Configuring the Atcom IP01

Rev 1, 6/3/2013

Preface

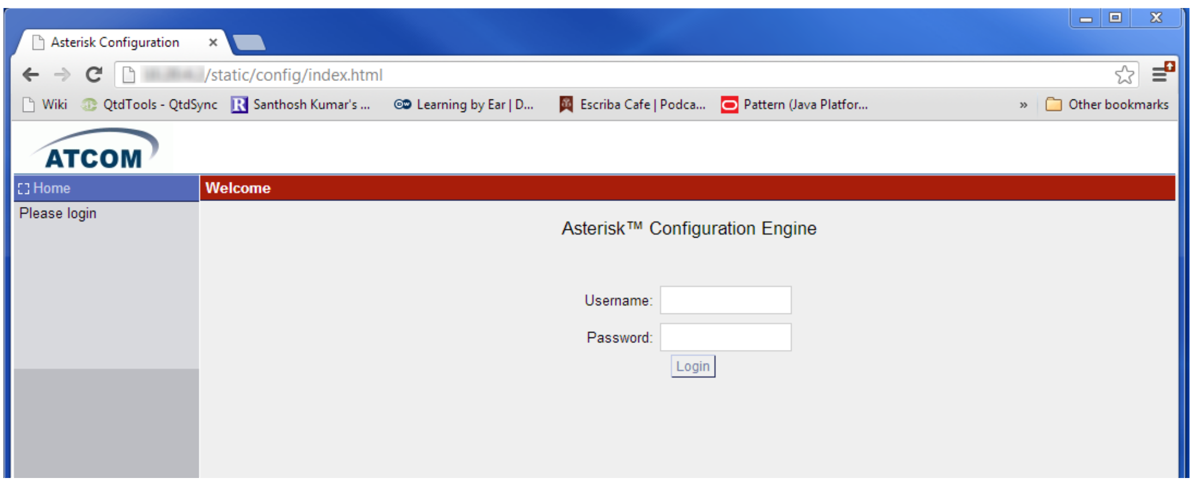
This guide provides information and hints for configuring and troubleshooting asterisk, in the context of the Atcom IP01 device. The user manual for the IP01 provides a great deal of further information, and should be consulted as well. It is available at:
http://www.atcom.cn/cn/download/pbx/ip01/ATCOM%20IP01-User%20Manual-V1.0-EN.pdf

