



User Documentation  
For version beta6

\*\*\* Work in progress \*\*\*

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Download from: [www.madikonda.com/downloads/asterisknow.pdf](http://www.madikonda.com/downloads/asterisknow.pdf)

## **Table of Contents:**

1.	About this document:.....	4
2.	Re-distribution:.....	4
3.	Copyright:.....	4
4.	Introduction: .....	5
5.	Pre-requisites:.....	5
6.	AsteriskNOW Installation:.....	6
6.1.	Partitioning: .....	10
6.2.	Network Configuration: .....	10
6.3.	Time Zone Selection: .....	12
6.4.	Administrator Password: .....	13
6.5.	About to Install: .....	14
6.6.	Formatting the System: .....	15
6.7.	Installing the Packages:.....	16
6.8.	Running post Installation scripts:.....	17
6.9.	Finishing installation:.....	18
7.	Starting AsteriskNOW:.....	19
7.1.	The AsteriskNOW console Menu:.....	19
7.1.1.	Update menu: .....	20
7.1.2.	Console Menu: .....	20
7.1.3.	Restart Menu:.....	20
7.1.4.	Shutdown Menu: .....	21
7.1.5.	Reboot Menu:.....	21
7.1.6.	Quit Menu: .....	21
8.	Configuring AsteriskNOW:.....	22
8.1.	Setup Wizard: .....	24
8.1.1.	Analog Ports:.....	24
8.1.2.	Local Extension Settings: .....	25
8.1.3.	Service Providers: .....	26
8.1.4.	Calling Rules:.....	29
8.1.5.	Voicemail:.....	34
8.1.6.	Extensions:.....	35
8.1.7.	Incoming Calls: .....	37
8.1.8.	Register your copy of AsteriskNOW: .....	39
8.2.	Asterisk Configuration Panel:.....	42
8.3.	User & Phone Configuration: .....	43
	Extension Options:.....	44
8.4.	Conferencing: .....	46
	Conference Room Options:.....	46
8.5.	Voicemail Configuration:.....	47
8.6.	Call Queues:.....	48
8.7.	Service Providers: .....	50
8.8.	Calling Rules: .....	51
8.9.	Incoming Calling Rules:.....	52
8.10.	Voice Menu Configuration: .....	53
8.11.	Time Based Rules: .....	54
8.12.	Call Parking: .....	54
8.13.	Ring Groups:.....	55
8.14.	Record a Menu: .....	55

8.15.	Active Channels: .....	58
8.16.	Graphs: .....	58
8.17.	System Information:.....	59
8.18.	Asterisk Logs: .....	60
8.19.	File Editor:.....	61
8.20.	Asterisk CLI:.....	61
9.	System Setup & Administration:.....	74
9.1.	Change Password:.....	76
9.2.	Email Configuration:.....	77
9.3.	Configure Networking: .....	78
9.4.	System Information:.....	81
9.5.	System Updates:.....	82
	Schedule your updates: .....	83
9.6.	Conary Configuration:.....	84
9.7.	Time Zone Configuration:.....	84
9.8.	Upload SSL Certificate: .....	85
9.9.	Services: .....	86
9.10.	Scheduled Reboot: .....	87
9.11.	Conary Log: .....	87
9.12.	View Log: .....	88
10.	Installing Other Programs: .....	89
10.1.	Installing Mysql: .....	89
10.2.	Install samba using conary: .....	89
11.	System Commands: .....	90
12.	To Get Root Access on Console:.....	90
13.	Advanced Configuration & User Tips: .....	91
	To Install Asterisk + Gui on a fresh Operating System:.....	91
	To allow for root login via ssh: .....	91
	Asterisk addons: .....	91
	Updating Providers.conf: .....	93
	Re-generate the GUI Certificate:.....	94
14.	Client Connections:.....	95
	Vi Commands:.....	97

## **1. About this document:**

This document is currently a work in progress project. While due care is taken in preparing this document. It is expected to contain errors or omissions.

Most of the document material comes from the AsteriskNOW software itself. So as the software develops, so does this document to keep in line with the released versions of the software.

This document also contains material from the asterisk forums.

## **2. Re-distribution:**

You may re-distribute this document in its original format without any alterations. You can always find the latest version of this document at <http://www.madikonda.com/downloads/asterisknow.pdf>

## **3. Copyright:**

All names and products mentioned in this document are for illustration and documentation purposes only.

Asterisk and AsteriskNOW are the trademarks of Digium Inc.

## **4. Introduction:**

### **What is AsteriskNOW**

AsteriskNOW™ is a Software Appliance; a customized Linux distribution that includes Asterisk®, the Asterisk GUI, and all other software needed for an Asterisk® system. The most popular open source IP PBX software, Asterisk®, can now be easily configured with a graphical interface. AsteriskNOW™ includes all the Linux components necessary to run, debug and build Asterisk®, and only those components, so installation is easy. You no longer have to worry about kernel versions and package dependencies. Unlike other Linux distributions used to deploy Asterisk, no unnecessary components that might compromise security or performance are included.

### **Asterisk®**

[Asterisk®](#) is a complete IP PBX in software. It runs on a wide variety of operating systems including Linux, Mac OS X, OpenBSD, FreeBSD and Sun Solaris and provides all of the features you would expect from a PBX including many advanced features that are often associated with high end (and high cost) proprietary PBXs. Asterisk® supports Voice over IP in many protocols, and can interoperate with almost all standards-based telephony equipment using relatively inexpensive hardware.

Asterisk® and AsteriskNOW™ are released as open source under the [GNU General Public License \(GPL\)](#), meaning that they are available for download free of charge.

Asterisk® was created by Mark Spencer of [Digium, Inc](#) in 1999. Code has been contributed from open source coders around the world, and testing and bug-patches from the community have provided invaluable aid to the development of this software.

## **5. Pre-requisites:**

- 1) A Cdrom writer to make a cdrom.
- 2) In order to successfully install and configure AsteriskNOW you need to use the Fire fox browser. Internet explorer is not supported as of Beta2 version. Download firefox from  
<http://www.mozilla.org/>

## 6. AsteriskNOW Installation:

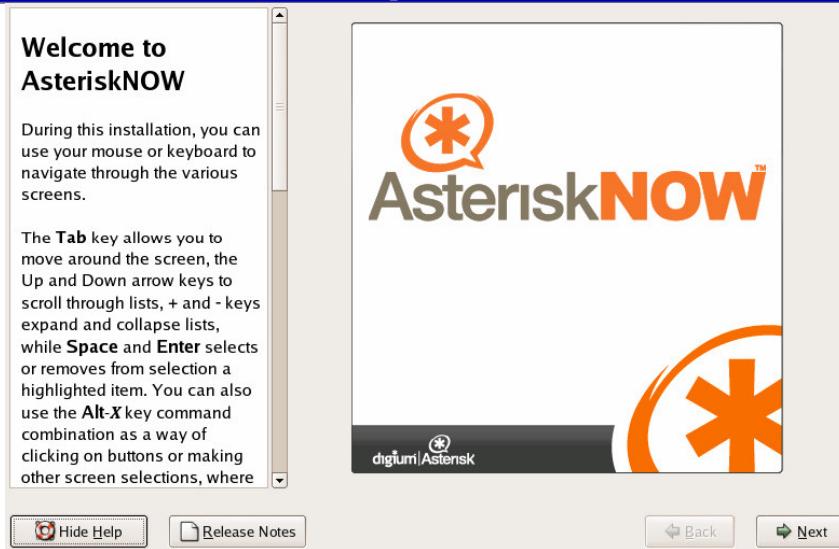
- To install AsteriskNOW on a dedicated machine, you need the iso cdrom image file from AsteriskNOW.
- Download the iso file from <http://www.asterisknow.org/downloads> and then create a cdrom from the file.
- Next insert the installation disk into the cdrom drive of your machine and then re-start the computer.
- After re-start, you will be prompted to install. Just follow the prompts and the screen shots as given below.



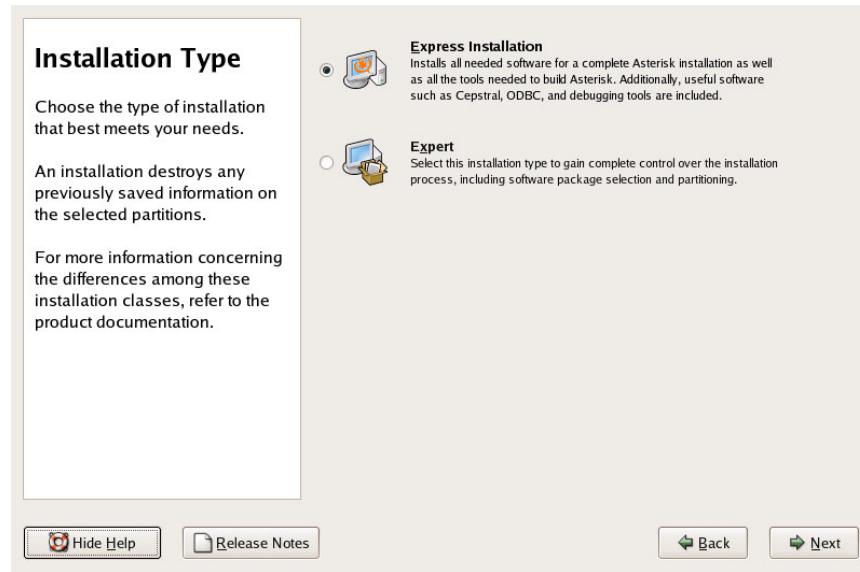
- Press the **enter** key to start the installation process using the graphical installation mode.

Welcome to rPath Linux

<Tab>/<Alt-Tab> between elements | <Space> selects | <F12> next screen

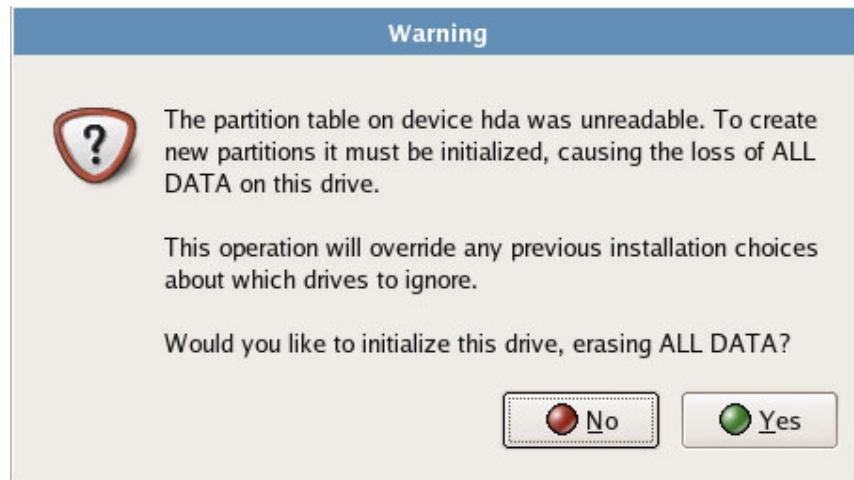


- Click **next** to continue



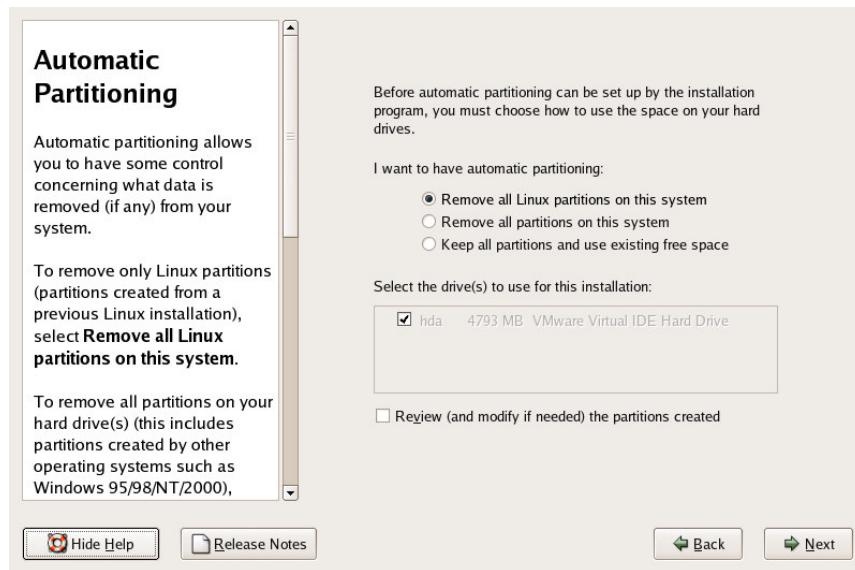
- Choose **Express installation** and click **next**.

- If you are installing on a new hard drive or over a used hard drive with windows partitions, then you will get this warning message.
- This is your final warning. You will lose all of your data you have on this hard drive if you proceed further by pressing the yes button.

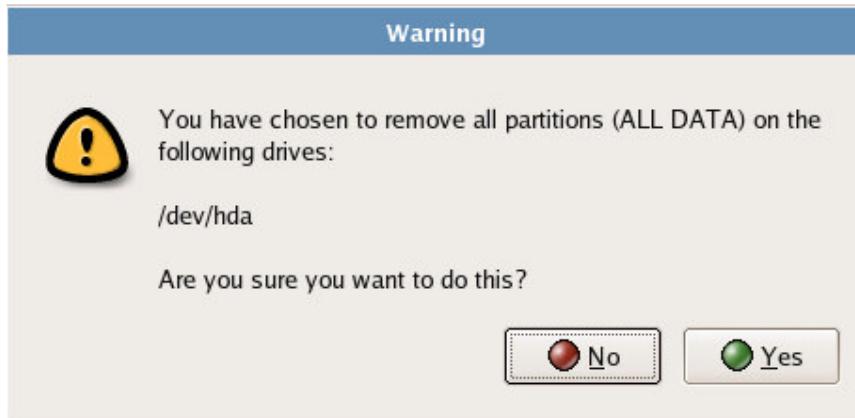


- Click on **Yes** to continue the setup.

## **6.1. Partitioning:**

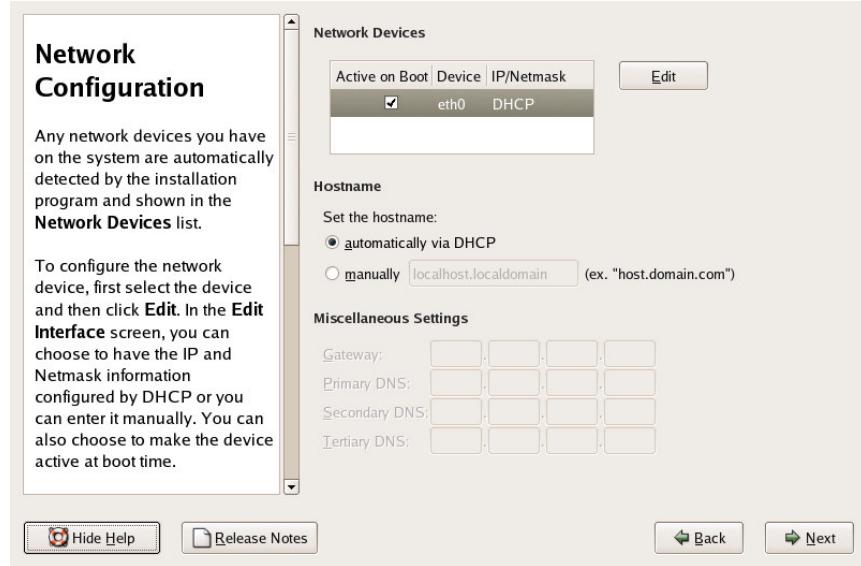


- Choose **remove all partitions on this system**
- Click **Next** to continue setup



- This is your final warning. You will lose all of your data you have on the partitions. Click **No** if you want to cancel the setup.
- Click **Yes** to continue setup and remove all partitions on your chosen hard drive.

