Polycom SoundPoint IP 501

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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Product Summary

Make:	Polycom SoundPoint IP 501
Firmware:	1.6.6.0036
Tested With:	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-sesitive soft buttons, these in conjustion 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

```
[ip501]
type=friend
context=sip-phones
username=ip501
secret=blah
host=dynamic
mailbox=5001@default
defaultip=192.168.0.99
dtmfmode=rfc2833
```

voicemail.conf

```
5001 => 5555,Polycom501,<email>
```

extensions.conf

Using old=style n+101 extensions:

```
[sip-phones]
...
exten => 5001,1,Dial(SIP/ip501,15)
exten => 5001,2,VoiceMail(u5001)
exten => 5001,3,Hangup
exten => 5001,102,VoiceMail(b5001)
exten => 5001,103,Hangup
...
```

Using stdexten macro:

```
[sip-phones]
...
exten => 5001,1,Macro(stdexten,501,SIP/ip501)
...
```

Hints for SIP presence:

```
[buddypres]
...
exten => 5001,hint,SIP/ip501
exten => 5001,1,Macro(line,${ip501})
```

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 firmware, 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 settting 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.

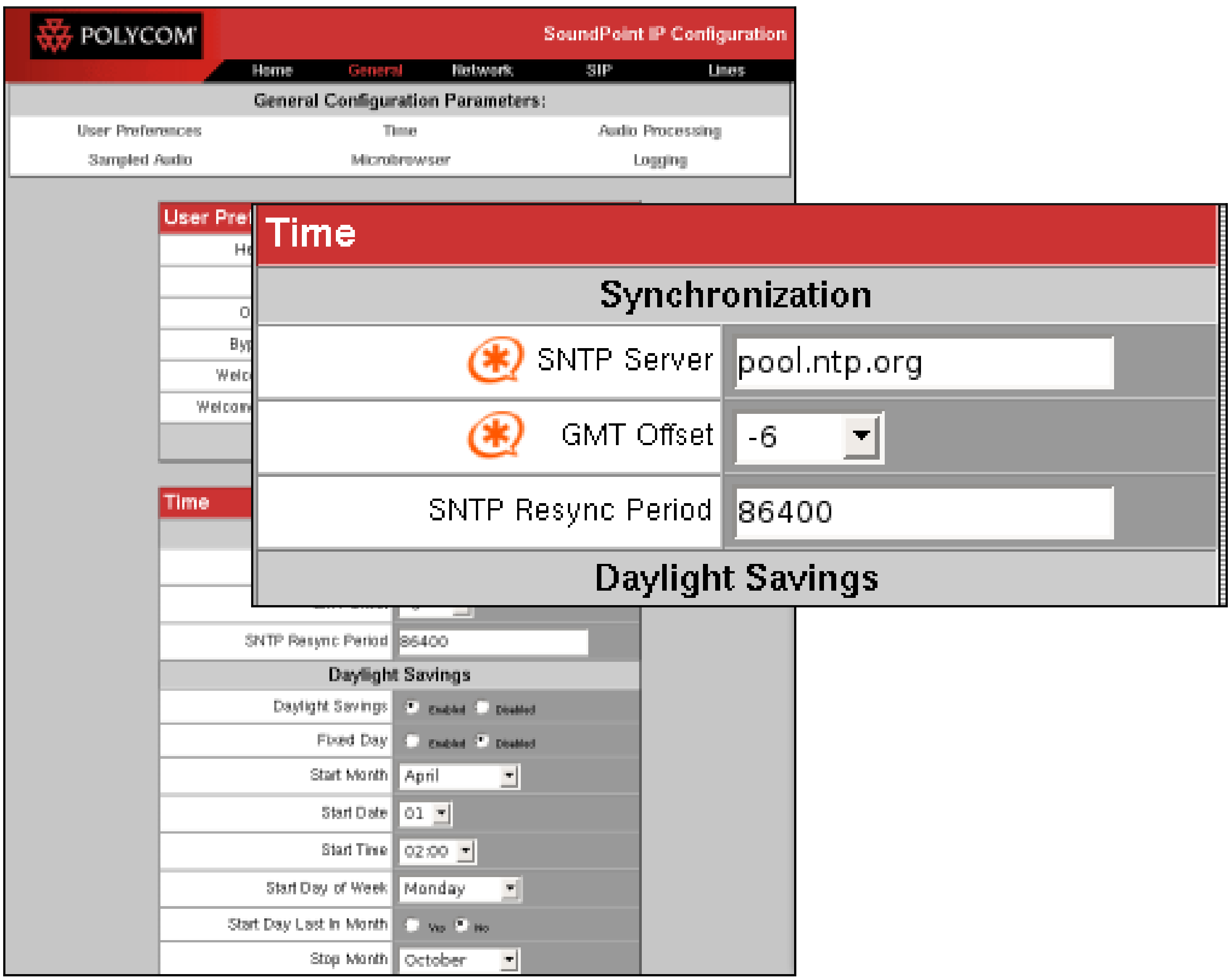
Directory with buddy watch enabled contacts:

```
<?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>
...
```

