

# **Polycom SoundPoint IP 501**

**Form:** Asterisk Interoperability Report

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A large, light orange speech bubble graphic with a thick border, containing a large orange asterisk in the center. The text is centered within the bubble.

## **Polycom SoundPoint IP 501**

**Asterisk Interoperability Report  
Aaron D. Lee - August 2006**

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## Polycom SoundPoint IP 501

### 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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# Polycom SoundPoint IP 501

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## Product Summary

<b>Make:</b>	Polycom SoundPoint IP 501
<b>Firmware:</b>	1.6.6.0036
<b>Tested With:</b>	Asterisk B.E. B.1.1

## Product Description

The Polycom SoundPoint IP501 is a SIP phone with advanced features and excellent voice quality. This IP phone features an intuitive interface, featuring dedicated hardkeys as well as context-sensitive soft buttons, these in conjunction with an easy to administer web configuration utility makes this phone ideal for businesses transitioning to an IP PBX .

## 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 / Buddy Watch (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 Polycom SoundPoint IP 501 for interoperability testing.

sip.conf	voicemail.conf
<pre>[ip501] type=friend context=sip-phones username=ip501 secret=blah host=dynamic mailbox=5001@default defaultip=192.168.0.99 dtmfmode=rfc2833</pre>	<pre>5001 =&gt; 5555,Polycom501,&lt;email&gt;</pre>
extensions.conf	
Using old=style n+101 extensions:	
<pre>[sip-phones] ... exten =&gt; 5001,1,Dial(SIP/ip501,15) exten =&gt; 5001,2,VoiceMail(u5001) exten =&gt; 5001,3,Hangup exten =&gt; 5001,102,VoiceMail(b5001) exten =&gt; 5001,103,Hangup ...</pre>	
Using stdexten macro:	
<pre>[sip-phones] ... exten =&gt; 5001,1,Macro(stdexten,501,SIP/ip501) ...</pre>	
Hints for SIP presence:	
<pre>[buddypress] ... exten =&gt; 5001,hint,SIP/ip501 exten =&gt; 5001,1,Macro(line,\${ip501})</pre>	

## SIP Device Configuration

### Three configuration overview:

- 1.Utilizing TFTP or FTP for handling phone configuration and log files.
- 2.Configuring the phone through web administration pages.
- 3.Configuring the phone's network and registration settings from within the phone's internal menu system.

### FTP Server Configuration

An FTP/TFTP server is required for upgrading the Polycom SoundPoint IP 501's firm-ware, viewing phone log files, and setting up the phone's configuration files. The phone will upload the log files to the FTP server upon boot, so it is important that write permissions have been granted.

After the FTP server is up and running plug the phone into the network and power it on, when it first boots select the setup option.

### Example for setting the phone to use the ftp server:

```
SETUP -> Enter password (default: 456) -> Server Menu

Server Menu

Server Type: FTP
Server Address: 192.168.0.134
Server User: ftp
Server Password: *****
Prov. Method: Default
```

## Phone Configuration Files

### <mac-address>-directory.xml

Rather than tediously entering each contact into the phone directory on the phone itself, the entire directory can be written in xml format in a file named according to the following format: <mac-address>-directory.xml. To enable SIP presence so that the directory displays defined contacts' status.

#### Directory with buddy watch enabled contacts:

```
<?xml version="1.0" encoding="UTF-8" standalone="yes"?>
<!-- $Revision: 1.2 $ $Date: 2004/12/21 18:28:05 $ -->
<directory>
  <item_list>
    <item>
      <ln>IP430</ln>
      <fn>Polycom</fn>
      <ct>4300</ct>
      <sd>1</sd>
      <rt>3</rt>
      <dc/>
      <ad>0</ad>
      <ar>0</ar>
      <bw>1</bw>
      <bb>0</bb>
    </item>
    <item>
      <ln>IP601</ln>
      <fn>Polycom</fn>
      <ct>6001</ct>
      <sd>2</sd>
      <rt>3</rt>
      <dc/>
      <ad>0</ad>
      <ar>0</ar>
      <bw>1</bw>
      <bb>0</bb>
    </item>
  </item_list>
</directory>
```

...