## **6.2. Network Configuration:**



- Choose **automatically via DHCP** if you are a typical home user.
- If a static IP address is needed for the server then click on the manually option button and then fill in the rest of the boxes.
- Click **Next** to continue setup

### 6.3. Time Zone Selection:



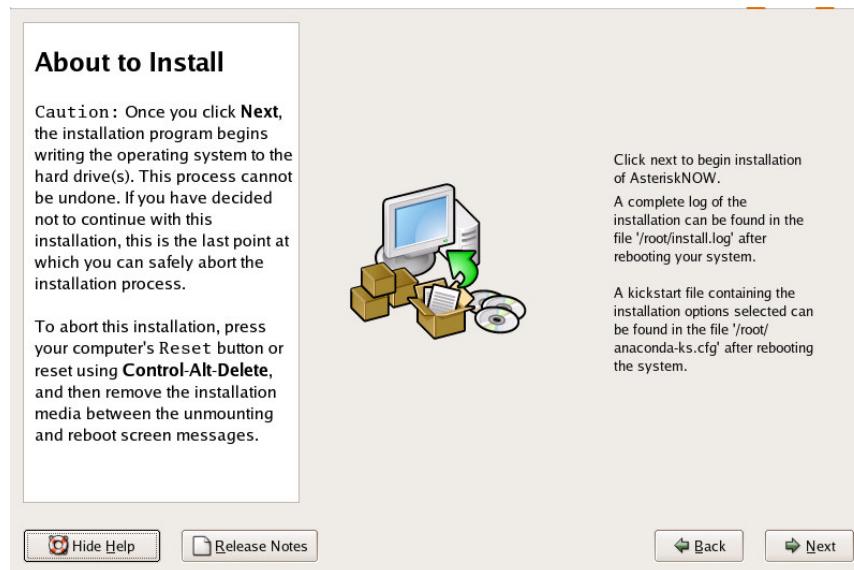
- Select your time zone.
- Click **Next** to continue setup

## 6.4. Administrator Password:



- You need to enter a password for the administrator account. This user account and the password will be used to administer the system.
- Do not leave it blank.
- Click **Next** to continue setup

## 6.5. About to Install:



- Click **Next** to continue setup

## **6.6. Formatting the System:**



- Click **Next** to continue setup

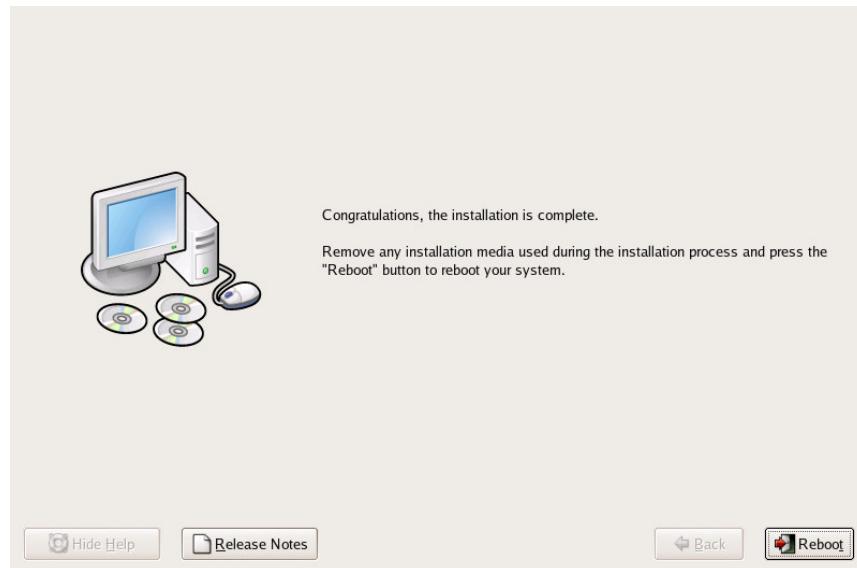
## 6.7. Installing the Packages:



## **6.8. Running post Installation scripts:**



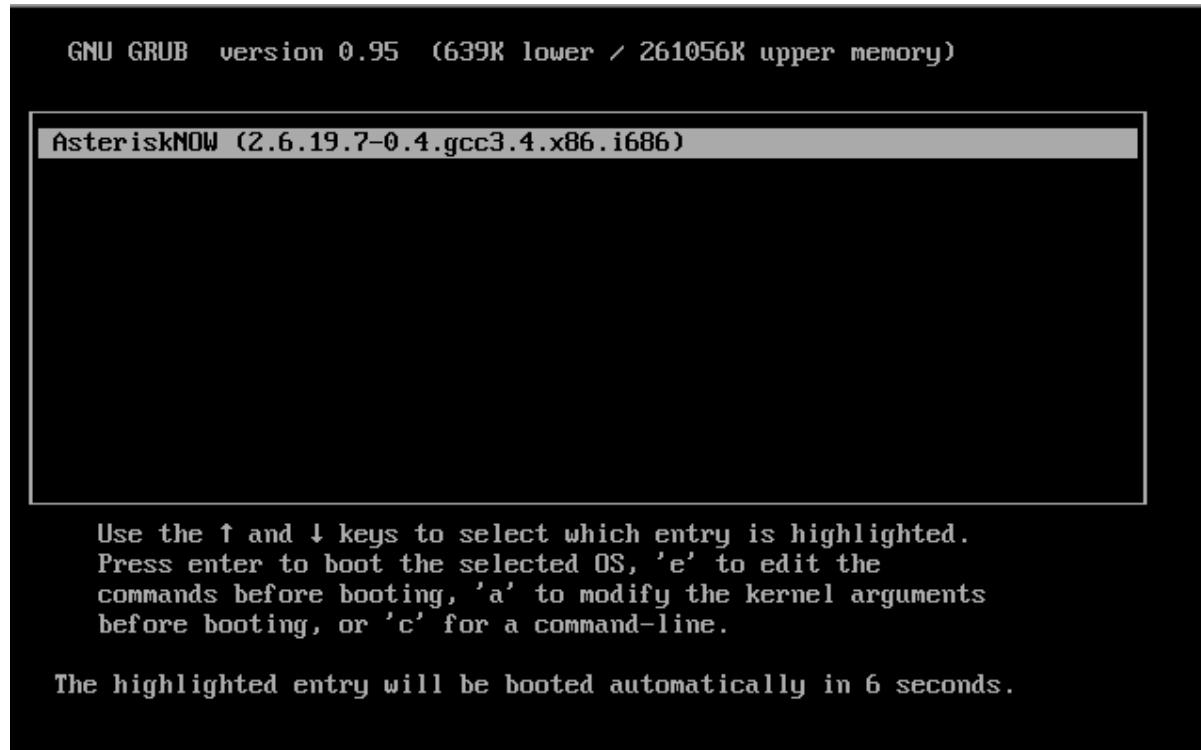
## **6.9. Finishing installation:**



- The installation now finally finished.
- Remove the Cdrom from the drive.
- Click on the **Reboot** to finish the installation process.

## 7. Starting AsteriskNOW:

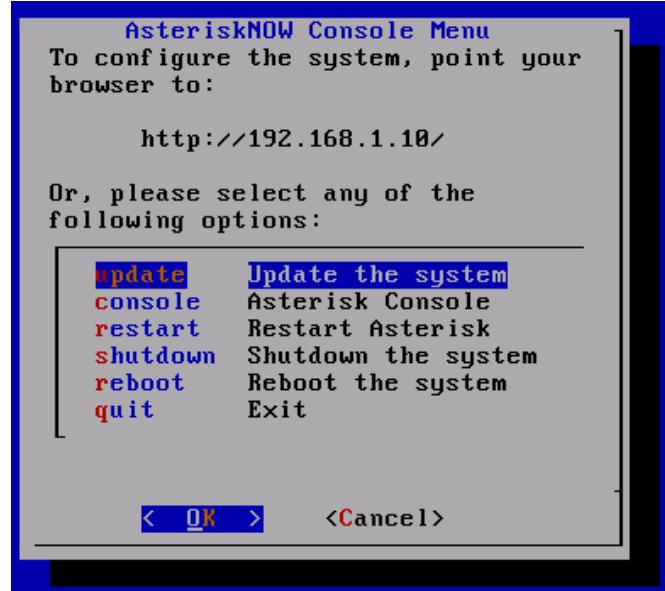
- At boot up you will see the following screen, if the installation was successful.



- You can press **enter** to start the system or you can wait a few seconds, for the system to start up automatically.
- The following is a first boot message.



### 7.1. The AsteriskNOW console Menu:

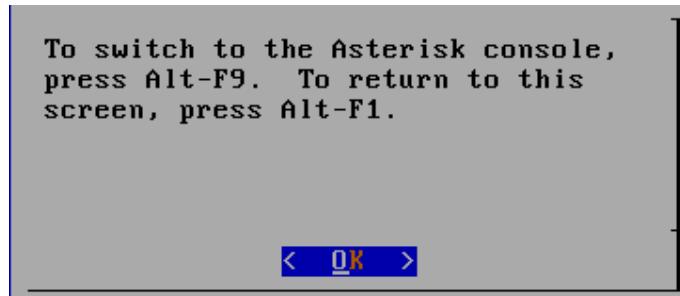


- Make a note of the Ip address displayed in the AsteriskNOW console menu. You need to use it to access the web menu later.

### 7.1.1. Update menu:

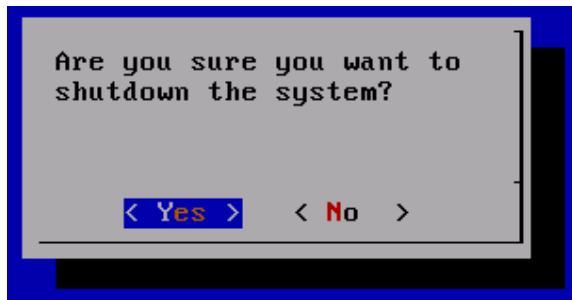
Use the update menu to update your system manually.

### 7.1.2. Console Menu:



### 7.1.3. Restart Menu:

#### **7.1.4. Shutdown Menu:**



#### **7.1.5. Reboot Menu:**

#### **7.1.6. Quit Menu:**

## 8. Configuring AsteriskNOW:

Congratulations, you have made it this far.

To continue further, open up your Firefox web browser in a second computer and type in the Ip address of your AsteriskNow server.

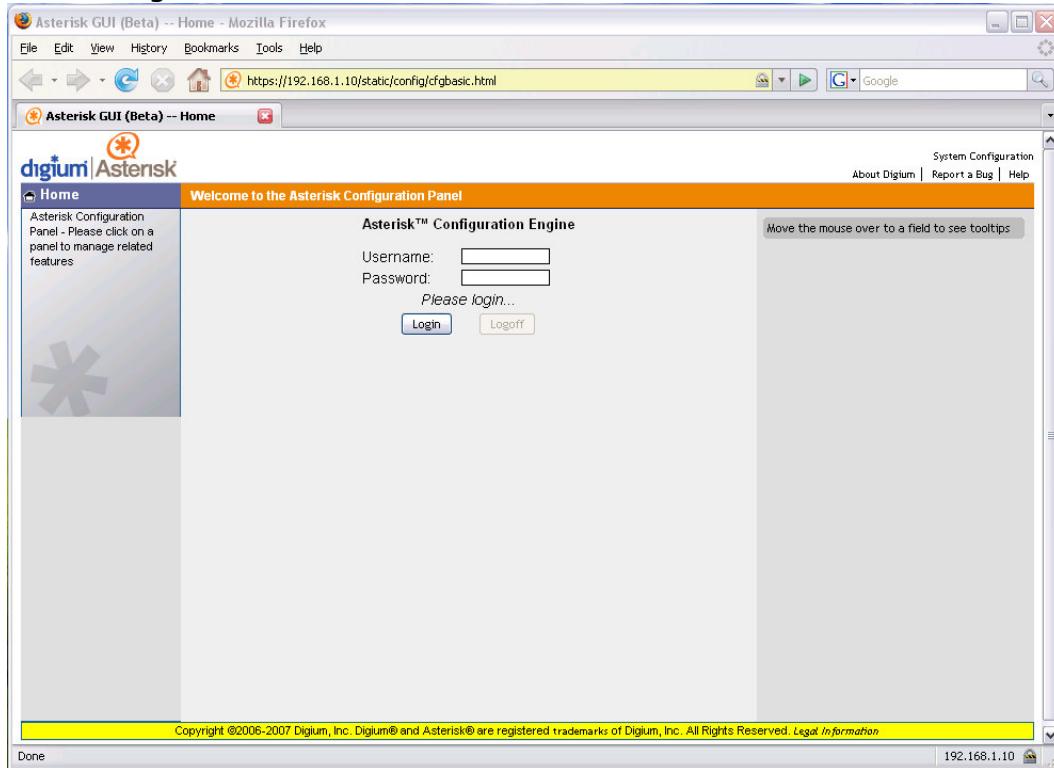
You can get the IP address by looking at the AsteriskNow console menu.

When you enter the web interface, you will be shown a warning message as below. You should choose **Accept this certificate temporarily for this session** and click on the **OK** button.



Then only you will be able to access the web based administration interface of AsteriskNOW.

In later stages, we can configure a proper website certificate, which will not raise any error warnings.



You need to login with the admin username to proceed any further.

**Asterisk™ Configuration Engine**

Username:       Password:

Please login...

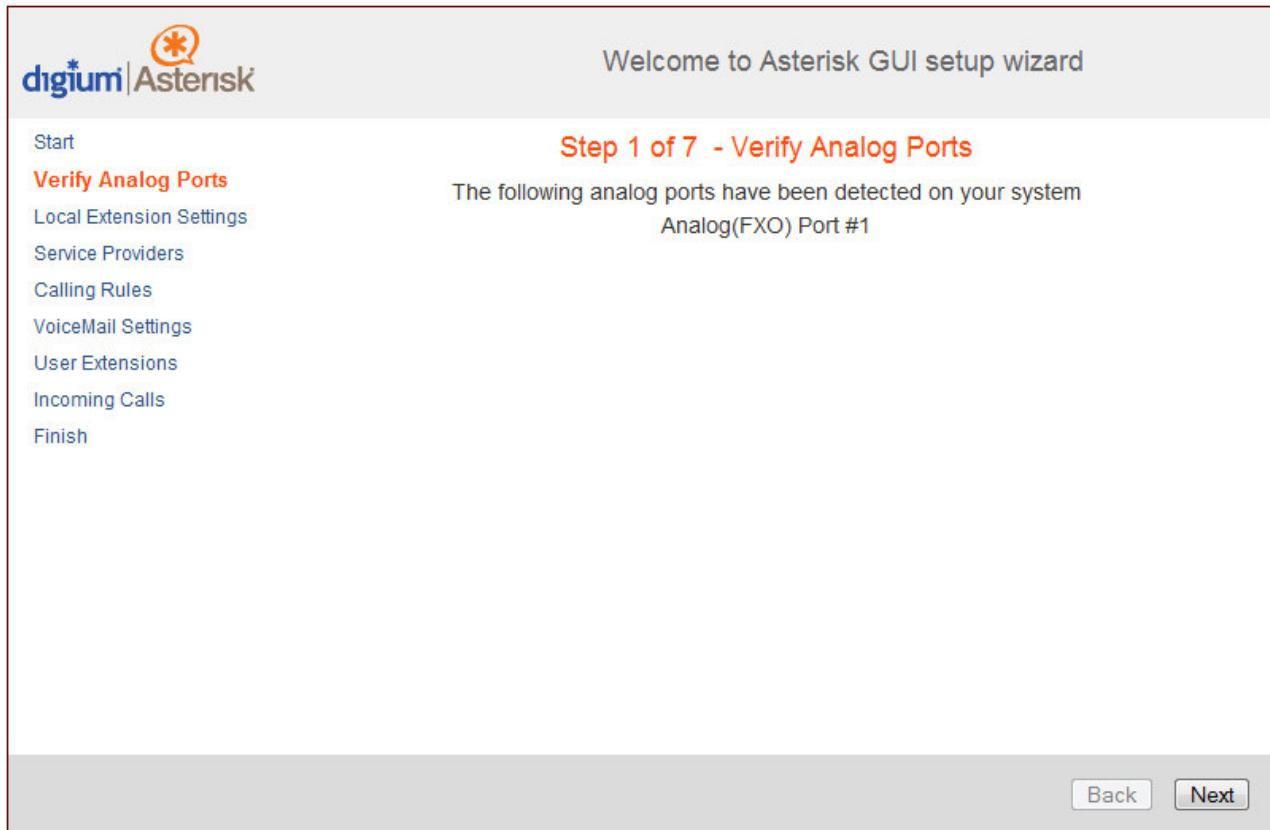
Username: admin

Password: xxxxx (the password you set when installing the system)