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:

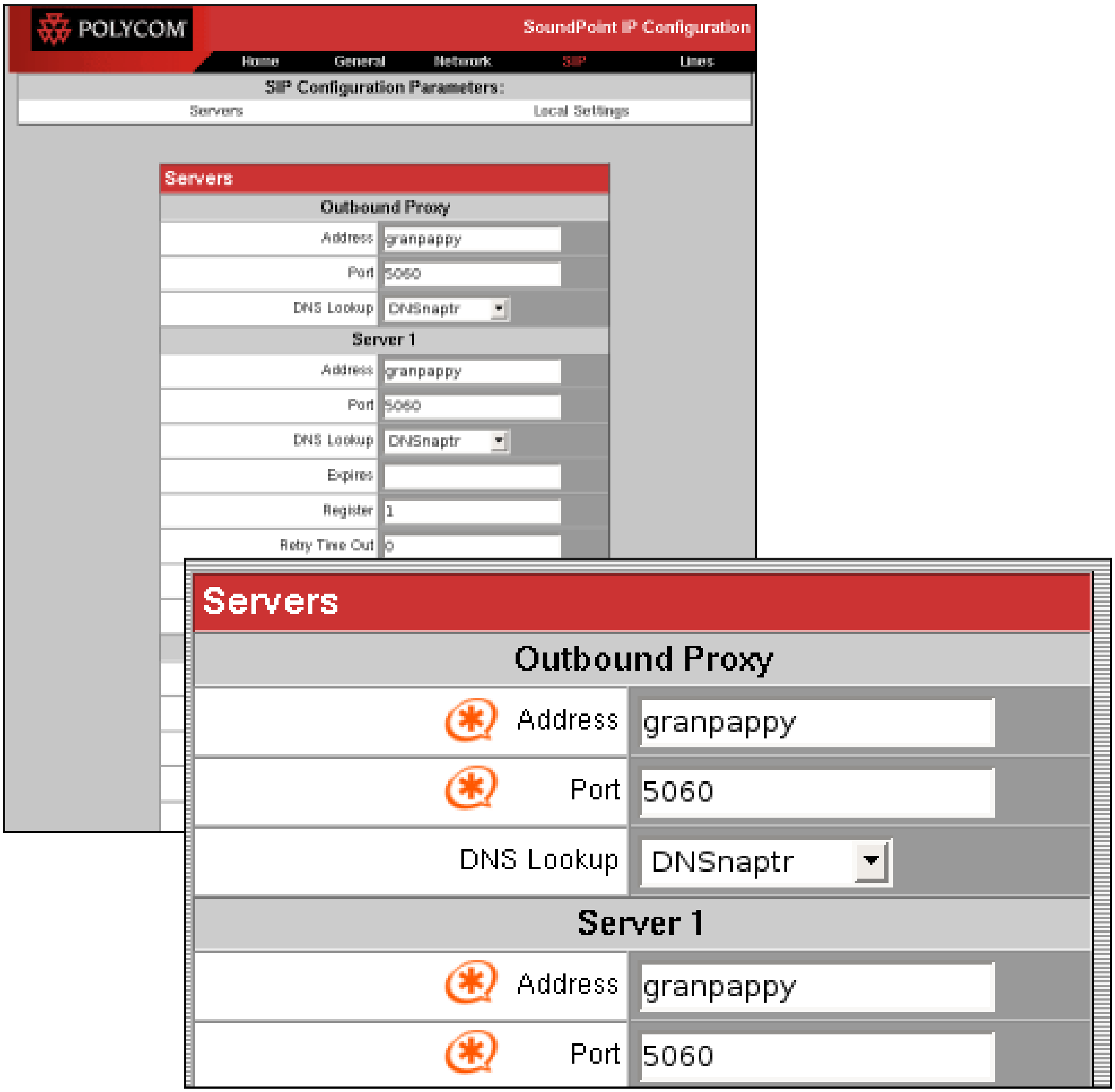
General Settings



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

Line Configuration

SIP Configuration



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, another call can be made, and the original call can be retrieved.
Procedure:	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.
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 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.
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 IP501.
Procedure:	Place a call to the IP501 during the conversation press the "Transfer" button, dial the number of the party to which you will be transfering the call, then after connection is established with said party, press "Transfer" 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 IP501 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 in the select "More" then "Confrnc" (bottom of screen during call) then dial the second member for the conference then select "More" then "Confrnc" once more to bridge all members.
Expected Results:	The conference should be initiated using the "Conference" button/menu option.
Actual Results:	As expected.

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 "Menu", then select "Features", then select "Call 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 "Call 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 IP501 will be sent directly to voicemail.
Procedure:	After registration, enable "Do not Disturb" by pressing the "Do Not Disturb" button and from another device place a call to the IP501.
Expected Results:	The call placed to the IP501 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:	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.
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.

Forwarding	
Test Objective:	Verify if specified calls can be forwarded to a specified extension.
Procedure:	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.
Expected Results:	The calls to the IP501 should be forwarded to whatever extension is specified.
Actual Results:	As expected.
Status:	**Pass**

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