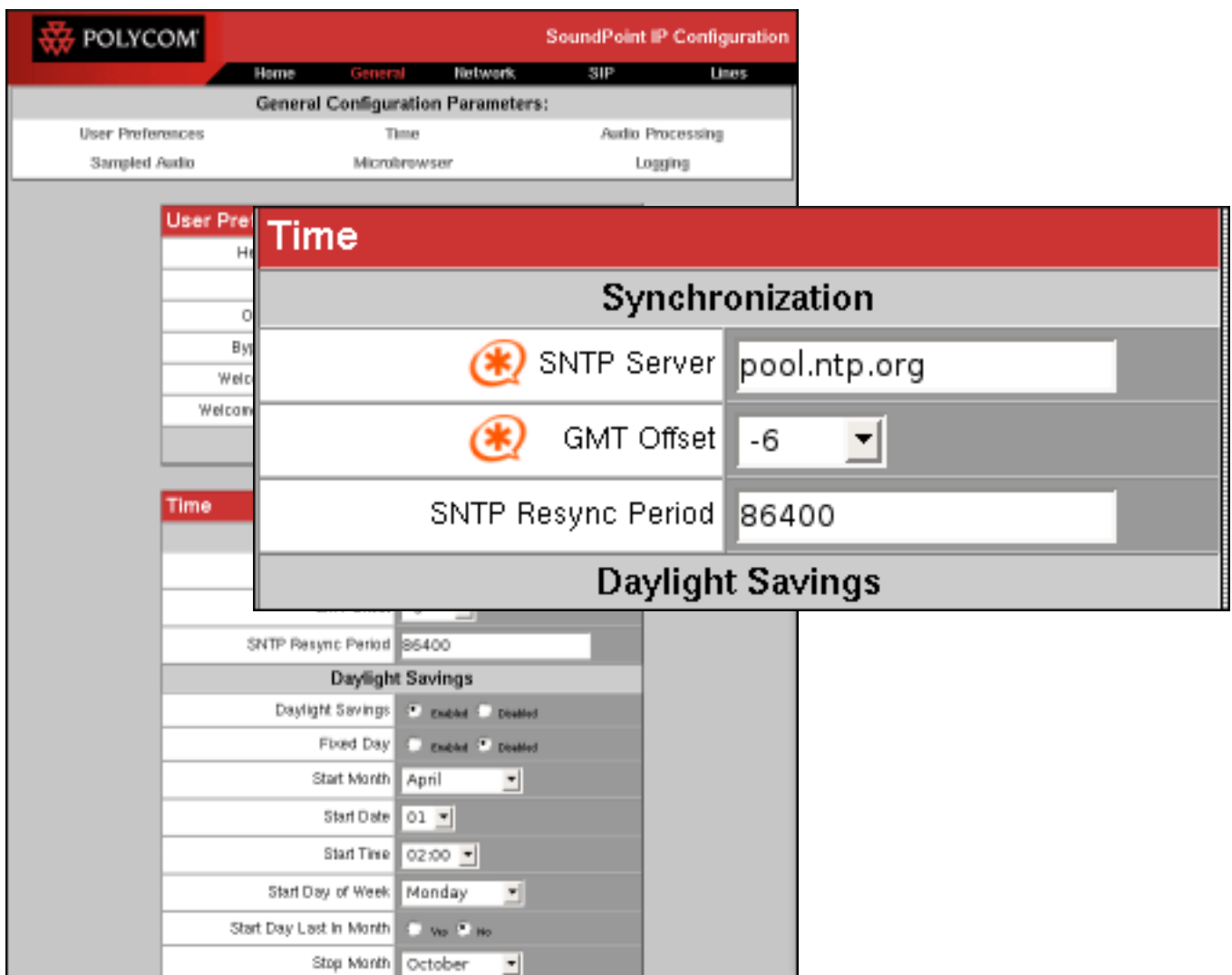
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## Web Configuration Pages