## **8.1. Setup Wizard:**

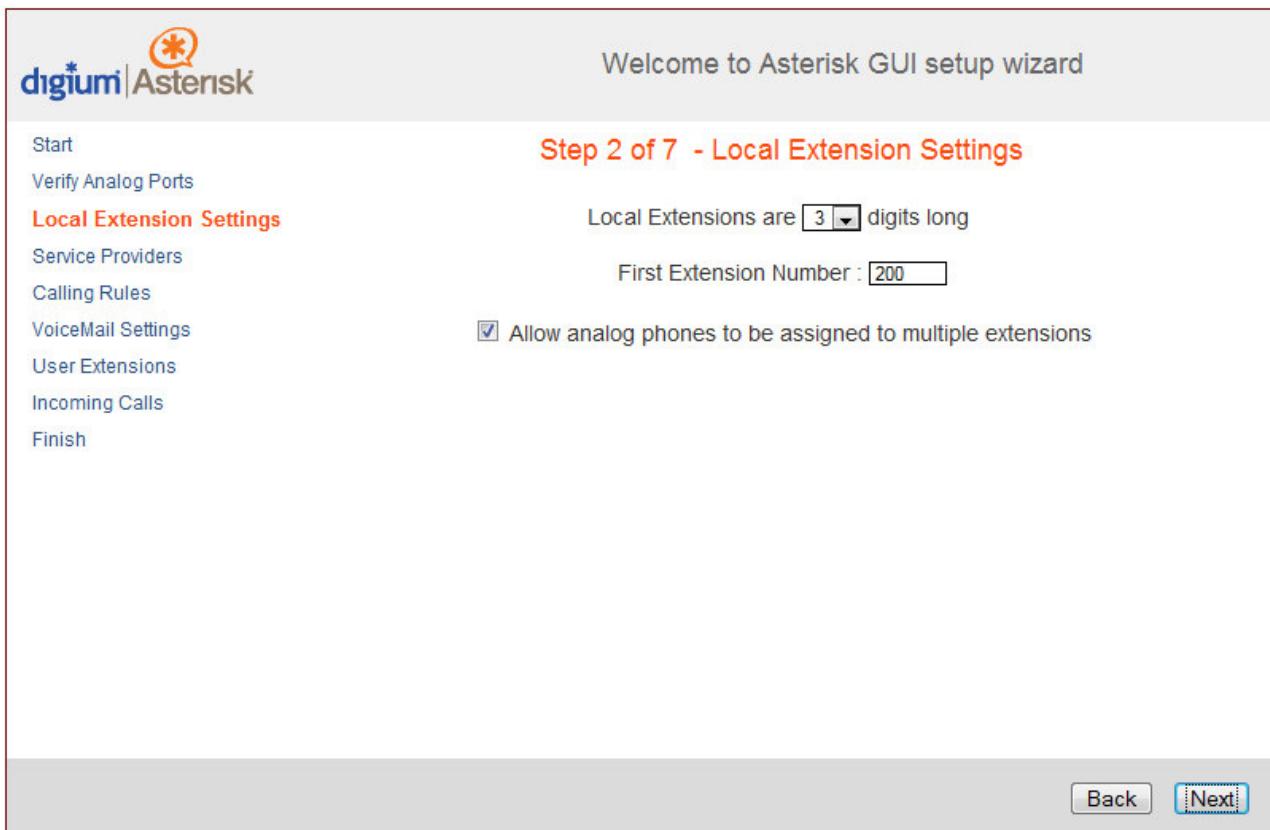
At first login, the Wizard is started automatically for you to walk through the steps. Just follow the prompts and finish the wizard's task.

### **8.1.1. Analog Ports:**



If you have any analog cards installed on your system, then the wizard will pick them up and prompt you to configure them.

### 8.1.2. Local Extension Settings:



- Here setup your local extensions length and the extensions starting point.
- Choose a pattern carefully.
- Changing these after entering your user accounts will break your system.
- Click **Next** to continue.

### 8.1.3. Service Providers:

In here you configure the service providers. That is, your voip service providers.

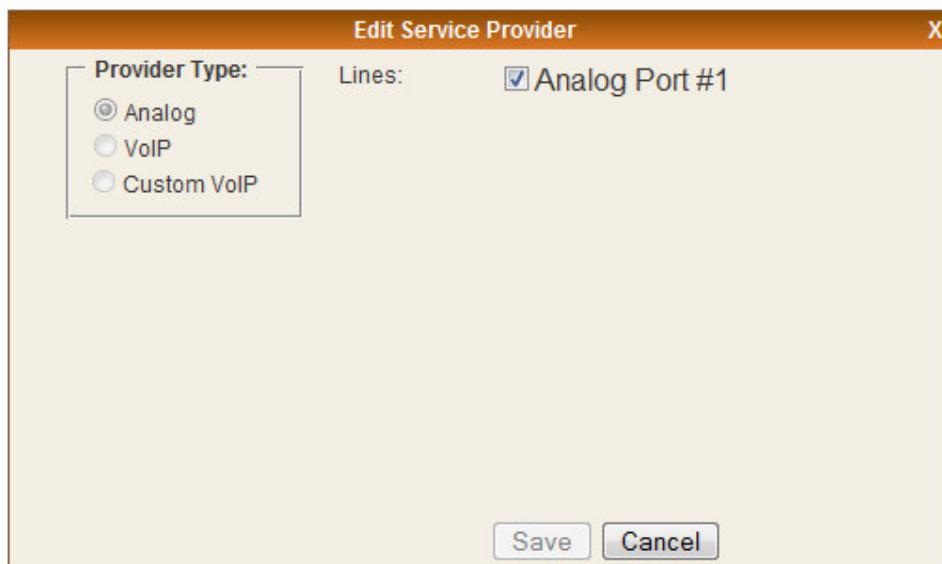
The screenshot shows the 'Step 3 of 7 - Service Providers' page of the Asterisk GUI setup wizard. On the left, there is a sidebar with the following menu items: Start, Verify Analog Ports, Local Extension Settings, **Service Providers** (which is bolded), Calling Rules, Voicemail Settings, User Extensions, Incoming Calls, and Finish. The main content area has a title 'Step 3 of 7 - Service Providers' and a message box containing the text 'A Service Provider is not defined' followed by 'Please click on the 'Add Service Provider' button to add a service provider'. At the bottom right of the content area is a button labeled 'Add Service Provider'. At the very bottom of the screen, there are 'Back' and 'Next' buttons.

If you want to dial out to the rest of the world then you need to have at least one provider, who will carry your calls.

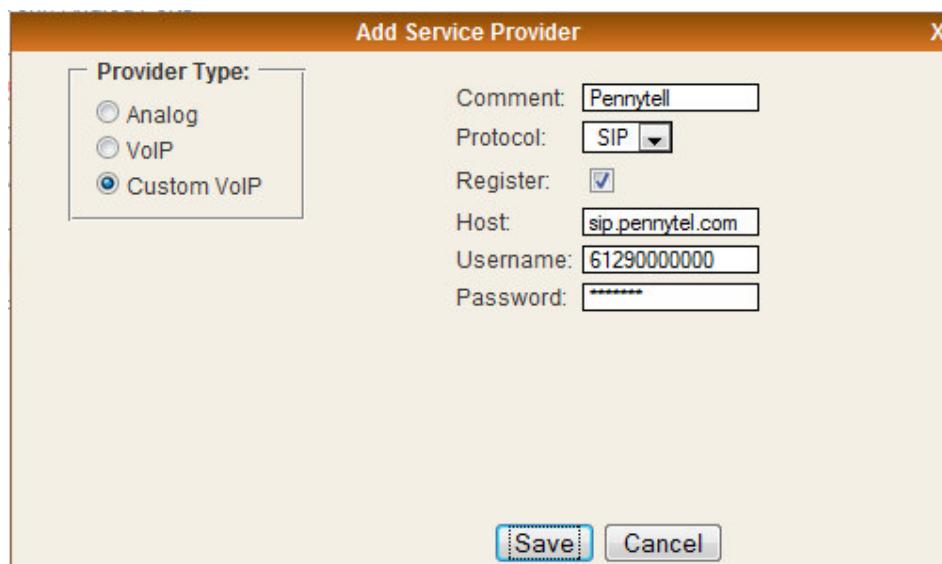
When you click on the Add Service Provider the following screen appears. You have a choice of selecting three types of service providers.

- 1) Analog
- 2) Voip
- 3) Custom Voip

Adding an Analog Port:



Adding a Custom Voip Service provider:





## Welcome to Asterisk GUI setup wizard

Start  
Verify Analog Ports  
Local Extension Settings  
**Service Providers**  
Calling Rules  
VoiceMail Settings  
User Extensions  
Incoming Calls  
Finish

### Step 3 of 7 - Service Providers

S.No	Service Provider	Type	
1	Custom - PennyToll	Custom Voip	<a href="#">Edit</a> <a href="#">Delete</a>

[Add Service Provider](#)

[Back](#) [Next](#)

### **8.1.4. Calling Rules:**

The calling rules are the most important settings after the service provider configuration. Here you need to define your outgoing calling rules (patterns).

#### **Rest of World:**

*To be done, if anyone has anything to contribute, then please contact me.*

S.No	RuleName	Dial Pattern	Call Using	Options
1	Longdistance	Begins with 91 and followed by 11 digits	Select a ServiceProvider	<a href="#">Edit</a> <a href="#">Delete</a>
2	IAXTEL	Begins with 91700 and followed by 8 digits	Select a ServiceProvider	<a href="#">Edit</a> <a href="#">Delete</a>
3	Local	Begins with 9256 and followed by 8 digits	Select a ServiceProvider	<a href="#">Edit</a> <a href="#">Delete</a>
4	International	Begins with 9011 and followed by 8 digits	Select a ServiceProvider	<a href="#">Edit</a> <a href="#">Delete</a>
5	Local	Begins with 9 and followed by 8 digits	Select a ServiceProvider	<a href="#">Edit</a> <a href="#">Delete</a>
6	911	Exactly matches 911.	Select a ServiceProvider	<a href="#">Edit</a> <a href="#">Delete</a>

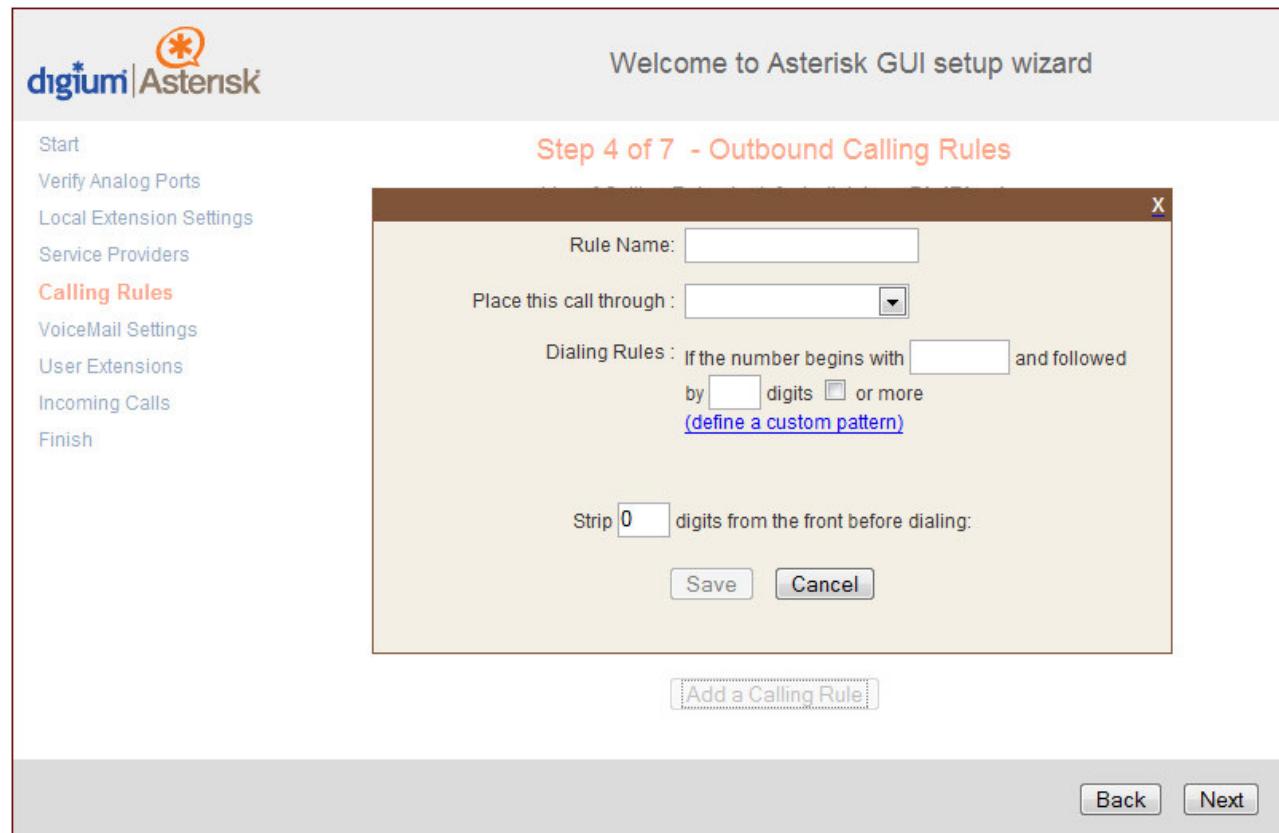
#### **Australia:**

*The notes described in this section relate to calling patterns as applicable to an Australian setup for Sydney Location, you will need to adapt the settings to suit the needs for your own location and country.*

For a Standard setup we will be using the following calling patterns:

- 1) Emergency Numbers (000)
- 2) Local Numbers (9XXXXXXX)
- 3) Local Numbers 1 (8XXXXXXX)
- 4) Long Distance (0X XXXXXXXX)
- 5) Mobile Phones (04XXXXXXXX)
- 6) International (0011 XXXXXXXXXXXXX)

Click on Add a Calling Rule button to add a new calling rule for your installation or you may edit the existing calling rules and change them to suit your local calling patterns.



Rule for local Numbers: (For Starting with 9)



## Rule for local Numbers: (For Starting with 8)

The screenshot shows a window titled "Edit Rule". The "Rule Name" field contains "Local2". The "Place this call through:" dropdown is set to "Custom - Pennytell". The "Dialing Rules:" section states: "If the number begins with 8" followed by "by 7 digits or more" and a link "(define a custom pattern)". Below this, the "Strip" field is set to "0" digits. At the bottom are "Save" and "Cancel" buttons.

Rule Name: Local2

Place this call through: Custom - Pennytell

Dialing Rules: If the number begins with 8 and followed by 7 digits or more  
(define a custom pattern)

Strip 0 digits from the front before dialing:

Save Cancel

## Rule for Long-distance National Call:

The screenshot shows a window titled "Edit Rule". The "Rule Name" field contains "Longdistance". The "Place this call through:" dropdown is set to "Custom - Pennytell". The "Dialing Rules:" section states: "If the number begins with 0" followed by "by 9 digits or more" and a link "(define a custom pattern)". Below this, the "Strip" field is set to "0" digits. At the bottom are "Save" and "Cancel" buttons.

Rule Name: Longdistance

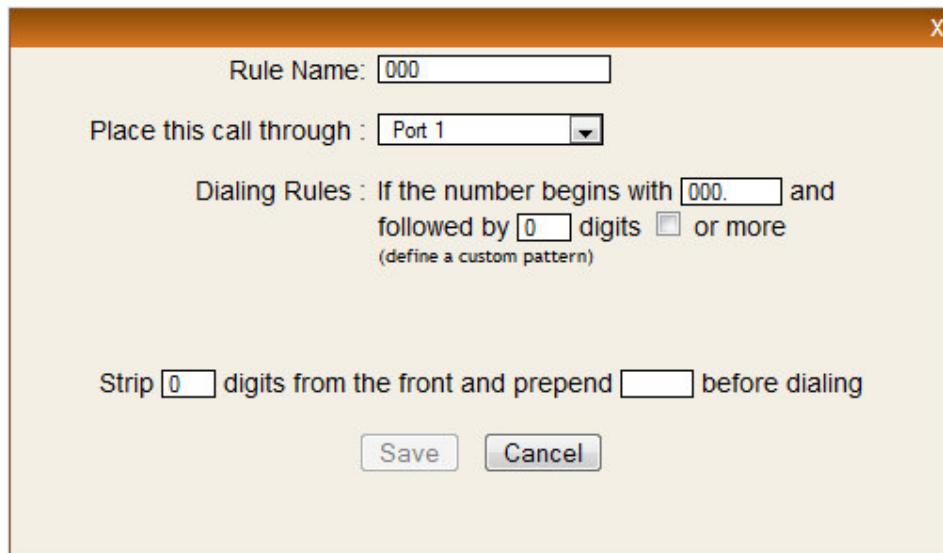
Place this call through: Custom - Pennytell

Dialing Rules: If the number begins with 0 and followed by 9 digits or more  
(define a custom pattern)

Strip 0 digits from the front before dialing:

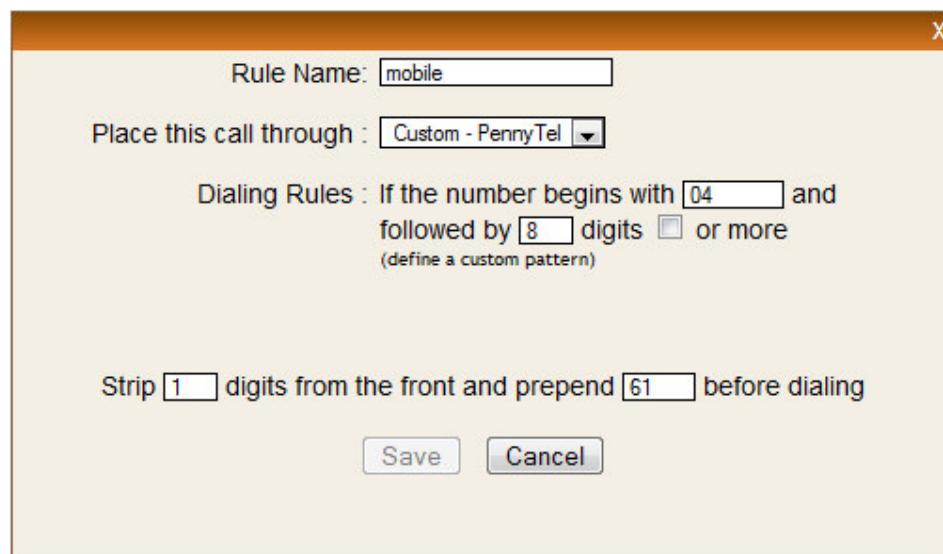
Save Cancel

## Rule for Emergency calls:



Remember it is always recommended to use your local landline to route your Emergency calls. Here I have chosen my analog Port 1 to route all of my emergency calls that start with 000. My analog port is an Xp100 card a very old version, but it still works.