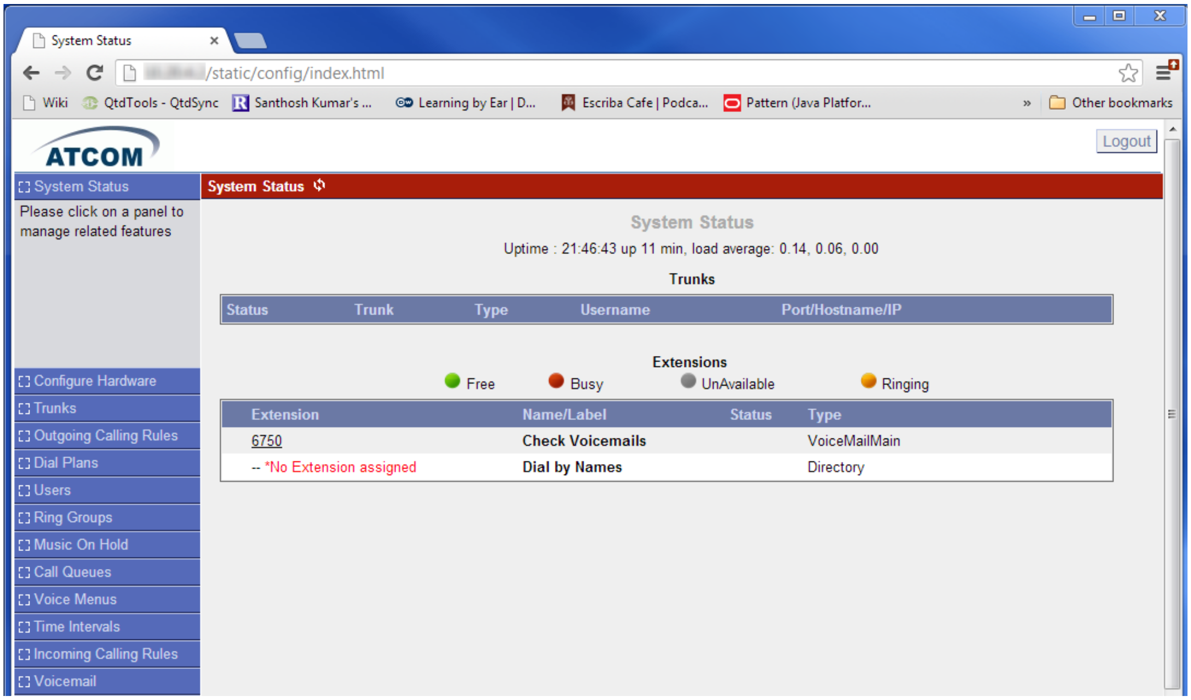
Setup

1) Login to the web admin

The IP address is 192.168.1.100 by default. It can be changed through the web interface. There is also a fallback address which cannot be change, and can be used whenever the primary ip address isn't known: 172.31.255.254 (subnet: 255.255.255.252)