To setup the phone to register with Asterisk, the phone must be on the network, booted and ready to configure via web browser. To configure the phone open a web browser and goto the address of the phone (ie. 192.168.1.105). Then configure the following pages:

### General Settings



The screenshot displays the Polycom SoundPoint IP Configuration web interface. The top navigation bar includes 'Home', 'General', 'Network', 'SIP', and 'Lines'. The 'General' tab is active, showing 'General Configuration Parameters' with sub-sections for 'User Preferences', 'Time', and 'Audio Processing'. The 'Time' sub-section is expanded, showing 'Synchronization' and 'Daylight Savings' settings. The 'Synchronization' section includes fields for 'SNTP Server' (pool.ntp.org), 'GMT Offset' (-6), and 'SNTP Resync Period' (86400). The 'Daylight Savings' section includes fields for 'Daylight Savings' (Enabled), 'Fixed Day' (Disabled), 'Start Month' (April), 'Start Date' (01), 'Start Time' (02:00), 'Start Day of Week' (Monday), 'Start Day Last in Month' (No), and 'Stop Month' (October). Orange Asterisk logos are placed next to the 'SNTP Server' and 'GMT Offset' fields to indicate they have been modified.

Synchronization	
SNTP Server	pool.ntp.org
GMT Offset	-6
SNTP Resync Period	86400

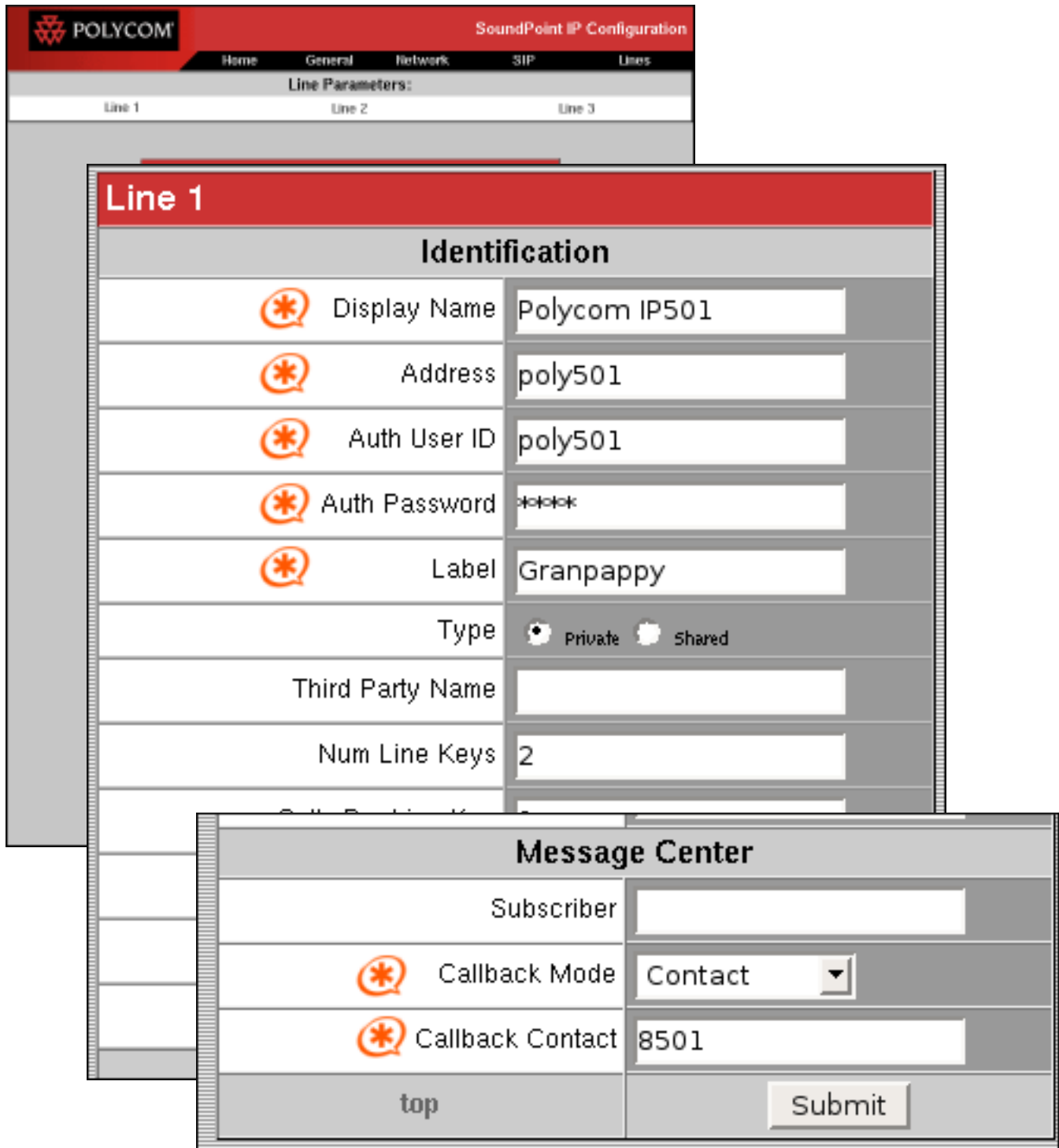
Daylight Savings	
Daylight Savings	<input checked="" type="checkbox"/> Enabled <input type="checkbox"/> Disabled
Fixed Day	<input type="checkbox"/> Enabled <input checked="" type="checkbox"/> Disabled
Start Month	April
Start Date	01
Start Time	02:00
Start Day of Week	Monday
Start Day Last in Month	<input type="checkbox"/> Yes <input checked="" type="checkbox"/> No
Stop Month	October