## Rule for Mobile Calls:



## Rule for International Calls:

Rule Name: International

Place this call through : Custom - Pennytell

Dialing Rules : If the number begins with 0011 and followed by 7 digits or more  
[\(define a custom pattern\)](#)

Strip 4 digits from the front before dialing:

After you have configured all your rules your calling rules screen should be similar to the following screen.

Welcome to Asterisk GUI setup wizard

Step 4 of 7 - Outbound Calling Rules

List of Calling Rules in default dialplan - DialPlan1

S.No	RuleName	Dial Pattern	Call Using	Options
1	local	Begins with 9 and followed by 7 digits	Custom - Pennytell	<a href="#">Edit</a> <a href="#">Delete</a>
2	Local2	Begins with 8 and followed by 7 digits	Custom - Pennytell	<a href="#">Edit</a> <a href="#">Delete</a>
3	Longdistance	Begins with 0 and followed by 9 or more digits	Custom - Pennytell	<a href="#">Edit</a> <a href="#">Delete</a>
4	000	Exactly matches 000	Custom - Pennytell	<a href="#">Edit</a> <a href="#">Delete</a>
5	International	Begins with 0011 and followed by 7 or more digits	Custom - Pennytell	<a href="#">Edit</a> <a href="#">Delete</a>

Click on next to continue.

### 8.1.5. Voicemail:

**Welcome to Asterisk GUI setup wizard**

**Step 5 of 7 - VoiceMail Settings**

Extension for checking messages:	<input type="text" value="850"/>
Attach recordings to e-mail:	<input checked="" type="checkbox"/>
Maximum messages per folder:	<input type="text" value="10"/> <input type="button" value="▼"/>
Maximum message time	<input type="text" value="1 minute"/> <input type="button" value="▼"/>
Max greeting (seconds)	<input type="text" value="60"/>

**VoiceMail Settings**

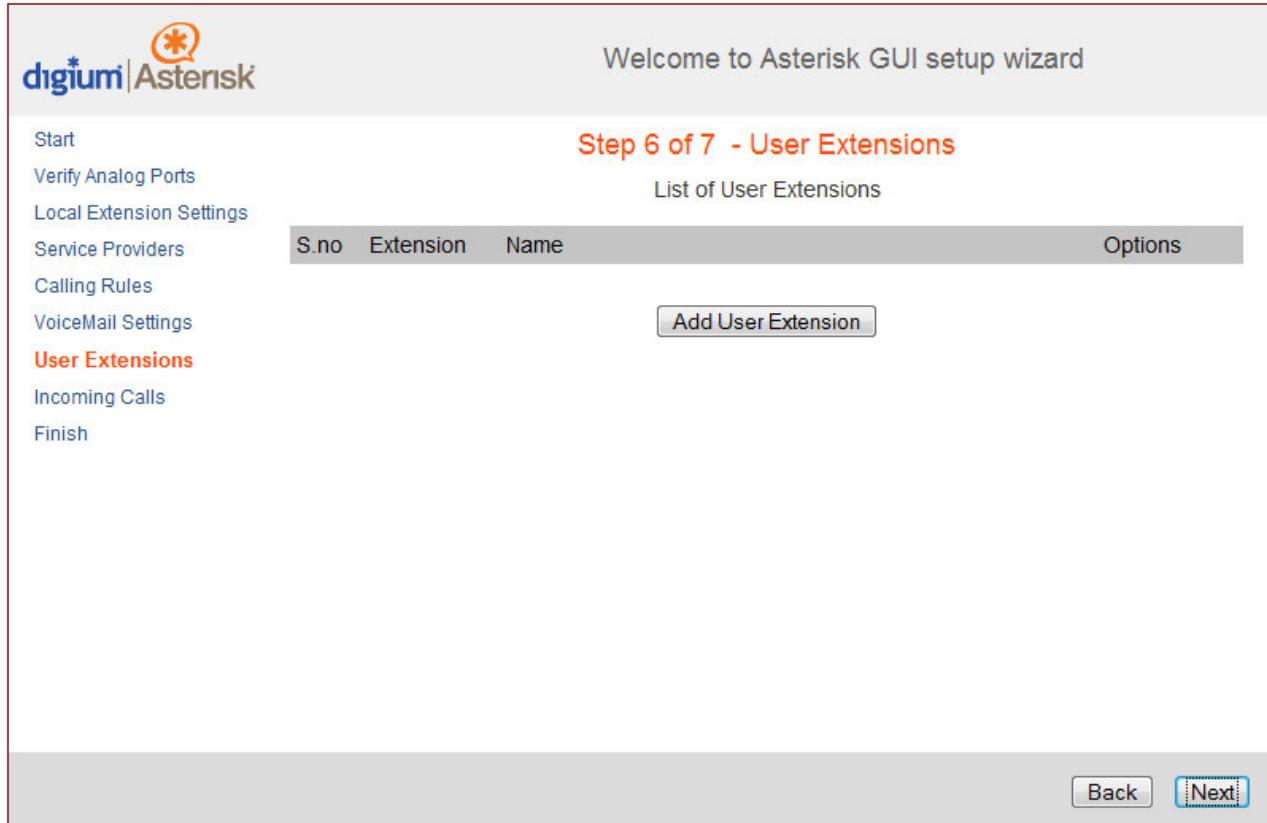
[Start](#)   [Verify Analog Ports](#)   [Local Extension Settings](#)   [Service Providers](#)   [Calling Rules](#)   [User Extensions](#)   [Incoming Calls](#)   [Finish](#)

[Back](#) [Next](#)

In this section you need to specify your voicemail settings.  
Take the time to setup these as well and then click on next to continue.

### 8.1.6. Extensions:

When you start off with a fresh installation you will not have any predefined extensions. Therefore your User Extensions will be blank as shown below.



Click on Add User Extension to start the form to create new extensions as below.

This is a dialog box for adding a user extension. It contains the following fields:

User Extension:	200	Full Name:	Anil Madikonda
Password:	200	Email:	anil@madikonda.com
Caller id:	200	Analog Phone:	No Analog lines installed.
Dial Plan:	DialPlan1	VoiceMail:	<input checked="" type="checkbox"/>
In Directory:	<input checked="" type="checkbox"/>	SIP:	<input checked="" type="checkbox"/>
CTI:	<input type="checkbox"/>	IAX:	<input checked="" type="checkbox"/>
CallWaiting:	<input checked="" type="checkbox"/>	3-Way Calling:	<input checked="" type="checkbox"/>
Is Agent:	<input type="checkbox"/>		

At the bottom are 'Save' and 'Cancel' buttons.

Once you have entered all the information relevant for this extension, then click on the save button and then keep creating additional user extensions as you need.

Once you have created all the extensions that you need, then your screen should look similar to this.

The screenshot shows the 'Asterisk GUI setup wizard' interface. At the top left is the 'digium|Asterisk' logo. To the right, the title 'Welcome to Asterisk GUI setup wizard' is displayed. Below the title, the heading 'Step 6 of 7 - User Extensions' is shown in red. A sub-header 'List of User Extensions' is also present. On the left side, there is a vertical navigation menu with the following items: Start, Verify Analog Ports, Local Extension Settings, Service Providers, Calling Rules, VoiceMail Settings, **User Extensions** (which is highlighted in red), Incoming Calls, and Finish. The main content area contains a table titled 'List of User Extensions'. The table has columns for S.no, Extension, Name, and Options. The data in the table is as follows:

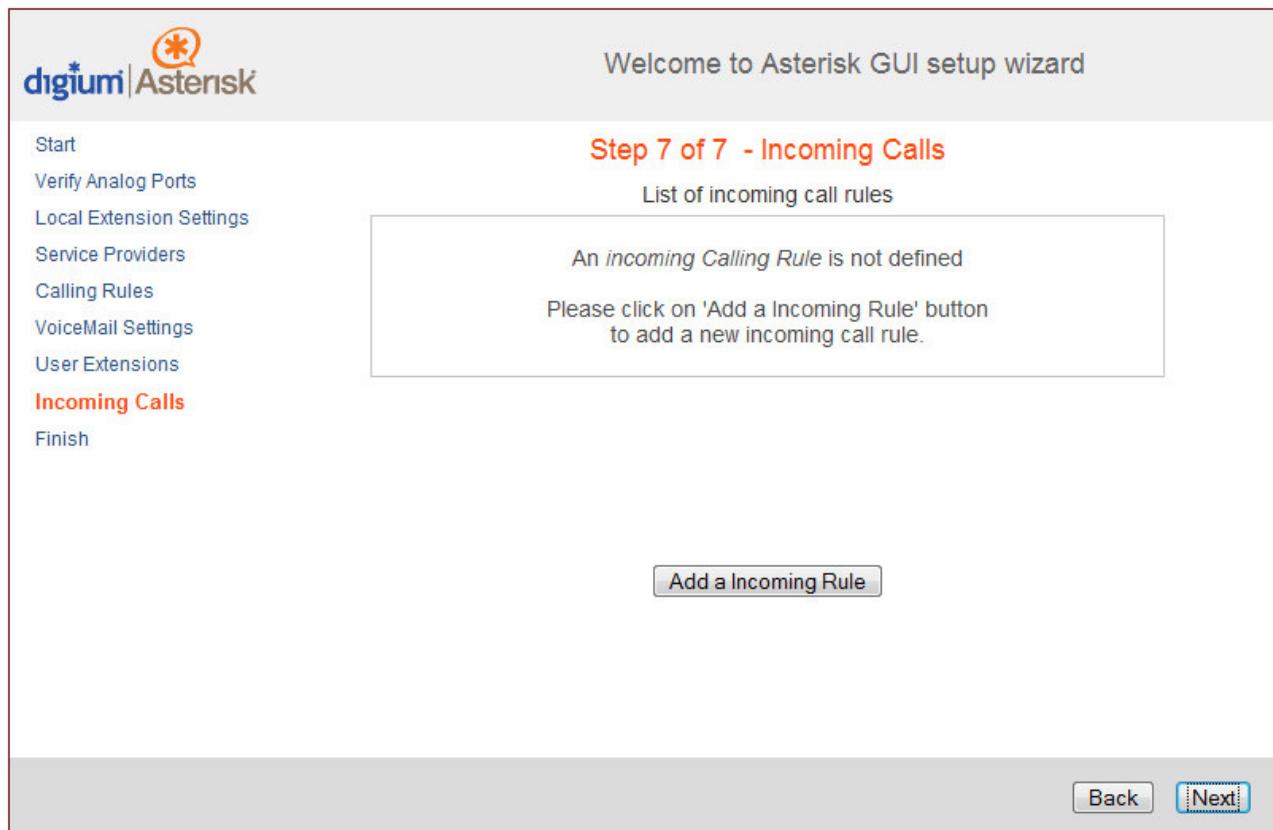
S.no	Extension	Name	Options
1	200	Anil Madikonda	<a href="#">Edit</a> <a href="#">Delete</a>
2	201	Line1	<a href="#">Edit</a> <a href="#">Delete</a>
3	202	Line2	<a href="#">Edit</a> <a href="#">Delete</a>
4	203	Sunitha	<a href="#">Edit</a> <a href="#">Delete</a>
5	204	Heeren Reddy	<a href="#">Edit</a> <a href="#">Delete</a>
6	205	MB Reddy	<a href="#">Edit</a> <a href="#">Delete</a>

Below the table is a button labeled 'Add User Extension'. At the bottom right of the main content area are 'Back' and 'Next' buttons.

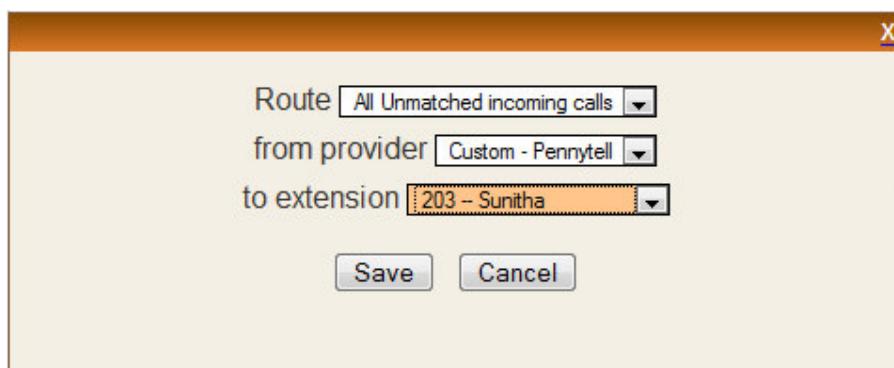
Once you are finished entering all of your user extensions, then click on next to continue.

### 8.1.7. Incoming Calls:

Incoming call rules define the action that you want to take for all incoming calls. There are now rules defined when you install. To create an incoming rule, click on Add a Incoming Rule to start the form.

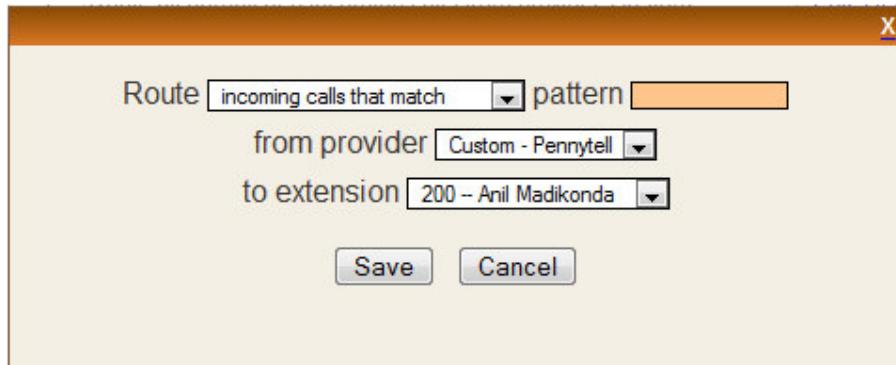


I have setup a default rule that applies to all unmatched calls. These call are routed to the extension 203. You can also setup a rule to match a certain incoming number and then route it to a different number.



Click on save button to save the rule.

Here in the next rule I am setting the rules to match a pattern and then route it to the extension 200.



Click on save to save the rule.