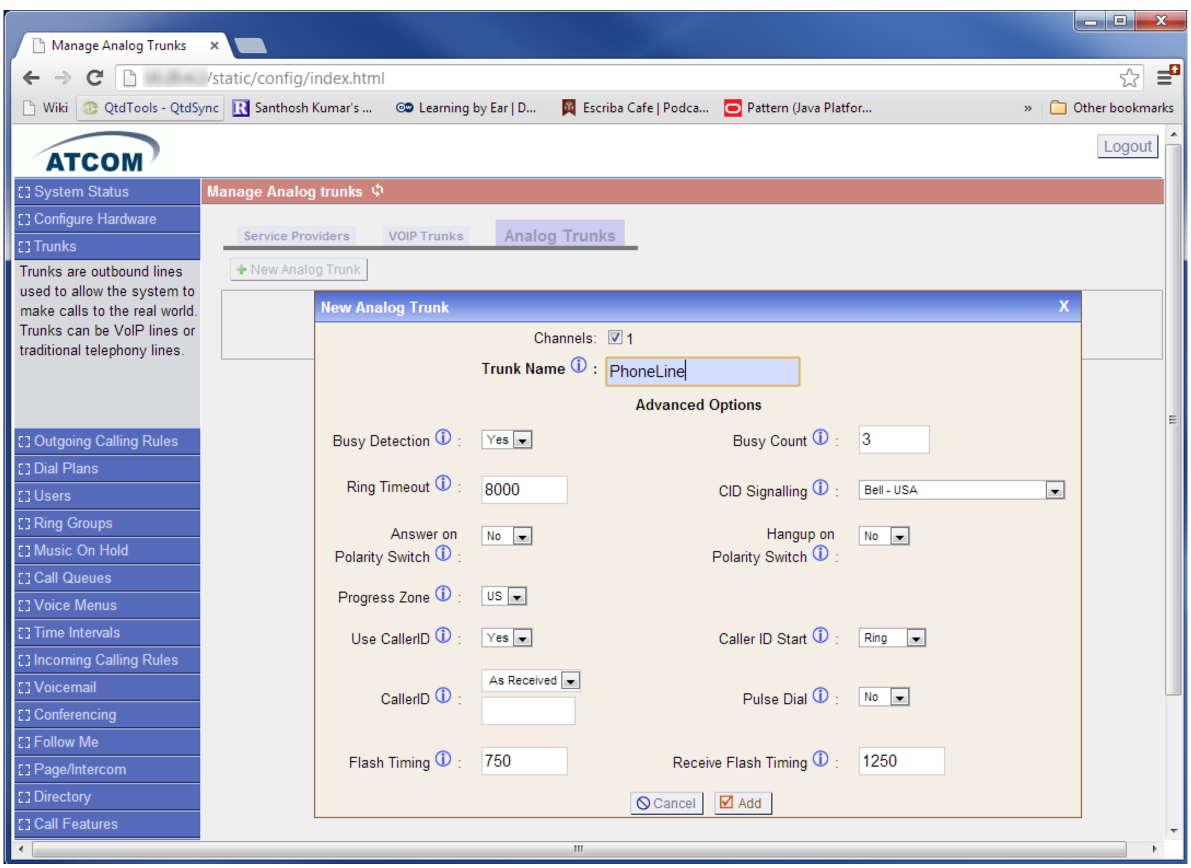
Username: admin
Password: atcom





The default screen after logging in.

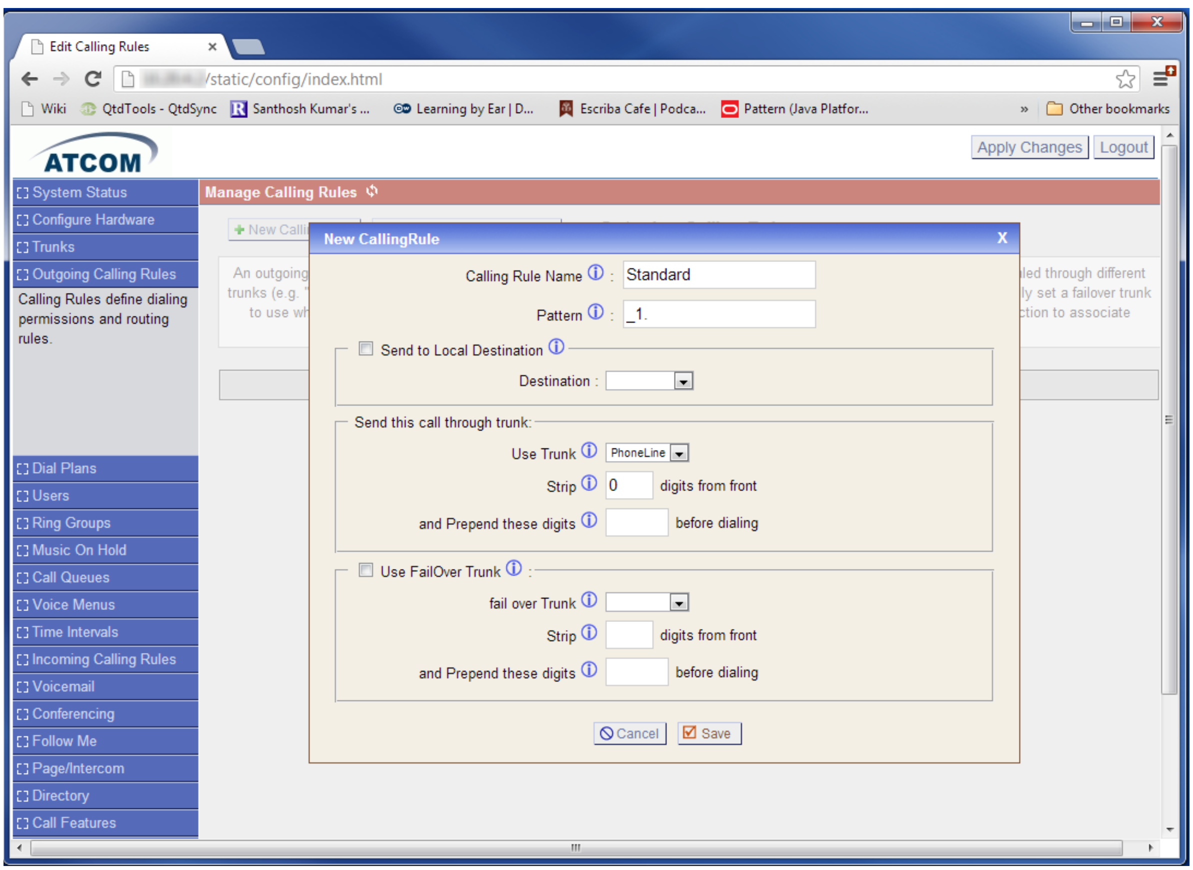
2) Create a new analog trunk by going to Trunks>Analog Trunks>New Analog Trunk Make sure to select the “channel” at the top.



