Cortelco VoIP2747

Asterisk Interoperability Report

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Asterisk Interoperability Report

Asterisk Interoperability Reports are designed to overview configuration of both the Asterisk server as well as the device under test, tests preformed on the unit under test, as well as to address any interoperability issues and work-arounds or fixes for said issues.

Table of Contents

SIP Device Summary

Make:	Cortelco VoIP2747
Firmware Version:	1.5.3.019
Tested With:	Asterisk B.E. - A.1 & A.2

Features Tested and Confirmed Working

- **Call Hold and Retrieve**
- **Call Waiting**
- **Call Transfer and Divert**
- **Other Party Identification (Caller ID)**
- **Call History**
- **Configurable 'Message" Button**
- **Call Forwarding**

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 Cortelco VoIP2747 for interoperability testing.

sip.conf

```
[cortelco]
type=friend
context=default
secret=blah
host=dynamic
dtmfmode=rfc2833
username=cortelco
subscribecontext=buddypres
progressinband=no
disallow=all
allow=ulaw
mailbox=5001
```

voicemail.conf

```
5001 => 1234,Cortelco, root@localhost
```

extensions.conf

```
[sip-phones]
...
exten => 5001,1,Dial(SIP/cortelco|20)
exten => 5001,n,Voicemail(u5001)
exten => 5001,n,Hangup
exten => 5001,102,Voicemail(b5001)
exten => 5001,103,Hangup
...
[buddypres]
...
exten => 5001,hint,SIP/cortelco
exten => 5001,1,Macro(line,${corltelco})
...
```

Phone Configuration

The VoIP2747 can be easily configured via the web interface.

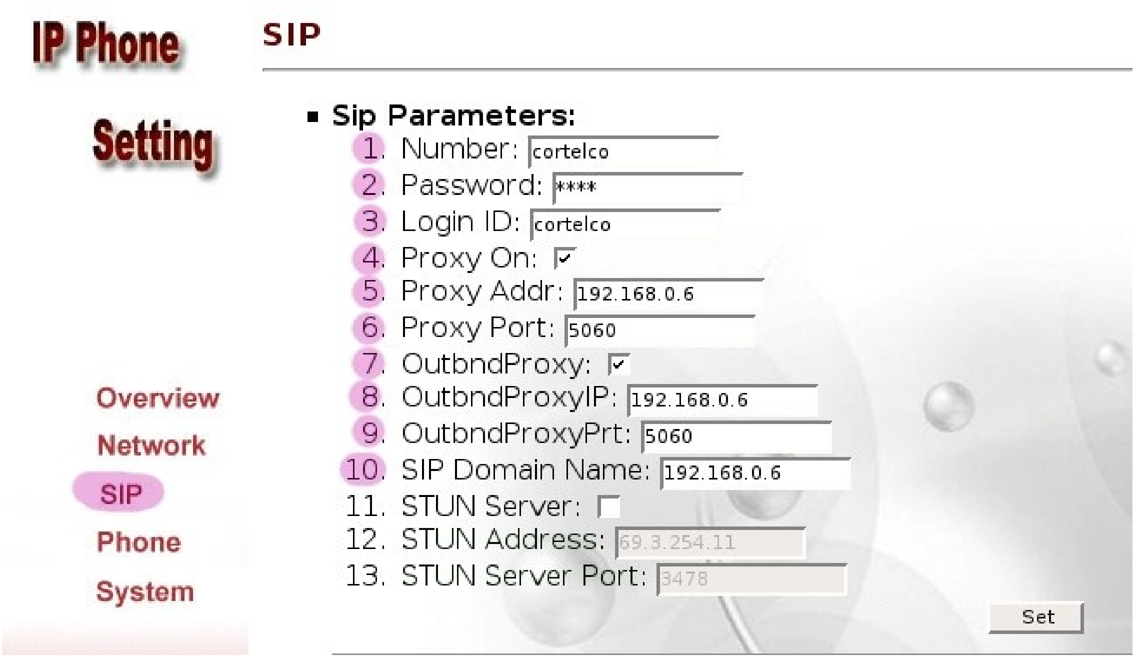
Finding the Phone's IP Address

Given that the phone was set up properly, connected to a network with a DHCP server, after the phone has booted up, press the “Menu/OK” button four times to view the phone's IP address.

Web Configuration

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:

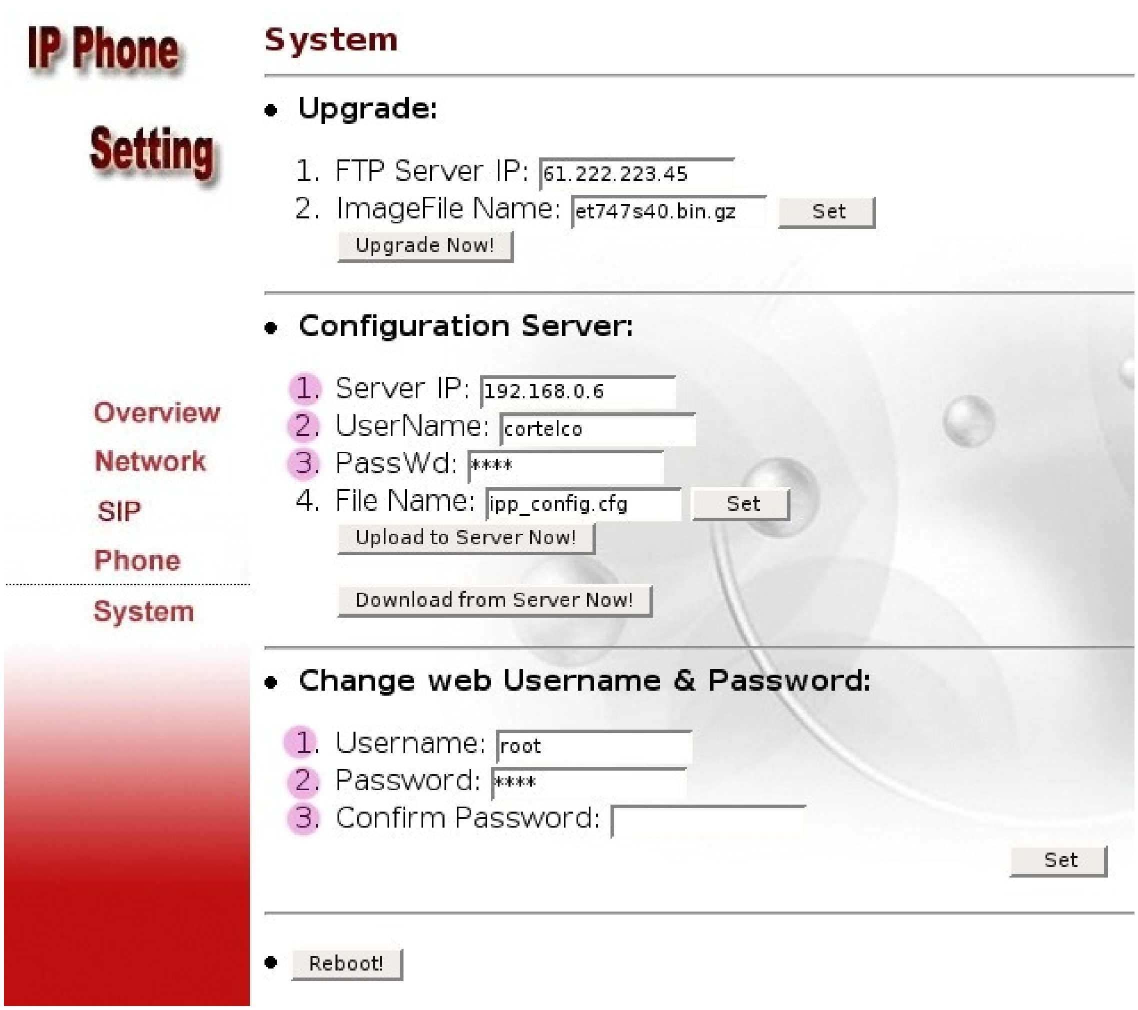
SIP Configuration



*Codec settings are found at the bottom of this page as well.

System Configuration

For the most part the settings on this page can be left alone, however if you would really like to have a copy of the phone's configuration file stored on a server and/or change the username and password used to login to the web interface, you can change the settings on this page.



Phone Menu Settings

The following settings are changed within the Cortelco VoIP2747's menu.

"Message" Button Configuration

The "Message" Button can be configured to automatically connect to Asterisk's VoiceMail system. To configure this feature, access the menu by pressing the "Menu/OK" button and navigate the menu like so (using the "UP" and "DOWN" keys to navigate):

Press "Menu/OK" -> Select "Configure" -> Enter password (Default: 135) -> Select "Message Number" -> Change to 8500 (or other VM ext.)

Forward Mode Configuration

The forward mode can be changed by navigating the menu like so:

Press "Menu/OK" -> Select "Configure" -> Enter password (Default: 135) -> Select "Forward Mode" -> Select either "Immediate", "Busy", or "No Answer" -> Select "Immed Number" and set the extension to forward to.

Only one of these modes can be enabled at any given time. After one has been selected

Memory Dial and PhoneBook

To set up the phone book and manage Memory Dial extensions press the "PhoneBook" button from there the menu allows you to add/list phonebook entries and add/change memory dial extensions.

Test Descriptions

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 VoIP2747 and place the calling party on hold. Then from the VoIP2747 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 VoIP2747 and answer it, with another device call the VoIP2747. 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 VoIP2747 during the conversation press the transfer button then dial the number to which the call will be transferred and then press the “Menu/OK” button.
Expected Results:	The call will be successfully transferred via the blind 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 VoIP2747 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**

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 the “UP” or “DOWN” button, from there view entries in “Missed Call”, “Incoming Call” and “Outgoing Call” verify they accurately reflect the previous sent/received calls.
Expected Results:	The call history should be recorded and displayed in the menus accessible through the “UP” and “DOWN” buttons.
Actual Results:	As expected.
Status:	**Pass**

Configurable "Message" Button	
Test Objective:	Verify phone can be configured to automatically connect to VoicemailMain by pressing the "Message" button.
Procedure:	Configure the "Message" button like show in the *"Message" Button Configuration* section. Then press the "Message" button and verify it dials the configured extension.
Expected Results:	The VoiceMailMain prompt will play after pressing the "Message" button.
Actual Results:	As expected.
Status:	**Pass**

Forwarding	
Test Objective:	Verify if specified calls can be forwarded to a specified extension.
Procedure:	Set the forwarding option to Busy and while the VoIP2747 is on a call, place a call to it and verify the call is forwarded to defined extension. Rpeat the test for the Immediate and No Answer modes.
Expected Results:	The calls to the VoIP2747 should be forwarded to whatever extension is specified.
Actual Results:	As expected.
Status:	**Pass**