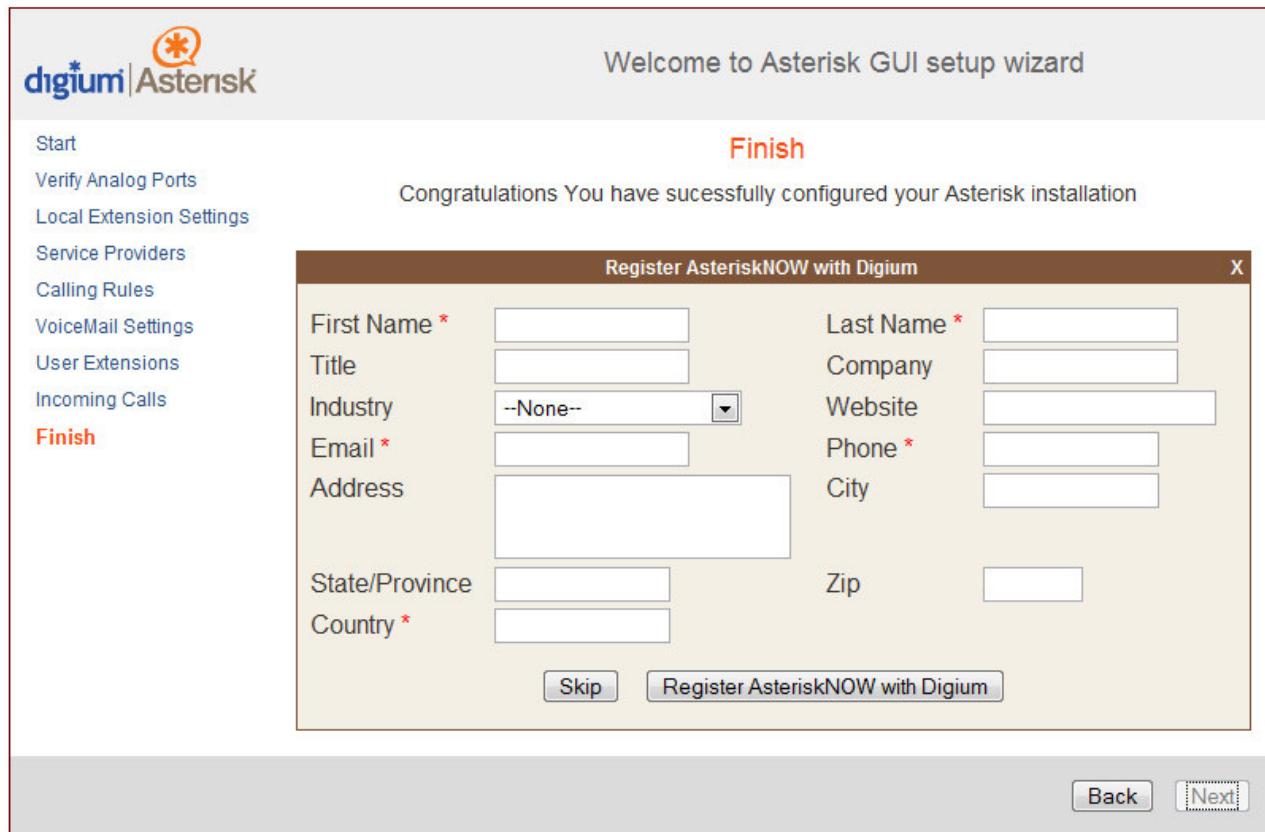
This screenshot shows the "Step 7 of 7 - Incoming Calls" screen of the Asterisk GUI setup wizard. The title bar says "Welcome to Asterisk GUI setup wizard". On the left is a sidebar with links: Start, Verify Analog Ports, Local Extension Settings, Service Providers, Calling Rules, VoiceMail Settings, User Extensions, **Incoming Calls** (which is highlighted in red), and Finish. The main area is titled "List of incoming call rules" and contains a table with one row:

1	Route all unmatched incoming calls from provider 'Custom - Pennytell' to extension '203'	<a href="#">Edit</a> <a href="#">Delete</a>
---	--	---

At the bottom right are "Add a Incoming Rule", "Back", and "Next" buttons.

After you have created your incoming rules, then click on next to continue.

### 8.1.8. Register your copy of AsteriskNOW:



You may now skip the registration screen by pressing on the skip button.  
Or you may continue the registration process and fill in the details.



## Welcome to Asterisk GUI setup wizard

Start  
Verify Analog Ports  
Local Extension Settings  
Service Providers  
Calling Rules  
VoiceMail Settings  
User Extensions  
Incoming Calls

**Finish**

**Finish**

Congratulations You have sucessfully configured your Asterisk installation

**Register AsteriskNOW with Digium**

First Name *	Anil	Last Name *	Madikonda
Title	Mr	Company	Madikonda.com
Industry	Consulting	Website	www.madikonda.com
Email *	anil@madikonda.cc	Phone *	61290xxxxxx
Address	PO Box 969 Strathfield	City	Sydney
State/Province	NSW	Zip	2135
Country *	Australia	<input type="button" value="Skip"/> <input type="button" value="Register AsteriskNOW with Digium"/>	

[Back](#) [Next](#)

Click on register AsteriskNow with Digium. This will bring up a small window as below, while your details are sent to digium.





## Welcome to Asterisk GUI setup wizard

Start

Verify Analog Ports

Local Extension Settings

Service Providers

Calling Rules

VoiceMail Settings

User Extensions

Incoming Calls

**Finish**

**Finish**

Congratulations You have sucessfully configured your Asterisk installation

You have been registered succesfully !!

[Back](#) [Next](#)

Click **next** to finish the Wizard setup process.

## **8.2. Asterisk Configuration Panel:**

Asterisk Configuration  
Panel - Please click on a  
panel to manage related  
features

- [Home](#)
- [Users](#)
- [Conferencing](#)
- [Voicemail](#)
- [Call Queues](#)
- [Service Providers](#)
- [Calling Rules](#)
- [Incoming Calls](#)
- [Voice Menus](#)
- [Time Based Rules](#)
- [Call Parking](#)
- [Ring Groups](#)
- [Record a Menu](#)
- [Active Channels](#)
- [Graphs](#)
- [System Info](#)
- [Asterisk Logs](#)
- [File Editor](#)
- [Asterisk CLI](#)
- [Backup](#)
- [Options](#)

The screenshot shows the Asterisk Configuration Engine login interface. On the left is a sidebar with a blue header "digium|Asterisk" and a list of configuration options. The main area has an orange header "Welcome to the Asterisk Configuration Panel". Below the header is the "Asterisk™ Configuration Engine" title. It contains fields for "Username" (admin) and "Password" (\*\*\*\*\*), both with validation messages. A "Connected!" message with a checkmark is displayed. Buttons for "Login" and "Logoff" are at the bottom. The top right corner has links for "System Configuration", "About Digium", "Report a Bug", and "Help". Buttons for "Activate Changes" and "Logout" are also present. A tooltip "Move the mouse over to a field to see tooltips" is shown in the top right corner.

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### **8.3. User & Phone Configuration:**

Users is a short cut for quickly adding and removing all the necessary configuration components for any new phone.

User and Phone Configuration

Activate Changes | Logout

User Extensions:

- 200 - Anil Madikonda**
- 201 -- Line1
- 202 -- Line2
- 203 -- Sunitha
- 204 -- Heeren Reddy
- 205 -- MB Reddy
- 850 -- Check Voicemail
- o -- Custom

Extension:  Name:  Password:  VM Password:   
 E-mail:  Caller ID:   
 Analog Phone: *No Analog lines detected.* Dial Plan:

3-Way Calling: Check this option if the User or Phone should have 3-Way Calling capability.

Extension Options:

<input checked="" type="checkbox"/> Voicemail	<input checked="" type="checkbox"/> In Directory
<input checked="" type="checkbox"/> SIP	<input checked="" type="checkbox"/> IAX
<input type="checkbox"/> CTI	<input type="checkbox"/> Is Agent
<input checked="" type="checkbox"/> Call Waiting	<input checked="" type="checkbox"/> 3-Way Calling
<input type="checkbox"/> Can Reinvite	<input type="checkbox"/> NAT

Codecs  
disallow:  DTMFMode:

New | Delete | Save | Cancel | BuyNow

Need a phone?  
Click on the BuyNow button to purchase a phone directly from the Asterisk Configuration GUI.

## General:

**Extension:** The numbered extension, i.e. 1234, that will be associated with this particular User / Phone.

**Name:** A character-based name for this user, i.e. "Bob Jones"

**Password:** The password for the user's sip/iax account , Ex: "12u3b6"

**VM Password:** Voicemail Password for this user, Ex: "1234".

**E-Mail:** The e-mail address for this user, i.e. [bobjones@bobjones.null](mailto:bobjones@bobjones.null)

**Caller ID:** The Caller ID (CID) string used when this user calls another user or number, i.e. "800-555-1234"

**Dial Plan:** Please choose the Calling Rule plan for this user as defined under the "Calling Rules" option to the left.

## Extension Options:

**Voicemail:** Check this box if the user should have a voicemail account.

**In Directory:** Check this option if the user is to be listed in the telephone directory.

**Session Initiation Protocol** Check this option if the User or Phone is using SIP or is a SIP device.

**InterAsterisk eXchange Protocol:** Check this option if the User or Phone is using IAX or is an IAX device.

**Computer Telephony Integration:** Check this option if the user is allowed to connect client applications to the Asterisk server.

**Call Waiting:** Check this option if the User or Phone should have Call-Waiting capability.

**3-Way Calling:** Check this option if the User or Phone should have 3-Way Calling capability.

**Is Agent:** Check this option if this User or Phone is an Call Queue Member (Agent)



## 8.4. Conferencing:

MeetMe conference bridging allows quick, ad-hoc conferences with or without security.

Conference Bridge Extensions Configuration

Activate Changes Logout

Conference Bridges:

- 200 -- Anil Madikonda
- 201 -- Line1
- 202 -- Line2
- 203 -- Sunitha
- 204 -- Heeren Reddy
- 205 -- MB Reddy
- 206 -- Conference Bridge**
- 850 -- Check Voicemail
- o -- Custom

Extension:  Room Override:

Play Hold Music for First Caller: Checking this option causes Asterisk to play Hold Music to the first user in a conference, until another user has joined the same conference.

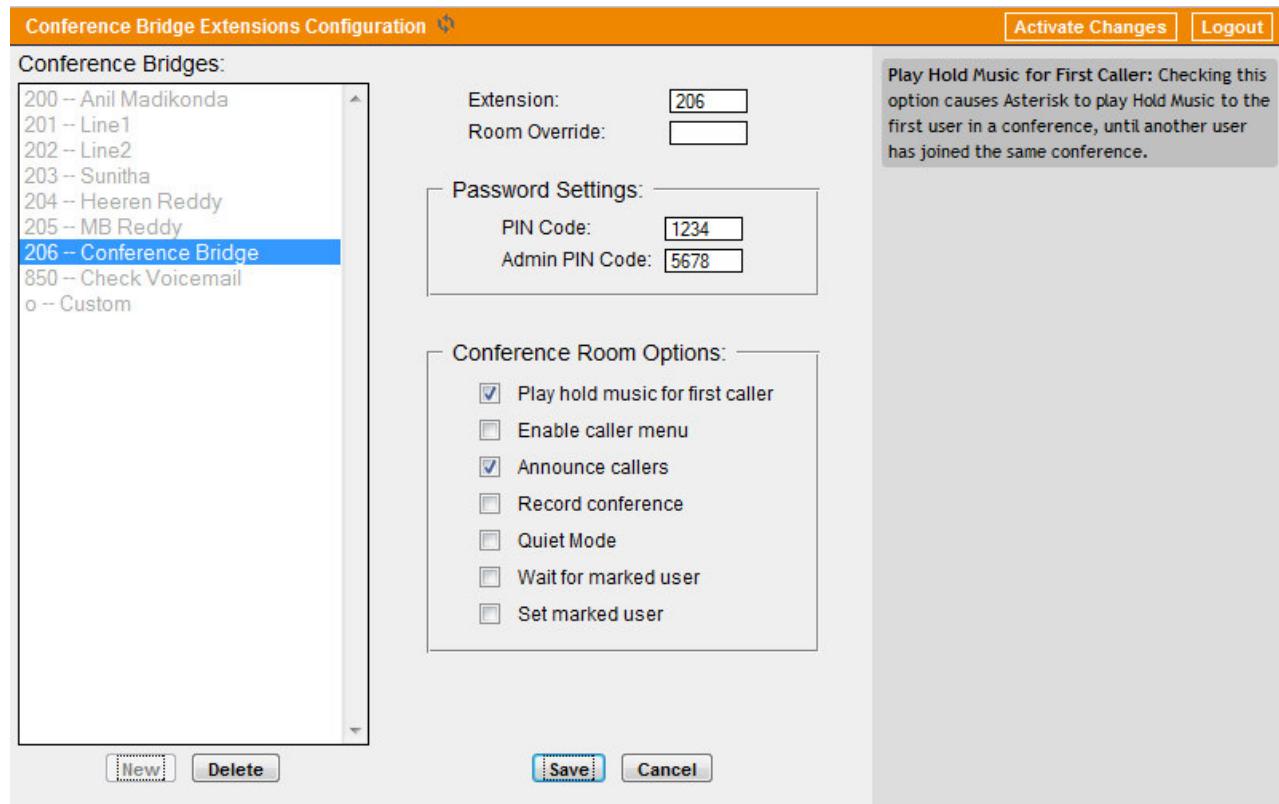
Password Settings:

PIN Code:  Admin PIN Code:

Conference Room Options:

- Play hold music for first caller
- Enable caller menu
- Announce callers
- Record conference
- Quiet Mode
- Wait for marked user
- Set marked user

New Delete Save Cancel



### General:

**Extension:** This is the number dialed to reach this Conference Bridge.

**Room Override:** This option allows the entry of a secondary extension that may be used to access this Conference Bridge. This is useful in the event that one wants to set a separate extension, having different options, to access the same Bridge.

### Password Settings:

**Personal Identification Number:** Defining this option, i.e. "1234" sets a code that must be entered in order to access the Conference Bridge.

**Administrator PIN Code:** Defining this option sets a PIN for Conference Administrators.

### Conference Room Options:

**Play Hold Music for First Caller:** Checking this option causes Asterisk to play Hold Music to the first user in a conference, until another user has joined the same conference.

**Enable Caller Menu:** Checking this option allows a user to access the Conference Bridge menu by pressing the \* "Asterisk" key on their dialpad.

**Announce Callers:** Checking this option announces, to all Bridge participants, the joining of any other participants.

**Record Conference:** Record this conference in a WAV format. Default filename is meetme-conf-rec- \${Conference Number}-\$({UNIQUEID}).

**Quiet Mode:** This option enables Quiet mode. If this option is checked, all users entering this conference will be marked as quiet, and will be in Listen-Only mode.

**Wait for Market User:** If this option is set, then users joining the conference will not be able to speak to one-another until the marked user has joined the conference.

**Set Marked User:** This option sets the person that enters the bridge using this extension as Marked. This option works in conjunction with the above "Wait for marked user" option.

## 8.5. Voicemail Configuration:

General settings for voicemail

The screenshot shows the 'Voicemail Configuration' page with the following details:

- VoiceMail Settings:** A list of extensions:
  - 200 -- Anil Madikonda
  - 201 -- Line1
  - 202 -- Line2
  - 203 -- Sunitha
  - 204 -- Heeren Reddy
  - 205 -- MB Reddy
  - 850 -- Check Voicemail** (highlighted in blue)
  - o -- Custom
- Extension for checking messages:** 850
- Attach recordings to e-mail:**
- Max greeting (seconds):** 60
- Dial '0' for Operator:**
- Message Options:**
  - Message Format:** WAV (GSM)
  - Maximum messages :** 25 (per folder)
  - Max message time:** 1 minute
  - Min message time:** No minimum
- Playback Options:**
  - Send messages by e-mail only
  - Say message Caller-ID
  - Say message duration
  - Play envelope
  - Allow users to review
- Allow Users to Review:** Checking this option allows the caller leaving the voicemail the opportunity to review their recorded message before it is submitted as a voicemail message.
- Buttons:** Activate Changes, Logout, Save, Cancel

### General Setting:

**Extension for checking Message:** This option, i.e. "2345," defines the extension that Users call in order to access their voicemail accounts.

**Attach recording to e-mail:** This option defines whether or not voicemails are sent to the Users' e-mail addresses as attachments.

**Max Greeting:** Defining this option sets a maximum time for a users's voicemail away message.

**Dail 'O' for Operator:** Checking this option enables callers entering the voicemail application to dial '0' to back out of the application and be sent to a voicemail menu or operator.

### Message Options:

**Message Format:** This selection box controls the format in which messages are stored on the system and delivered by e-mail.

**Maximum messages per folder:** This select box sets the maximum number of messages that a user may have in any of their folders.

**Maximum Message Time:** This select box sets the maximum duration of a voicemail message. Message recording will not occur for times greater than this amount.

**Minimum message Time:** This select box sets the minimum duration of a voicemail message. Messages below this threshold will be automatically deleted.

## Playback Options:

**Send messages by e-mail only:** If this option is set, then voicemails will not be checkable using a Phone. Messages will be sent via e-mail, only.