3) Create a new Outgoing Call Rule
Patterns do not need to include "-", those will be ignored should the number contain them.
Example patterns:
`_1NXXNXXXXXX` : A 10 digit number starting with 1 for the country code.
`_NXXXXXX` : A local number, no area code.
`_1.` : any number starting with 1, no matter the length.

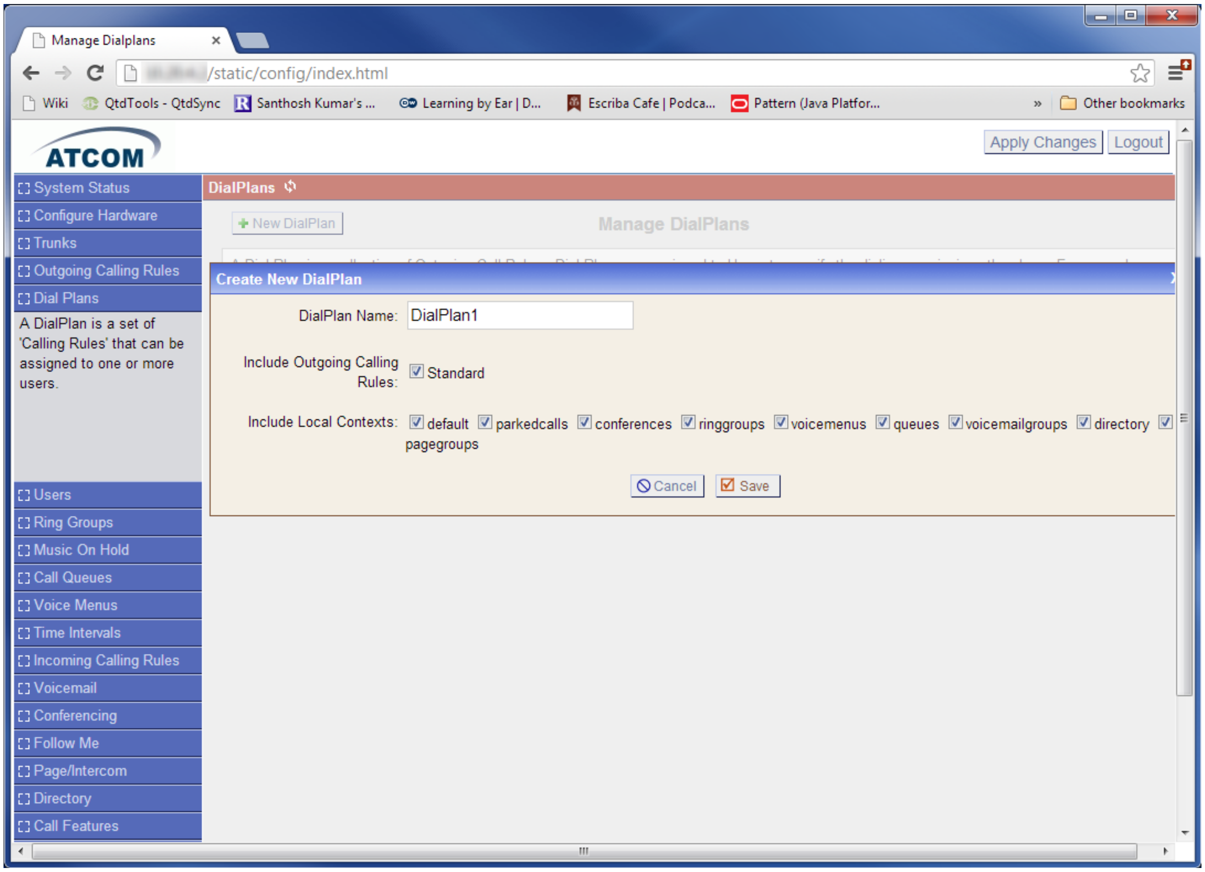
X = digit, 0-9
N = digit, 2-9

Select the recently created trunk for the "Use Trunk" setting.
Use the "strip" and "prepend" options as necessary for your phone network. For example, if 9 must be dialed to reach an outside line, but you want the phone numbers in Ignition to be standard "1-555-5555" numbers, you could use a pattern like "_1.", and pre-pend the required 9.