**\*Note:** The orange Asterisk logo denotes fields that have been modified.

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## Line Configuration



The screenshot displays the Polycom SoundPoint IP Configuration web interface. The top navigation bar includes 'Home', 'General', 'Network', 'SIP', and 'Lines'. The 'Lines' tab is active, showing 'Line Parameters' for Line 1, Line 2, and Line 3. The 'Line 1' configuration window is open, showing the following fields:

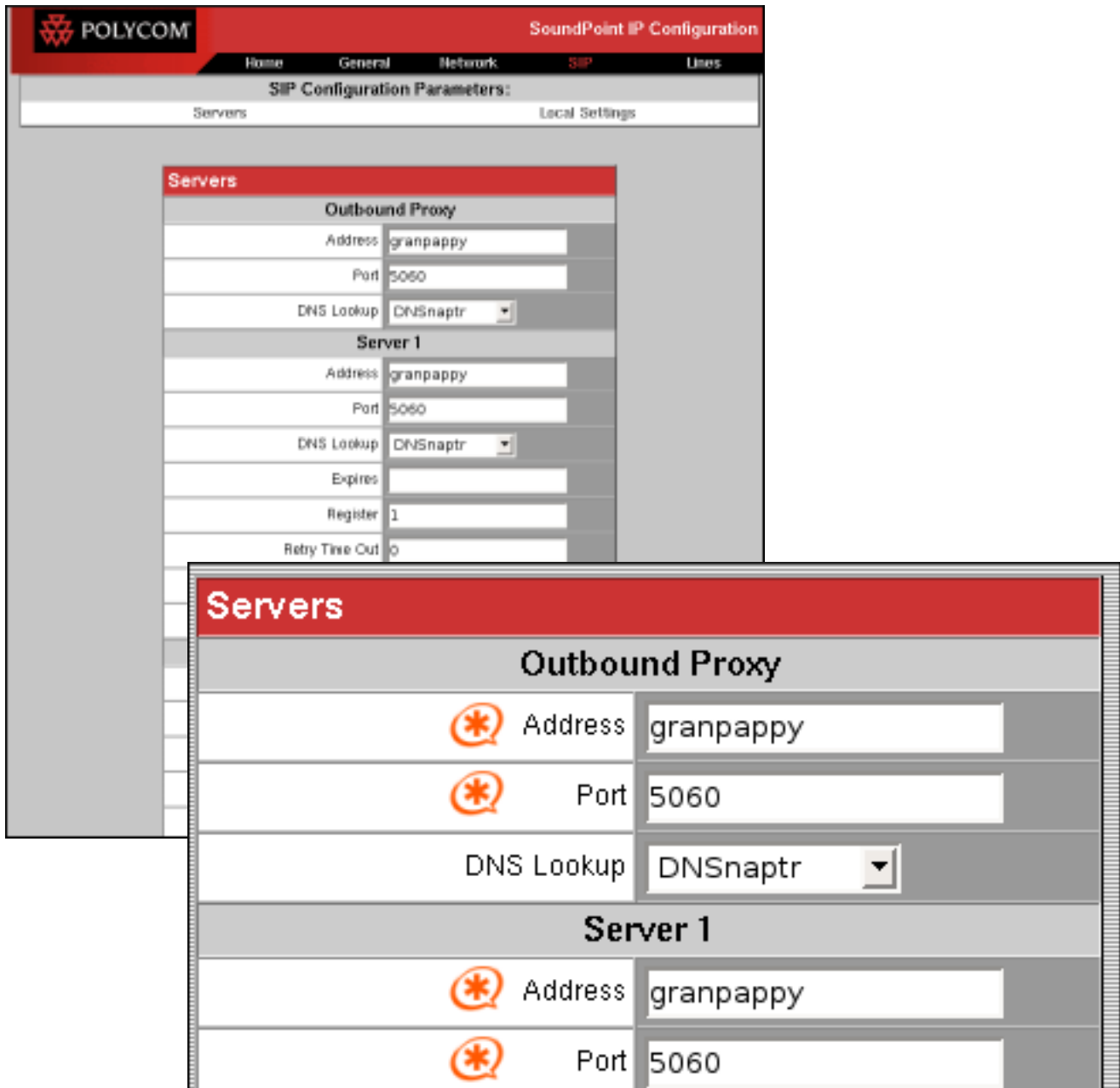
Identification	
Display Name	Polycom IP501
Address	poly501
Auth User ID	poly501
Auth Password	*****
Label	Granpappy
Type	<input checked="" type="radio"/> Private <input type="radio"/> Shared
Third Party Name	
Num Line Keys	2

Below the identification section, the 'Message Center' configuration is visible:

Subscriber	
Callback Mode	Contact
Callback Contact	8501
top	
Submit	



## SIP Configuration



The image shows a screenshot of the Polycom SoundPoint IP Configuration web interface. The page title is "SIP Configuration Parameters:" and it has tabs for "Servers" and "Local Settings". The "Servers" tab is active, showing configuration for "Outbound Proxy" and "Server 1".

**Servers**

**Outbound Proxy**

Address	granpappy
Port	5060
DNS Lookup	DNSnaptr



**Server 1**

Address	granpappy
Port	5060
DNS Lookup	DNSnaptr
Expires	
Register	1
Retry Time Out	0


The inset shows a zoomed-in view of the "Servers" section, highlighting the "Outbound Proxy" and "Server 1" configurations with Asterisk icons next to the fields.

**Servers**

**Outbound Proxy**

 Address	granpappy
 Port	5060
DNS Lookup	DNSnaptr

**Server 1**

 Address	granpappy
 Port	5060

## Test Reports

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

<b><i>Hold and Retrieve</i></b>	
<b>Test Objective:</b>	Verify that a call can be placed on hold, another call can be made, and the original call can be retrieved.
<b>Procedure:</b>	Place a call to the IP501 and place the calling party on hold. Then from the IP501 call out to another party, disconnect newest call and retrieve the call on hold.
<b>Expected Results:</b>	The call will be placed on hold and can be retrieved whenever.
<b>Actual Results:</b>	As expected.
<b>Status:</b>	<b>Pass</b>