**Say Message Caller-ID:** If this option is enabled, the Caller ID of the party that left the message will be played back before the voicemail message begins playing back.

**Say Message Duration:** If this option is set, the duration of the message will be played back before the voicemail message begins playing back.

**Play Envelope:** Selecting this option causes Asterisk not to play introductions about each message when accessing them from the voicemail application.

**Allow Users to Review:** Checking this option allows the caller leaving the voicemail the opportunity to review their recorded message before it is submitted as a voicemail message.

## 8.6. Call Queues:

Call queues allow calls to be sequenced to one or more agents.

Queue Extension Configuration

Activate Changes Logout

Queues:

- 200 -- Anil Madikonda
- 201 -- Line1
- 202 -- Line2
- 203 -- Sunitha
- 204 -- Heeren Reddy
- 205 -- MB Reddy
- 206 -- Queue 'General Queue'**
- 850 -- Check Voicemail
- o -- Custom

Queue:  Full Name:  Strategy:

Agents:  Anil Madikonda (200)  Sunitha (203)

Agents: This selection shows all Users defined as Agents in their User conf. Checking a User here makes them a member of the current Queue.

Queue Options:

- TimeOut:
- Wrapup Time:
- Max Len:
- Music On Hold:
- Auto Fill:
- Auto Pause:
- JoinEmpty:
- LeaveWhenEmpty:
- Report Hold Time:

New Delete Save Cancel

The screenshot shows the 'Queue Extension Configuration' window. In the 'Queues' list on the left, '206 -- Queue 'General Queue'' is selected. The main panel displays fields for 'Queue' (206), 'Full Name' (General Queue), and 'Strategy' (ringall). Under 'Agents', two users are selected: Anil Madikonda (200) and Sunitha (203). A tooltip explains that this selection shows all users defined as agents in their user configuration. Below these, 'Queue Options' are listed with checkboxes for various features like Auto Fill, Auto Pause, and LeaveWhenEmpty. At the bottom are 'New', 'Delete', 'Save', and 'Cancel' buttons, along with 'Activate Changes' and 'Logout' links at the top right.

**Queue:** This option defines the numbered extension that may be dialed to reach this Queue.

**Full Name:** This option defines a name for this Queue, i.e. "Sales"

**Strategy:** This option sets the Ringing Strategy for this Queue. The options are:

1. RingAll - Ring All available Agents until one answers.
2. RoundRobin - Take turns ringing each available Agent
3. LeastRecent - Ring the Agent which was least recently called
4. FewestCalls - Ring the Agent with the fewest completed calls
5. Random - Ring a Random Agent
6. RRmemory - RoundRobin with Memory, Rembers where it left off in the last ring pass

**Agents:** This selection shows all Users defined as Agents in their User conf. Checking a User here makes them a member of the current Queue.

## Queue Options:

**Timeout:** This option defines the time in seconds that an Agent's phone rings before the next Agent is rung, i.e. "15"

**Wrapup Time:** After a successful call, time time in seconds that an Agent remains free before another call is sent to them. Default is 0, which is No Delay.

**MaxLen:** This option sets the maximum number of callers that may wait in a Queue. Default is 0, Unlimited.

**Music On Hold:** Select the 'Music on Hold' Class for this Queue

**AutoFill** Defining this option causes the Queue, when multiple calls are in it at the same time, to push them to Agents simultaneously. Thus, instead of completing one call to an Agent at a time, the Queue will complete as many calls simultaneously to the available Agents.

**AutoPause:** Enabling this option pauses an Agent if they fail to answer a call.

**JoinEmpty:** Defining this option allows callers to enter the Queue when no Agents are available. If this option is not defined, callers will not be able to enter Queues with no available agents.

**LeaveWhenEmpty:** Defining this option forces all callers to exit the Queue if New Callers are also not able to Enter the Queue. This option should generally be set in concert with the JoinEmpty option.

**Report Hold Time:** Enabling this option causes Asterisk to report, to the Agent, the hold time of the caller before the caller is connected to the Agent.

## 8.7. Service Providers:

Service Providers are outbound lines used to allow the system to make calls to the real world. Trunks can be VoIP lines or traditional telephony lines.

The screenshot shows the 'List of Service Providers' page. At the top right are 'Activate Changes' and 'Logout' buttons. A note on the right says 'AutoPause: Enabling this option pauses an Agent if they fail to answer a call.' Below is a table with one row:

S.No	Service Provider	Type	Options
1	Custom - Pennyell	Custom Voip	<ul style="list-style-type: none"><li>Edit</li><li>Codecs</li><li>Advanced</li><li>Delete</li></ul>

At the bottom is an 'Add Service Provider' button.



**Analog/Voip Trunks:** Analog lines are attached to analog interfaces of the PBX using FXO cards. Voice over IP (VoIP) connections are provided by an Internet Telephony Service Provider (ITSP).

The screenshot shows the 'Codec Preferences' dialog box. It has two main sections: 'Allowed' and 'Disallowed'. The 'Allowed' section contains 'GSM', 'a-law', and 'u-law'. The 'Disallowed' section contains 'ILBC', 'SPEEX', 'G.726', 'ADPCM', 'LPC10', and 'G.729'. There is a checkbox for 'Disallow All' and two buttons at the bottom: 'update' and 'Cancel'.

## 8.8. Calling Rules:

The Calling Rules define dialing permissions and least cost routing rules.

The screenshot shows the 'Calling Rules' section of the AsteriskNOW web interface. At the top, there's a navigation bar with 'Calling Rules' and icons for 'Activate Changes' and 'Logout'. Below the navigation is a 'List of DialPlans' section with a dropdown menu set to 'DialPlan1' and buttons for 'new' and 'delete'. To the right of this is a note about adding a new calling rule. The main area shows a table titled 'List of Calling Rules in the selected DialPlan' with the following data:

S.No	RuleName	Dial Pattern	Call Using	Options
1	local	Begins with 9 and followed by 7 digits	Custom - Pennytell	Edit Delete
2	Local2	Begins with 8 and followed by 7 digits	Custom - Pennytell	Edit Delete
3	Longdistance	Begins with 0 and followed by 9 or more digits	Custom - Pennytell	Edit Delete
4	000	Exactly matches 000	Custom - Pennytell	Edit Delete
5	International	Begins with 0011 and followed by 7 or more digits	Custom - Pennytell	Edit Delete

At the bottom of the main area is a button labeled 'Add a Calling Rule'.

This is a modal dialog box for creating a new calling rule. It contains the following fields:

- Rule Name:** A text input field containing 'local'.
- Place this call through :** A dropdown menu set to 'Custom - Pennytell'.
- Dialing Rules :** A text area with the following description: 'If the number begins with  and followed by  digits  or more (define a custom pattern)'.
- Strip :** A text area with the description 'Strip  digits from the front and prepend  before dialing'.
- Buttons:** 'Save' and 'Cancel' at the bottom.

**Rule Name:** A name for this Calling Rule. Ex: 'Local' or 'Long Distance' etc.

**Place this call through :** Select a Service Provider through which this call should be placed.

**Dialing Rules:** Ex: If the number begins with '256' and followed by 7 digits or more

**Custom Pattern:** Ex: \_91NXXNXXXXXX

**Strip:** Strip 1 digits from the front and prepend 256 before dialing

## Dial Plan 2:

The screenshot shows the AsteriskNOW web interface for managing a dial plan. At the top, there's a navigation bar with tabs for 'Calling Rules' (selected), 'Dial Plans', 'Extensions', 'Queues', and 'Groups'. On the right of the navigation bar are buttons for 'Activate Changes' and 'Logout'. Below the navigation, there's a message box with the text: 'new: Add a new DialPlan. A DialPlan is a set of calling rules that can be assigned to one or more User Extensions.' In the main content area, there's a section titled 'List of DialPlans:' with a dropdown menu set to 'DialPlan2' and buttons for 'new' and 'delete'. Below this is a section titled 'List of Calling Rules in the selected DialPlan' which contains the message 'A Calling Rule is not defined' and a note: 'Please click on the 'Add a Calling Rule' button to add a Calling Rule'. At the bottom of this section is a button labeled 'Add a Calling Rule'.

### 8.9. Incoming Calling Rules:

Define how your incoming calls should be handled & configure DID (Direct inward Dialing)

**Add a Incoming Rule:** Define a new Rule for handling Incoming calls based on service provider and/or the number called.

**Incoming Calls**

[Activate Changes](#) [Logout](#)

### Incoming Call Rules

S.No	Incoming Rule	Options
1	Route all unmatched incoming calls from provider 'Custom - PennyToll' to '203 -- Sunitha'	<a href="#">Edit</a> <a href="#">Delete</a>

[Add a Incoming Rule](#)

Add a Incoming Rule: Define a new Rule for handling Incoming calls based on service provider and/or the number called.

### 8.10. Voice Menu Configuration:

Menus allow for more efficient routing of calls from incoming callers. Also known as IVR (Interactive Voice Response) menus or Digital Receptionist

**Voice Menus Configuration**

[Activate Changes](#) [Logout](#)

Voice Menus:

<b>VoiceMenu - mainmenu</b>	<p>Name: <input type="text" value="mainmenu"/> Extension: <input type="text"/></p> <p>Steps:</p> <p style="margin-left: 20px;">Up</p> <p style="margin-left: 20px;">Down</p> <p style="margin-left: 20px;">Answer the Call Play 'thank-you-for-calling' &amp; Listen for KeyPress Play 'if-u-know-ext-dial' &amp; Listen for KeyPress Play 'otherwise' &amp; Listen for KeyPress Play 'pls-hold-while-try' &amp; Listen for KeyPress</p> <p>Add a new Step:  <input type="button" value="-- Select --"/> <input type="button" value="Add"/> <input type="button" value="Delete"/>  <input checked="" type="checkbox"/> Dial other Extensions?</p> <p><b>'Keypress' Events</b></p> <table border="1"> <thead> <tr> <th>Key</th> <th>Action</th> </tr> </thead> <tbody> <tr><td>0</td><td>Disabled</td></tr> <tr><td>1</td><td>Disabled</td></tr> <tr><td>2</td><td>Disabled</td></tr> <tr><td>3</td><td>Disabled</td></tr> <tr><td>4</td><td>Disabled</td></tr> <tr><td>5</td><td>Disabled</td></tr> <tr><td>6</td><td>Disabled</td></tr> </tbody> </table>	Key	Action	0	Disabled	1	Disabled	2	Disabled	3	Disabled	4	Disabled	5	Disabled	6	Disabled	<p>Keypress Events: Define the actions that occur when a user presses the corresponding digit.</p>
Key	Action																	
0	Disabled																	
1	Disabled																	
2	Disabled																	
3	Disabled																	
4	Disabled																	
5	Disabled																	
6	Disabled																	

[New](#) [Delete](#) [Save](#) [Cancel](#)

This is the main program setup of AsteriskNOW. How you setup your configuration here affects your whole pbx operations.

## **8.11. Time Based Rules:**

define call routing rules based on date and time

The screenshot shows a web-based configuration interface for 'Time Based Rules'. At the top, there's a navigation bar with 'Time Based Rules' (highlighted in orange), 'New Time Rule' (in blue), 'Activate Changes' (in green), and 'Logout' (in black). A message box displays 'No Previous Time Rules found !!' and instructs the user to click 'New Time Rule' to define a new rule. To the right, a note about 'Keypress Events' is visible.

Add a new rule to reflect your business timings, as below:

The 'Add new Time Rule' dialog box has a title bar 'Add new Time Rule' with a close button 'X'. It contains fields for 'Rule Name' (set to 'Business Hours') and a note '(Ex: July4)'. Below this is a section for 'Time & Date Conditions' with dropdown menus for 'Start Time' (08:00), 'End Time' (17:00), 'Start Day' (Sun), 'End Day' (Sat), 'Start Date' (01), 'End Date' (31), 'Start Month' (January), and 'End Month' (December). Under 'Destination', it lists 'if time matches: VoiceMenu mainmenu' and 'if time did not match: VoiceMenu mainmenu'. At the bottom are 'Save' and 'Cancel' buttons.

The screenshot shows a list of time rules. One rule is listed: '1 Business Hours'. To the right of the list are 'Edit' and 'Delete' buttons. A note about 'Keypress Events' is present on the right side.

## **8.12. Call Parking:**

Configure call parking features;

The 'Call Parking Preferences' dialog box has a title bar 'Call Parking Preferences' (highlighted in orange) and a note '(Ex: 701-720)' next to the 'What extensions to park calls on' field. It includes fields for 'Extension to Dial for Parking Calls' (700), 'What extensions to park calls on' (701-720), 'Number of seconds a call can be parked for' (45), 'Pickup Extension' (\*8), and 'Timeout for answer on attended transfer' (15). At the bottom are 'Save' and 'Cancel' buttons. A note about 'Keypress Events' is on the right.

### **8.13. Ring Groups:**

Define RingGroups to dial more than one extension

No Previous Ring Groups found !!

To create a ring group click on the 'New Ring Group' button

Keypress Events: Define the actions that occur when a user presses the corresponding digit.

Click on New Ring Group to create your new ring Group.  
Do not confuse this with the call queues.

#### Add Ring Group

Name: <input type="text" value="Sales"/>	Strategy: <input type="button" value="Ring all"/>
SIP/200 SIP/203	<input type="button" value="←"/> <input type="button" value="→"/> <input type="button" value="»»"/> <div style="border: 1px solid black; padding: 5px; margin-top: 10px;">           IAX2/200            SIP/201            IAX2/201            SIP/202            IAX2/202            IAX2/203            SIP/204            IAX2/204         </div>
<b>Ring Group Members</b> <b>Available Channels</b> Extension for this ring group (optional) : <input type="text"/> Ring (each/all) for these many seconds : <input type="text" value="30"/> If not answered <input checked="" type="radio"/> Goto Voicemail of this user <input type="radio"/> Goto an IVR menu <input type="radio"/> HangUp <input type="button" value="Save"/> <input type="button" value="Cancel"/>	

S.No	Ring Group	Options
1	Sales	<input type="button" value="Edit"/> <input type="button" value="Delete"/>

Keypress Events: Define the actions that occur when a user presses the corresponding digit.

### **8.14. Record a Menu:**

Allows you to record custom voicemenus over a phone

## List of Recorded VoiceMenus

*No Recorded menus found !!*

Please click on 'Record a new Voice Menu' button to record one.

Keypress Events: Define the actions that occur when a user presses the corresponding digit.

[Record a new Voice Menu](#)

## List of Recorded VoiceMenus

*No Recorded menus found !!*

Please click on 'Record a new Voice Menu' button to record one.

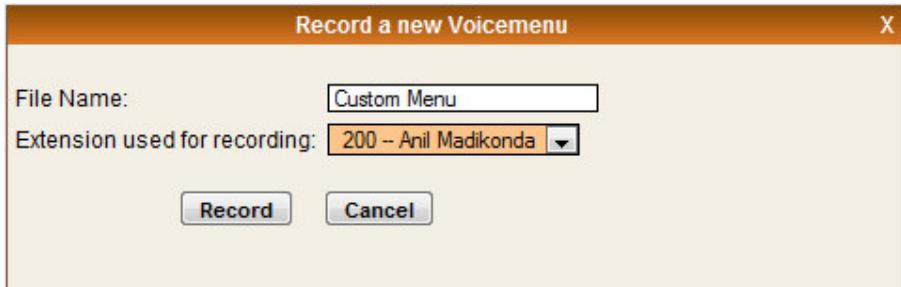
Record a new Voice Menu: Lets you record a new voice menu using any user extension device

## Record a new Voicemenu

X

File Name:	<input type="text" value="Custom Menu"/>
Extension used for recording:	<input type="text" value="200 -- Anil Madikonda"/> <input type="button" value="▼"/>

[Record](#) [Cancel](#)[Record a new Voice Menu](#)



**Filename:** File name under which the recorded file should be saved to. Ex: MainGreeting

**Extension used for recording:** Select a device through which this voice menu will be recorded.

### **8.15. Active Channels:**

Monitor and manage your active calls through this interface.