4) Create a new Dial plan

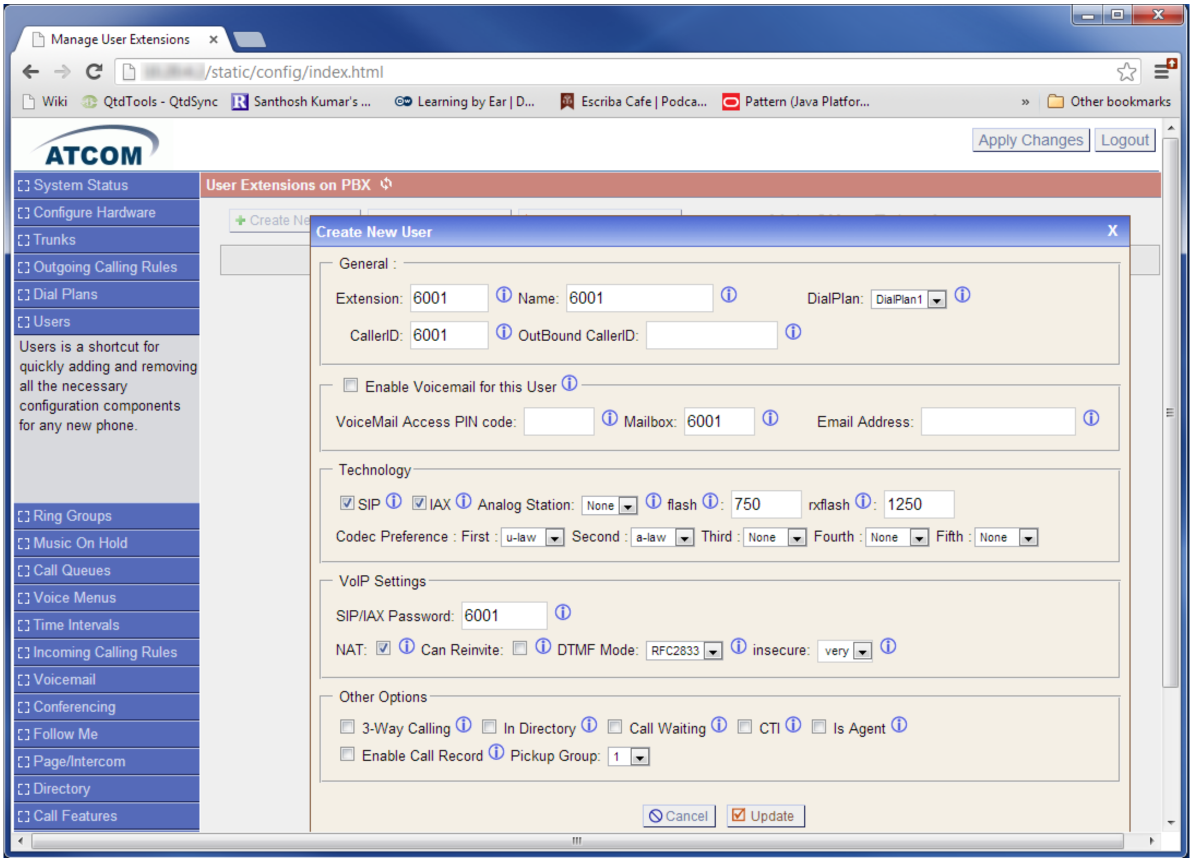
- Select the outgoing call rule created above
- Make it the default plan



5) Create a new user

- Select the dial plan created above
- Set the password to something memorable (usually just the extension number).
- Do *not* enable voicemail support.

Each Ignition system should have its own user, although redundancy may share the same user.