<b><i>Call Waiting</i></b>	
<b>Test Objective:</b>	Verify that call waiting is functional, allowing a new call to be answered by placing existing conversing party on hold.
<b>Procedure:</b>	Place a call to the IP501 and answer it, with another device call the IP501. 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.
<b>Expected Results:</b>	The original caller will be on hold until new caller is disconnected or put on hold itself.
<b>Actual Results:</b>	As expected.
<b>Status:</b>	<b>Pass</b>

<b><i>Transfer and Divert</i></b>	
<b>Test Objective:</b>	Verify transferring calls works using the transfer button on the IP501.
<b>Procedure:</b>	Place a call to the IP501 during the conversation press the "Transfer" button, dial the number of the party to which you will be transferring the call, then after connection is established with said party, press "Transfer" once more to complete the transfer.
<b>Expected Results:</b>	The call will be successfully transferred via the attended transfer method.
<b>Actual Results:</b>	As expected.
<b>Status:</b>	<b>Pass</b>

<b><i>Other Party Identification</i></b>	
<b>Test Objective:</b>	Verify the phone displays the proper caller ID information.
<b>Procedure:</b>	Place a call to the IP501 and verify caller ID information is displayed correctly.
<b>Expected Results:</b>	Caller ID information should be displayed upon receiving a call.
<b>Actual Results:</b>	As expected.
<b>Status:</b>	<b>Pass</b>

<b><i>Conferencing</i></b>	
<b>Test Objective:</b>	Verify that conferences can be initiated using the Conf option within the phone itself.
<b>Procedure:</b>	Place a call to the first conference member then in the select "More" then "Confrcnc" (bottom of screen during call) then dial the second member for the conference then select "More" then "Confrcnc" once more to bridge all members.
<b>Expected Results:</b>	The conference should be initiated using the "Conference" button/menu option.
<b>Actual Results:</b>	As expected.

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

<b>Do Not Disturb</b>	
<b>Test Objective:</b>	Verify if "Do not Disturb" mode is turned on calls to the IP501 will be sent directly to voicemail.
<b>Procedure:</b>	After registration, enable "Do not Disturb" by pressing the "Do Not Disturb" button and from another device place a call to the IP501.
<b>Expected Results:</b>	The call placed to the IP501 will jump directly to voicemail.
<b>Actual Results:</b>	As expected.
<b>Status:</b>	<b>Pass</b>

<b>Waiting Message Indication</b>	
<b>Test Objective:</b>	Verify Asterisk phone receives WMI from Asterisk and displays this information.
<b>Procedure:</b>	Place a call to the IP501 and leave a voicemail (by rejecting the call or letting it timeout. Verify the letter icon is present in the top left corner of the screen (this is the notification icon for MWI. Then press the "Messages" button. Verify the phone accurately presents the number of new and old messages.
<b>Expected Results:</b>	After a voicemail is placed, Asterisk will send WMI to phone, and the information will be displayed on-screen.
<b>Actual Results:</b>	As expected.

<b>Forwarding</b>	
<b>Test Objective:</b>	Verify if specified calls can be forwarded to a specified extension.
<b>Procedure:</b>	Select the "Forward" option from the top screen and enter the extension to which the calls should be forwarded. Then place a call to the IP501 and verify it gets forwarded to the destination extension.
<b>Expected Results:</b>	The calls to the IP501 should be forwarded to whatever extension is specified.
<b>Actual Results:</b>	As expected.
<b>Status:</b>	<b>Pass</b>

<b>SIP Presence / Busy Lamp Field (BLF)</b>	
<b>Test Objective:</b>	Verify if "Buddy Watch" is enabled for contacts the phone represents buddies currently on calls with a red LED.
<b>Procedure:</b>	Make the necessary additions to extensions.conf and <mac>-directory.xml, then with buddies being displayed call from one buddy to another.
<b>Expected Results:</b>	The IP501 should show both buddies as busy in the "Contact Directory" which can be accessed by selecting "Directories" -> "Contact Directory".
<b>Actual Results:</b>	As expected.
<b>Status:</b>	<b>Pass</b>