margin-bottom: 5px;">Basic Network Settings</div> <p>DHCP <input checked="" type="checkbox"/> Enabled</p> <p>IP Address <input type="text" value="192.168.0.131"/></p> <p>Subnet Mask <input type="text" value="255.255.255.0"/></p> <p>Gateway <input type="text" value="192.168.0.1"/></p> <p>Primary DNS <input type="text" value="192.168.0.1"/></p> <p>Secondary DNS <input type="text" value="0.0.0.0"/></p> <div style="background-color: #e6f2ff; padding: 2px; margin-bottom: 5px;">Advanced Network Settings</div> <p>NAT IP <input type="text" value="0.0.0.0"/></p> <p>NAT Port <input type="text" value="0"/></p> <p>Nortel NAT Traversal Enabled <input type="text" value="No"/></p> <p>Nortel NAT Timer (seconds) <input type="text" value="60"/></p> <p>NTP Time Servers <input checked="" type="checkbox"/> Enabled</p> <p>Time Server 1 <input checked="" type="checkbox"/> <input type="text" value="time"/></p> <p>Time Server 2 <input checked="" type="checkbox"/> <input type="text" value="pool.ntp.org"/></p> <p>Time Server 3 <input type="text" value="0.0.0.0"/></p> <div style="background-color: #e6f2ff; padding: 2px; margin-top: 5px;">Type of Service, DSCP</div> |
|---|--|

Configuration Settings

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

9133i

[Log Off](#)

Status
System Information


Operation
User Password
Programmable Keys
Directory
Reset

Basic Settings
Preferences
Call Forward



Advanced Settings
Network
Global SIP
Line 1
Line 2
Line 3
Line 4
Line 5
Line 6
Line 7
Line 8
Line 9
Configuration Server
Firmware Update
Troubleshooting

Configuration Server Settings

Settings

| | | |
|--------------------|---|--------------|
| Download Protocol | | TFTP ▾ |
| TFTP Server |  | 192.168.0.33 |
| Alternate TFTP | | 0.0.0.0 |
| Use Alternate TFTP | <input checked="" type="checkbox"/> | Enabled |
| FTP Server | | |
| FTP User Name | | |
| FTP Password | | |
| HTTP Server | | |
| HTTP Path | | |

Auto-Resync

| | | |
|----------------|---|---|
| Mode | |  Configuration Files ▾ |
| Time (24-hour) |  | 02:00 ▾ |

XML Push Server List(Approved IP Addresses)

Line Settings

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


9133i

[Log Off](#)

Status
System Information

Operation
User Password
Programmable Keys
Directory
Reset

Basic Settings
Preferences
Call Forward

Advanced Settings
Network
Global SIP
Line 1
Line 2
Line 3
Line 4
Line 5
Line 6
Line 7
Line 8
Line 9
Configuration Server
Firmware Update
Troubleshooting

Configuration Line 1

Basic SIP Authentication Settings

| | | |
|---------------------|---|---|
| Screen Name | * | <input type="text" value="Grantpappy"/> |
| Phone Number | * | <input type="text" value="aas9133i"/> |
| Caller ID | * | <input type="text" value="9133"/> |
| Authentication Name | * | <input type="text" value="aas9133i"/> |
| Password | * | <input type="text" value="****"/> |
| BLA Number | * | <input type="text" value="9133"/> |
| Line Mode | | <input type="text" value="Generic"/> |

Basic SIP Network Settings

| | | |
|-----------------------|---|---|
| Proxy Server | * | <input type="text" value="grantpappy"/> |
| Proxy Port | | <input type="text" value="0"/> |
| Outbound Proxy Server | * | <input type="text" value="grantpappy"/> |
| Outbound Proxy Port | | <input type="text" value="0"/> |
| Registrar Server | * | <input type="text" value="grantpappy"/> |
| Registrar Port | | <input type="text" value="0"/> |
| Registration Period | | <input type="text" value="0"/> |

9

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.



The screenshot shows the Aastra 9133i web interface. The top header includes the Aastra logo and the text "9133i" and "Log Off". A left sidebar contains a navigation menu with categories: Status (System Information), Operation (User Password, Programmable Keys, Directory, Reset), Basic Settings (Preferences, Call Forward), and Advanced Settings (Network, Global SIP, Line 1-9, Configuration Server, Firmware Update, Troubleshooting). The main content area is titled "Programmable Keys Configuration" and contains a table with columns: Key, Type, Value, and Line. The table lists seven keys (1-7) with their respective types and values. Key 1 is "none", Key 2 is "do not disturb", Key 3 is "speeddial" with value "8501", Key 4 is "BLF" with value "8410", Key 5 is "BLF" with value "3600", Key 6 is "BLF" with value "4800", and Key 7 is "BLF" with value "3200". All keys are assigned to Line 1. Below the table is a field for "BLF List URI".

| Key | Type | Value | Line |
|-------------|----------------|-------|------|
| Hard Key 1: | none | | 1 |
| Hard Key 2: | do not disturb | | 1 |
| Hard Key 3: | speeddial | 8501 | 1 |
| Hard Key 4: | BLF | 8410 | 1 |
| Hard Key 5: | BLF | 3600 | 1 |
| Hard Key 6: | BLF | 4800 | 1 |
| Hard Key 7: | BLF | 3200 | 1 |

BLF List URI:

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.

| <i>Hold and Retrieve</i> | |
|---------------------------------|---|
| 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 |

| <i>Call Waiting</i> | |
|----------------------------|---|
| 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 |

| <i>Transfer and Divert</i> | |
|-----------------------------------|--|
| 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 |

| <i>Other Party Identification</i> | |
|--|--|
| 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 |

| <i>Conferencing</i> | |
|----------------------------|---|
| 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 receives 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 on-screen. |
| 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 "speedial" extensions are setup only "BLF" is specified rather than "speedial") |
| 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 |