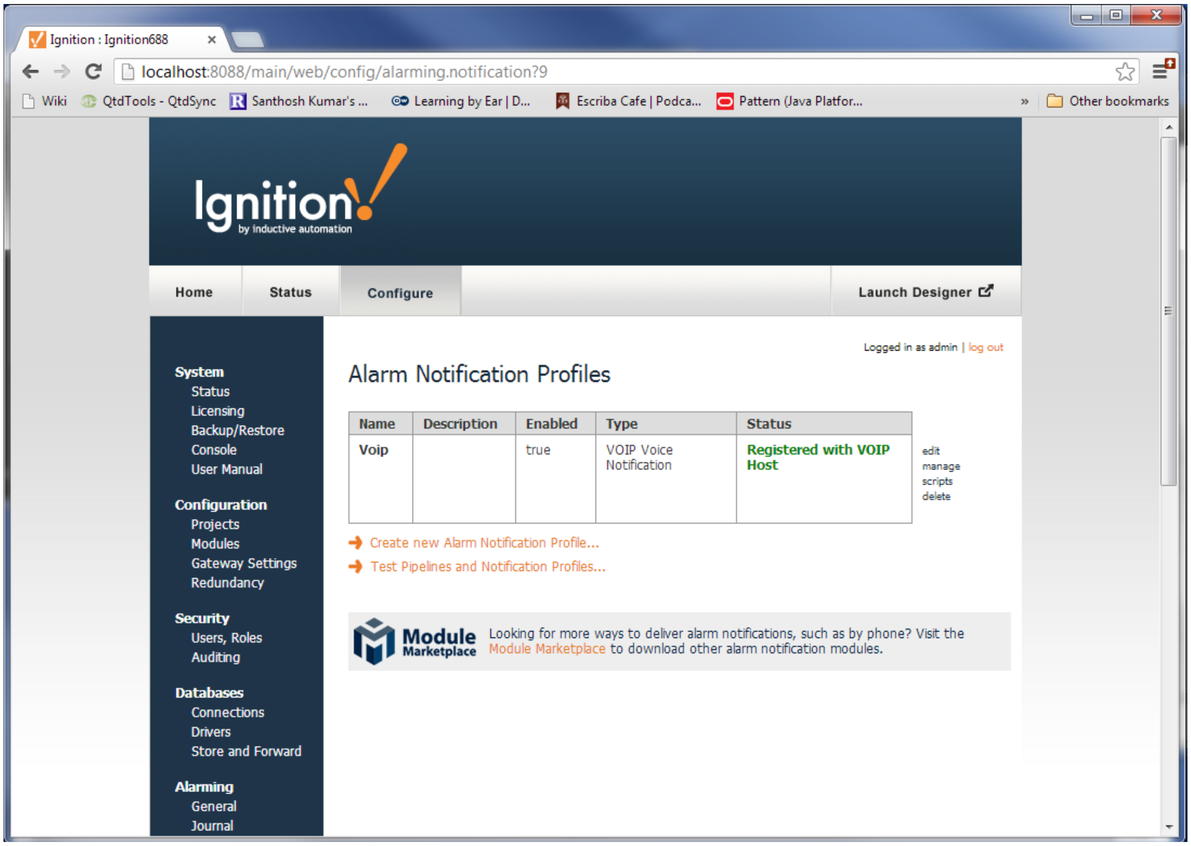


6) Apply Changes. You may need to reboot the box, which you can do by cycling power, or by going to the “Reboot” tab under the “Options” menu item.

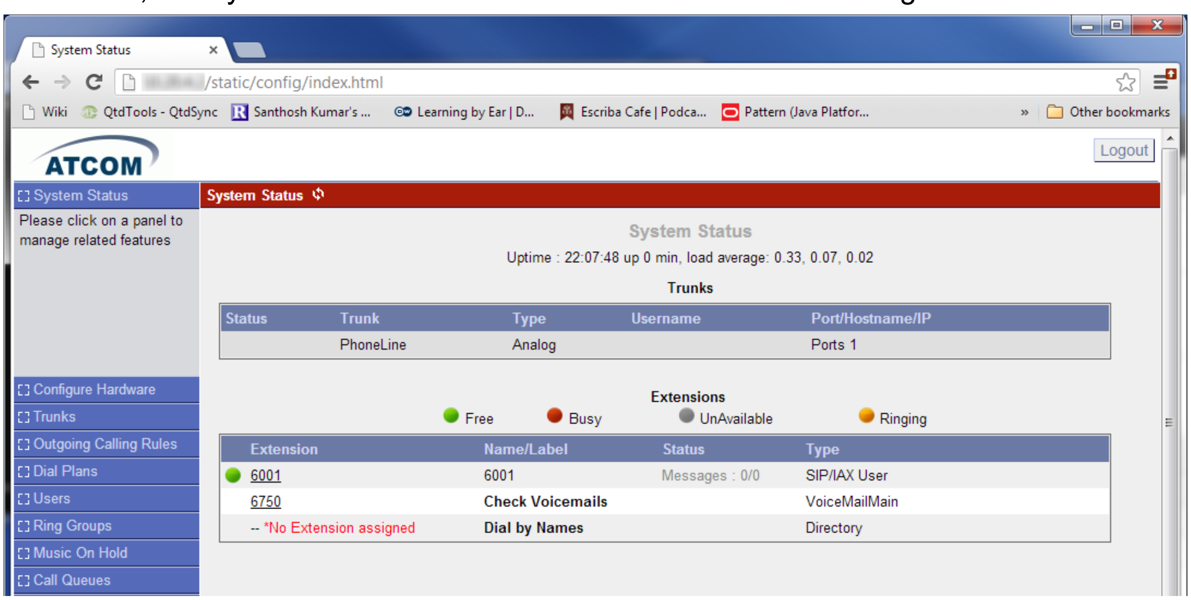
7) Configure Ignition

- Create a new Voip Notification Profile.
- Set the IP address for the “domain”. The outbound proxy is not used.
- Enter the extension as the username.
- Enter the password created for the extension.
- Save the profile

After a few seconds, you should see the profile become “Registered”:



In Asterisk, the System Status screen should show the extension in green:



Troubleshooting

Most issue arise due to issues with the call rules/dialplan, or incorrect dial patterns for the phone numbers used in Ignition. After double-checking all of the settings, the most effective way to troubleshoot is to connect directly to the box with a terminal and interactively work with the asterisk command line interface.

1) Connect to the box over SSH (“putty” is a popular client for windows).
Username: root
Password: 12xerXes16

2) Connect to the running asterisk instance, with verbose logging enabled:
> asterisk -rvvv

This level of verbosity will print out messages concerning call execution, with a fairly high level of details. For example, here is a message received when a user was not properly linked to a dialplan:

```
[Jun  3 19:59:21] NOTICE[177]: chan_sip.c:14377
handle_request_invite: Call from '6001' to extension '5551234'
rejected because extension not found.
```

Error Codes and Causes

Unfortunately, the defined SIP error codes are often used in multiple ways, and may not point directly to a root cause. However, the following descriptions are observed behaviors, and may be useful in troubleshooting.

403

During registration: The password specified for the user is not correct.

404

During registration: The specified user does not exist. Check the defined users in asterisk.
During call: A dialplan (or outbound rule) was not found for the specified number.

405

This code is SENT by Ignition when voicemail is enabled on a user, and asterisk attempts to notify Ignition of the voicemail status. Voicemail is not supported and should be turned off if possible. It may be seen at other times, but sent from Ignition, it’s harmless.

Advanced Configuration

Asterisk is a very powerful platform offering a great deal of functionality. There are many resources available to guide you for most tasks.

In regards to using the IP01 with Ignition, there are a few relatively easy things you can do to make your phone system more robust:

- Create additional extensions and use them with VOIP phone on your network, or Softphones on users' computers.
- Use a hosted phone service, such as Skype Connect, and set up a SIP trunk in asterisk. Then use that as the fall back trunk for the analog line, or create a separate call rule for long distance numbers.
- Use a SIP compatible IP speaker to create a PA system.