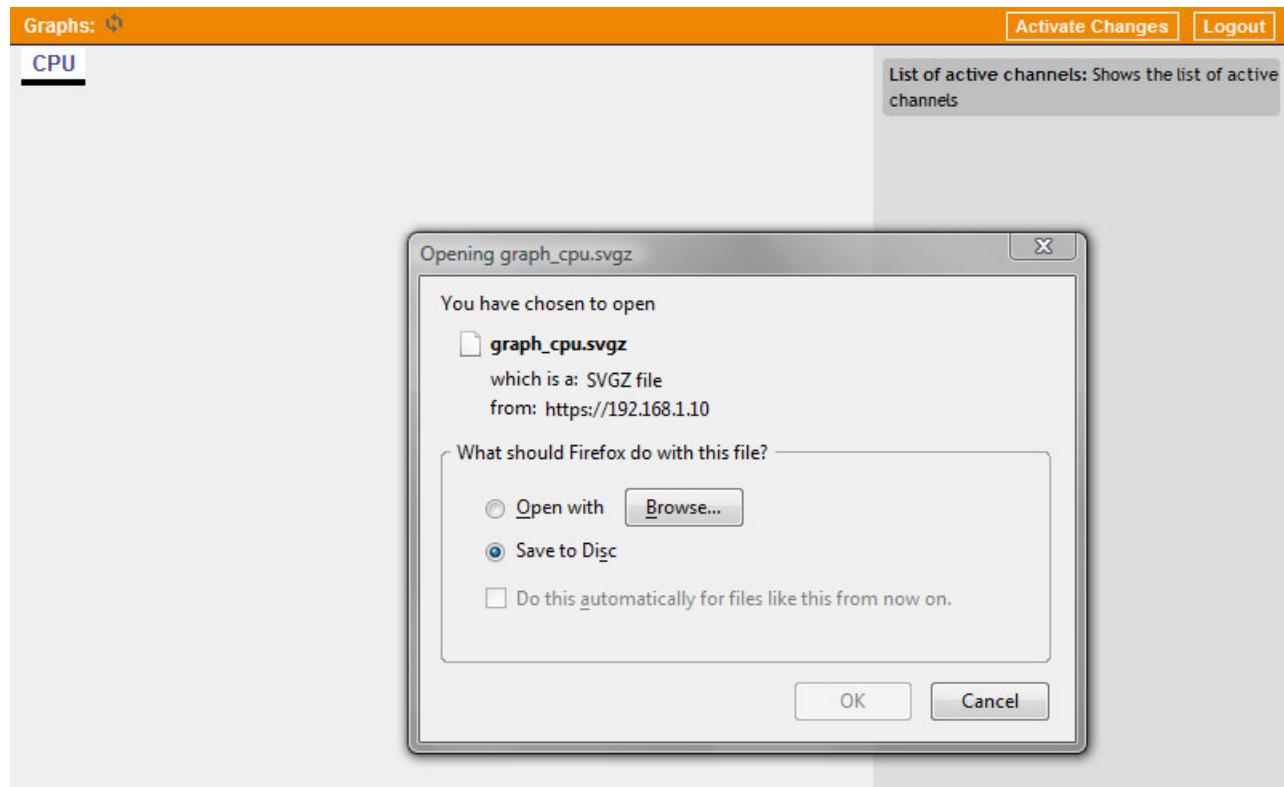
The screenshot shows the 'Channels Status' page. At the top, there are buttons for 'Activate Changes' and 'Logout'. Below that, it says 'Active Channels: Ready'. There are three buttons: 'Refresh', 'Transfer...', and 'Hangup'. A table header row is shown with columns: Channel, State, Caller, Location, and Link. Below the header, a message says 'No active channels'.

Channel	State	Caller	Location	Link
No active channels				

**List of active channels:** Shows the list of active channels

### **8.16. Graphs:**

View Graphs of your System Information.



## 8.17. System Information:

A screenshot of the 'System Information' page. The top navigation bar includes tabs for 'System Information' (selected), 'ifconfig', and 'Resources'. On the right, there are 'Activate Changes' and 'Logout' buttons. A tooltip for the 'General' tab states: 'General: Information about OS, Uptime, Asterisk, Date, Timezone and Hostname'. The main content area contains several sections: 'OS Version' (Linux localhost.localdomain 2.6.19.7-0.4.gcc3.4.x86.i686 #1 Tue May 1 13:00:52 EDT 2007 i686 i686 i386 GNU/Linux), 'Uptime' (12:06:09 up 3:40, 0 users, Load Average: 0.04, 0.11, 0.08), 'Asterisk Build' (Asterisk 1.4.9 Asterisk GUI-version Revision: 11425 \$), 'Server Date &amp; TimeZone' (Sun Nov 4 12:06:10 EST 2007), and 'Hostname' (localhost.localdomain).

Use this function to:

- Check your system information
- Ip config details
- Resources
- Asterisk logs

**System Information**  Activate Changes Logout

**ifconfig** ifconfig: Network devices information (ifconfig)

```
ifconfig:

eth0      Link encap:Ethernet HWaddr 00:03:FF:54:57:00
          inet addr:192.168.1.10 Bcast:192.168.1.255 Mask:255.255.255.
          inet6 addr: fe80::203:ffff:fe54:5700/64 Scope:Link
            UP BROADCAST RUNNING MULTICAST MTU:1500 Metric:1
            RX packets:8881 errors:0 dropped:0 overruns:0 frame:0
            TX packets:5191 errors:0 dropped:0 overruns:0 carrier:0
            collisions:0 txqueuelen:1000
            RX bytes:1102629 (1.0 Mb) TX bytes:1703708 (1.6 Mb)
            Interrupt:11 Base address:0x6000

lo       Link encap:Local Loopback
          inet addr:127.0.0.1 Mask:255.0.0.0
          inet6 addr: ::1/128 Scope:Host
            UP LOOPBACK RUNNING MTU:16436 Metric:1
            RX packets:85909 errors:0 dropped:0 overruns:0 frame:0
            TX packets:85909 errors:0 dropped:0 overruns:0 carrier:0
            collisions:0 txqueuelen:0
            RX bytes:14238288 (13.5 Mb) TX bytes:14238288 (13.5 Mb)
```

**System Information**  Activate Changes Logout

**Resources** Resources: Disk and Memory usage information

Disk Usage:

Filesystem	Size	Used	Avail	Use%	Mounted on
/dev/hda2	15G	1.1G	14G	8%	/
/dev/hda1	99M	12M	83M	12%	/boot
/dev/shm	126M	0	126M	0%	/dev/shm

Memory Usage:

	total	used	free	shared	buffers	cached
Mem:	256184	175028	81156	0	32884	65204
-/+ buffers/cache:	76940	179244				
Swap:	522104	0	522104			

## 8.18. Asterisk Logs:

Asterisk Log messages  Nov  4  Go

```
[Nov 4 04:40:57] NOTICE[2094] cdr.c: CDR simple logging enabled.
[Nov 4 04:40:57] NOTICE[2094] loader.c: 157 modules will be loaded.
[Nov 4 04:40:58] WARNING[2094] res_config_pgsql.c: Postgresql RealTime: No database socket found, using '/tmp/'.
[Nov 4 04:40:58] ERROR[2094] res_config_pgsql.c: Postgresql RealTime: Failed to connect database server asteri...
[Nov 4 04:40:58] WARNING[2094] res_config_pgsql.c: Postgresql RealTime: Couldn't establish connection. Check d...
[Nov 4 04:40:58] NOTICE[2094] config.c: Registered Config Engine pgsql
[Nov 4 04:41:00] WARNING[2094] res_smidi.c: No SDMI interfaces are available to listen on, not starting SDMI li...
[Nov 4 04:41:00] NOTICE[2094] config.c: Registered Config Engine odbc
[Nov 4 04:41:00] NOTICE[2094] res_odbc.c: Adding ENV var: INFORMIXSERVER=my_special_database
[Nov 4 04:41:00] NOTICE[2094] res_odbc.c: Adding ENV var: INFORMIXDIR=/opt/informix
[Nov 4 04:41:00] NOTICE[2094] res_odbc.c: Connecting asterisk
[Nov 4 04:41:00] WARNING[2094] res_odbc.c: res_odbc: Error SQLConnect=-1 errno=0 [unixODBC][Driver Manager]Dat...
[Nov 4 04:41:00] WARNING[2094] res_odbc.c: Failed to connect to asterisk
[Nov 4 04:41:00] NOTICE[2094] res_odbc.c: Registered ODBC class 'asterisk' dsn->[asterisk]
[Nov 4 04:41:00] NOTICE[2094] res_odbc.c: res_odbc loaded.
[Nov 4 04:41:02] NOTICE[2094] pbx_ael.c: Starting AEL load process.
[Nov 4 04:41:02] NOTICE[2094] pbx_ael.c: AEL load process: calculated config file name '/etc/asterisk/extensio...
[Nov 4 04:41:02] WARNING[2094] ael.y: === File: /etc/asterisk/extensions.ael, Line 112, Cols: 34-34: Warning!
[Nov 4 04:41:02] WARNING[2094] ael.y: === File: /etc/asterisk/extensions.ael, Line 120, Cols: 34-34: Warning!
[Nov 4 04:41:02] WARNING[2094] ael.y: === File: /etc/asterisk/extensions.ael, Line 128, Cols: 33-33: Warning!
[Nov 4 04:41:02] NOTICE[2094] pbx_ael.c: AEL load process: parsed config file name '/etc/asterisk/extensions.a...
[Nov 4 04:41:02] WARNING[2094] pbx_ael.c: Warning: file /etc/asterisk/extensions.ael, line 141-145: The includ...
[Nov 4 04:41:02] WARNING[2094] pbx_ael.c: Warning: file /etc/asterisk/extensions.ael, line 141-145: The includ...
[Nov 4 04:41:02] WARNING[2094] pbx_ael.c: Warning: file /etc/asterisk/extensions.ael, line 141-145: The includ...
[Nov 4 04:41:02] WARNING[2094] pbx_ael.c: Warning: file /etc/asterisk/extensions.ael, line 276-283: The includ...
[Nov 4 04:41:02] NOTICE[2094] pbx_ael.c: AEL load process: checked config file name '/etc/asterisk/extensions...
[Nov 4 04:41:02] NOTICE[2094] pbx_ael.c: AEL load process: compiled config file name '/etc/asterisk/extensions...
[Nov 4 04:41:02] NOTICE[2094] pbx_ael.c: AEL load process: merged config file name '/etc/asterisk/extensions.a...
[Nov 4 04:41:02] WARNING[2094] pbx.c: Context 'ael-local' tries includes nonexistent context 'ael-parkedcalls'
[Nov 4 04:41:02] WARNING[2094] pbx.c: Context 'ael-dundi-e164-local' tries includes nonexistent context 'ael-d...
[Nov 4 04:41:02] WARNING[2094] pbx.c: Context 'ael-dundi-e164-local' tries includes nonexistent context 'ael-d...
[Nov 4 04:41:02] NOTICE[2094] pbx_ael.c: AEL load process: verified config file name '/etc/asterisk/extensions...
[Nov 4 04:41:03] WARNING[2094] frame.c: Cannot disallow unknown format ''
```

## 8.19. File Editor:



## 8.20. Asterisk CLI:

## Asterisk Command Line Interface

Command> **core show version**

Asterisk 1.4.9 built by admin @ aomori on a i686 running Linux on 2007-07-25 21:01:48 UTC

Asterisk CLI>

## Backup:

### Backup / Restore Configurations

[Activate Changes](#) [Logout](#)

List of Previous Configuration Backups

Move the mouse over to a field to see tooltips

*No Previous Backup configurations found !!*

Please click on the 'Take a BackUp' button  
to take a backup of the current system configuration

[Take a Backup](#)

## List of Previous Configuration Backups

Move the mouse over to a field to see tooltips

*No Previous Backup configurations found !!*

Please click on the 'Take a BackUp' button  
to take a backup of the current system configuration

**Create New Backup**

File Name:

(do not enter any extension )

[Take a Backup](#)

**Create New Backup**

File Name:

(do not enter any extension )

## List of Previous Configuration Backups

Move the mouse over to a field to see tooltips

S.No	Name	Date	Options
1	backup09042007	Apr 09, 2007	<a href="#">Restore</a> <a href="#">Delete</a>

[Take a Backup](#)

## Restoration:

Alert !

X

Configuration restored !!

[Ok](#)

## Options:

[Local Extension settings](#)[Change Password](#)[Avanced](#)[Run Setup Wizard](#)

Move the mouse over to a field to see tooltips

## Local Extension Settings:

Local Extensions are 4 digitsFirst Extension Number : 

- Allow analog phones to be assigned to multiple extensions
- Allow extensions to be AlphaNumeric (SIP/IAX users)

## Default Settings for a New User:

- |  |   |
|--|---|
| <input type="checkbox"/> Is Agent                | <input checked="" type="checkbox"/> Voicemail     |
| <input type="checkbox"/> In Directory            | <input type="checkbox"/> CTI                      |
| <input checked="" type="checkbox"/> SIP          | <input checked="" type="checkbox"/> IAX           |
| <input checked="" type="checkbox"/> Call Waiting | <input checked="" type="checkbox"/> 3-Way Calling |
| <input type="text"/> VoiceMail Password          |   |

[Save](#)[Cancel](#)

**Admin Options**

[Activate Changes](#) [Logout](#)

[Local Extension settings](#) [\*\*Change Password\*\*](#) [Advanced](#) [Run Setup Wizard](#)

**Retype New Password:**  [Retype New Password](#)

Enter New Password:

Retype New Password:

[Update](#)

## The Advanced Menu:

**digium|Asterisk**

System Configuration  
[About Digium](#) | [Report a Bug](#) | [Help](#)

[Home](#) **Welcome to the Asterisk Configuration Panel** [Activate Changes](#) [Logout](#)

Asterisk Configuration Panel - Please click on a panel to manage related features

- [Users](#)
- [Conferencing](#)
- [Voicemail](#)
- [Call Queues](#)
- [Service Providers](#)
- [Calling Rules](#)
- [Incoming Calls](#)
- [Voice Menus](#)
- [Record a Menu](#)
- [Music On Hold](#)
- [SIP](#)
- [IAX](#)
- [Jabber](#)
- [Jingle](#)
- [Zap Channel](#)
- [Active Channels](#)
- [System Info](#)
- [Backup](#)
- [Options](#)

**Asterisk™ Configuration Engine**

Username:   
 Password:

✓ Connected!

[Login](#) [Logoff](#)

Move the mouse over to a field to see tooltips

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## Music On Hold:

Music on hold sometimes keeps people less angry while they wait for an answer

**Music on Hold Classes** 

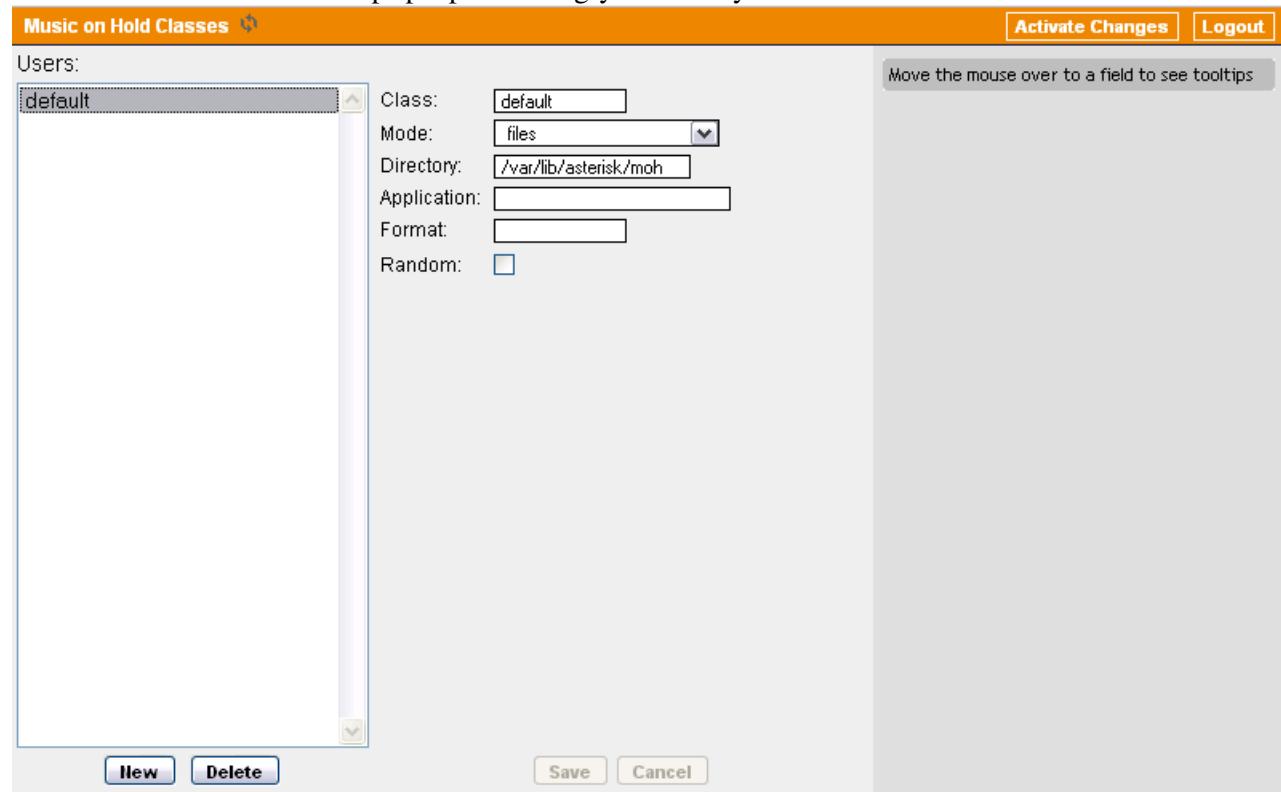
**Activate Changes** **Logout**

Move the mouse over to a field to see tooltips

Users:
default

Class:  Mode:  Directory:  Application:   
Format:  Random:

**New** **Delete** **Save** **Cancel**



SIP:

## SIP (Session Initiation Protocol) Configuration

The screenshot shows the SIP configuration interface. Key settings include:

- Context:** default
- Realm for digest authentication:** [empty field]
- UDP Port to bind to:** 5060
- IP address to bind to:** 0.0.0.0
- Domain:** [empty field]
- Allow guest calls:**
- Overlap dialing support:**
- Allow Transfers:**
- Enable DNS SRV lookups (on outbound calls):**
- Pedantic:**

**Type of Service**

TOS for Signalling packets:	[dropdown menu]
TOS for RTP audio packets:	[dropdown menu]
TOS for RTP video packets:	[dropdown menu]
Max Registration/Subscription Time:	3600
Min Registration/Subscription Time:	60
Default Incoming/Outgoing Registration Time:	[dropdown menu]
Min RoundtripTime (T1 Timeout):	1000

**Buttons:** Save, Cancel

**Context:** Default context for incoming calls

**Realm for digest authentication:** Realm for digest authentication.defaults to 'asterisk'. If you set a system name in asterisk.conf, it defaults to that system name. Realms MUST be globally unique according to RFC 3261. Set this to your host name or domain name

**UDP Port to bind to:** SIP standard port is 5060

**IP address to bind to:** 0.0.0.0 binds to all

**Domain:** Comma separated list of domains which Asterisk is responsible for

**Allow guest calls:** Enable guest calls.

**Overlap dialing support:** Enable dialing support

**Allow Transfers:** Enable Transfers

**Enable DNS SRV lookups (on outbound calls):** Enable DNS SRV lookups on calls

**Pedantic:** Enable slow, pedantic checking of Call-ID:s, multiline SIP headers and URI-encoded headers

### Type of Service

**TOS for Signalling packets:** Sets Type of Service for SIP packets

**TOS for RTP audio packets:** Sets Type of Service for RTP audio packets

**TOS for RTP video packets:** Sets Type of Service for RTP video packets

**Max Registration/Subscription Time:** Maximum duration (in seconds) of incoming registration/subscriptions we allow. Default 3600 seconds.

**Min Registration/Subscription Time:** Minimum duration (in seconds) of registrations/subscriptions. Default 60 seconds

**Default Incoming/Outgoing Registration Time:** Default duration (in seconds) of incoming/outgoing registration

**Min RoundtripTime (T1 Time):** Minimum roundtrip time for messages to monitored hosts, Defaults to 100 ms

**Override Notify MIME Type:** Allow overriding of mime type in MWI NOTIFY

**Time between MWI Checks:** Default Time between Mailbox checks for peers

**Music On Hold Interpret:** This option specifies a preference for which music on hold class this channel should listen to when put on hold if the music class has not been set on the channel with Set(CHANNEL(musicclass)=whatever) in the dialplan, and the peer channel putting this one on hold did not suggest a music class

**Music On Hold Suggest:** This option specifies which music on hold class to suggest to the peer channel when this channel places the peer on hold. It may be specified globally or on a per-user or per-peer basis.

**Language:** Default language setting for all users/peers

**Enable Relaxed DTMF:** Relax dtmf handling

**RTP TimeOut:** Terminate call if 60 seconds of no RTP activity when we're not on hold

**RTP HoldTimeOut:** Terminate call if 300 seconds of no RTP activity when we're on hold (must be > rtptimeout)

**Trust Remote Party ID:** If Remote-Party-ID should be trusted

**Send Remote Party ID:** If Remote-Party-ID should be sent

**Generate In-Band Ringing:** If we should generate in-band ringing always use 'never' to never use in-band signalling, even in cases where some buggy devices might not render it. Default: never

**Server UserAgent:** Allows you to change the user agent string

**Allow Nonlocal Redirect:** If checked, allows 302 or REDIR to non-local SIP address Note that promiscredir when redirects are made to the local system will cause loops since Asterisk is incapable of performing a 'hairpin' call

**Add 'user=phone' to URI:** If checked, 'user=phone' is added to uri that contains a valid phone number

**DTMF Mode:** Set default dtmfmode for sending DTMF. Default: rfc2833H

**Send Compact SIP Headers:** send compact sip headers

## SIP Video Related

**Max Bitrate (kb/s):** Maximum bitrate for video calls (default 384 kb/s)

**Support for SIP Video:** Turn on support for SIP video

**Generate Manager Events:** Generate manager events when sip ua performs events (e.g. hold)

**Reject NonMatching Invites:** When an incoming INVITE or REGISTER is to be rejected, for any reason, always reject with '401 Unauthorized' instead of letting the requester know whether there was a matching user or peer for their request

**NonStandard G.726 Support:** If the peer negotiates G726-32 audio, use AAL2 packing order instead of RFC3551 packing order (this is required for Sipura and Grandstream ATAs, among others). This is contrary to the RFC3551 specification, the peer should be negotiating AAL2-G726-32 instead

## T.38 FAX Passthrough Support

**T.38 fax (UDPTL) Passthrough:** Enables T.38 fax (UDPTL) passthrough on SIP to SIP calls

## Sip Debugging

**Enable SIP debugging:** Turn on SIP debugging by default

**Record SIP History:** Record SIP history by default

**Notify on Ringing:** Notify subscriptions on RINGING state

## Outbound SIP Registrations

**Register:** Register as a SIP user agent to a SIP proxy (provider)

**Register TimeOut:** Retry registration calls at every 'x' seconds (default 20)

**Register Attempts:** Number of registration attempts before we give up; 0 = continue forever

## NAT Support

**Extern ip:** Address that we're going to put in outbound SIP messages if we're behind a NAT

**Extern Host:** Alternatively you can specify an external host, and Asterisk will perform DNS queries periodically. Not recommended for production environments! Use externip instead

**Extern Refresh:** How often to refresh externhost if used. You may specify a local network in the field below

**Local Network Address:** '192.168.0.0/255.255.0.0' : All RFC 1918 addresses are local networks,

'10.0.0.0/255.0.0.0' : Also RFC1918, '172.16.0.0/12' : Another RFC1918 with CIDR notation,

'169.254.0.0/255.255.0.0' : Zero conf local network

**NAT mode:** Global NAT settings (Affects all peers and users); yes = Always ignore info and assume NAT; no = Use NAT mode only according to RFC3581; never = Never attempt NAT mode or RFC3581 support; route = Assume NAT, don't send rport

**Allow RTP Reinvite:** Asterisk by default tries to redirect the RTP media stream (audio) to go directly from the caller to the callee. Some devices do not support this (especially if one of them is behind a NAT).

## IAX:

**IAX (Inter Asterisk Exchange Protocol) Configuration**

Bind Port:   
Bind Address:   
IAX1 Compatibility:   
No Checksums:   
Delay Reject:   
ADS:

**Call Detail Records**

AMA Flags:   
Accountcode:   
Music On Hold Interpret:   
Music On Hold Suggest:   
Language:  en  
Bandwidth:  low

**Jitter Buffer**

Enable Jitter Buffer:   
Force Jitter Buffer:   
Drop Count:   
Max Jitter Buffer:   
Max Interpolation Frames:

## Jabber:

**Jabber™ Users Configuration**

**Activate Changes** **Logout**

Users:

Extension:	<input type="text"/>
Username:	<input type="text"/>
Password:	<input type="password"/>
Server Host:	<input type="text"/>
Buddy:	<input type="text"/>
Status Message:	<input type="text"/>
Type:	<input type="text"/> client
Port:	<input type="text"/>
TimeOut:	<input type="text"/>
Use TLS:	<input type="checkbox"/>
Use SASL:	<input type="checkbox"/>

Move the mouse over to a field to see tooltips

Jabber™ is a collection of open, XML-based protocols for instant messaging and presence information.  
Jabber® is a registered trademark of Jabber, Inc.

**New** **Delete** **Save** **Cancel**

## Jingle:

Jingle Configuration

Activate Changes Logout

Users:

Extension:	<input type="text"/>
Username:	<input type="text"/>
Disallowable Codecs:	<input type="text"/>
Allowed Codecs:	<input type="text"/> Edit
Context:	<input type="text"/>
Connection:	<input type="text"/>

New Delete Save Cancel

Move the mouse over to a field to see tooltips

This screenshot shows the 'Jingle Configuration' page. At the top, there are 'Activate Changes' and 'Logout' buttons. Below that, a section titled 'Users:' contains a table with six rows: Extension, Username, Disallowable Codecs, Allowed Codecs, Context, and Connection. Each row has an input field and a tooltip message 'Move the mouse over to a field to see tooltips'. At the bottom of the table are 'New', 'Delete', 'Save', and 'Cancel' buttons.

## Zap Channels:

Configure the Zap channel

Activate Changes   Logout

Move the mouse over to a field to see tooltips

language:	en
Context:	default
switchtype:	National ISDN 2
nsf:	
pridialplan:	
prilocaldialplan:	
internationalprefix:	
nationalprefix:	
localprefix:	
privateprefix:	
unknownprefix:	
resetinterval:	
overlapdial:	<input type="checkbox"/>
priindication:	

Save   Cancel

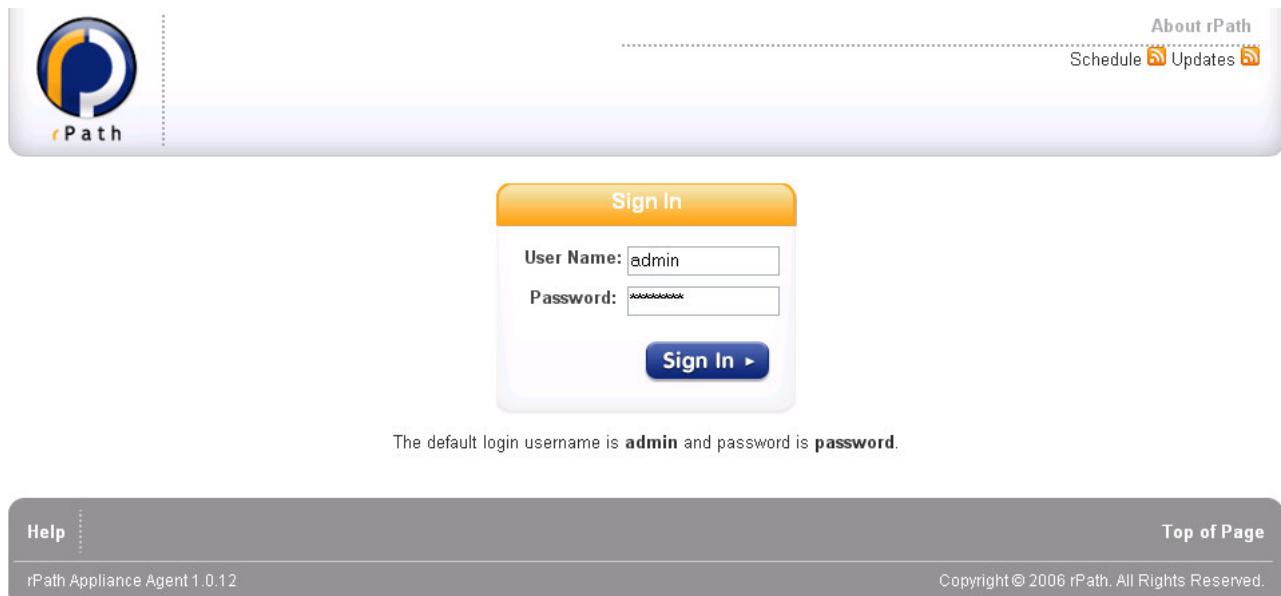
## 9. System Setup & Administration:



To enter into the system administration menu. Click on the System configuration link at the top right hand corner of the webpage.



You will then get another popup screen as shown below.



The default login username is **admin** and password is **password**.

The login use the default username of **admin** and the password being **password**



About rPath  
Welcome admin. Logout  
Schedule Updates

- Change Password
- Configure Email
- Configure Notification
- Configure Networking
- Backup and Restore

## rPath Appliance Agent Configuration Wizard

Add or update configuration information in the following pages. Click OK to save information for the page, and advance to the next configuration page. If the information does not require updating, click OK to continue on to the next page.

**OK**

Help

[Top of Page](#)

rPath Appliance Agent 1.0.12

Copyright © 2006 rPath. All Rights Reserved.

## **9.1. Change Password:**

**Change Password**

To change the rPath Appliance Agent administrator password, enter the current password, a new password, and confirmation of the new password into the appropriate text input fields below.

Click **Change Password** to complete the password change.

Current Password

New Password

Password Again

**Learn More** **Change Password**

## **9.2. Email Configuration:**

**Email Configuration**

Configure email relay server and mail originator address by entering the hostname for a valid email server through which the rPath Appliance Agent may relay email, and an email address in the appropriate text input fields below.

Click OK to save the configuration.

**Mail Relay**

**Mail From**

**Learn More** **OK**



- Enter your email server address.
- Enter the email address which you want to use as the senders address.

**Email Notification Addresses**

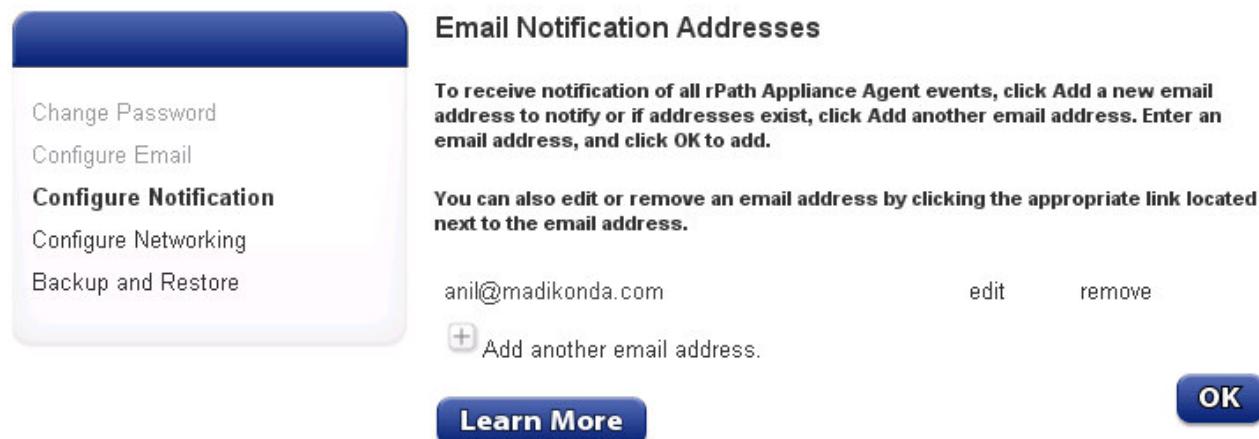
To receive notification of all rPath Appliance Agent events, click Add a new email address to notify or if addresses exist, click Add another email address. Enter an email address, and click OK to add.

You can also edit or remove an email address by clicking the appropriate link located next to the email address.

anil@madikonda.com [edit](#) [remove](#)

**Add another email address.**

**Learn More** **OK**



- Here you need to enter at least one email address to which all system notifications will be sent to. This should be the email of your system administrator.

### **9.3. Configure Networking:**

In this section you can change the network configuration of the system. You can change the IP address and the Hostname of the machine.

Generally if you intend to use it in a Home environment you can leave it to use the dhcp method. If you need to setup a static Ip address then be sure to enter all the required fields. Then don't forget to reboot.

**Configure Networking**

Configure network host name, domain name service, and network interface settings, such as IP address and default gateway using the controls below:

Click OK to update your changes and Restart to apply your changes to the interface.

**Host Name** asterisk.madikonda.con

**DNS Search Domain**

**DNS Server(s)** 192.168.1.1 edit remove  
+ Add another DNS server.

**Cancel** **OK**

**eth0** 00:0C:29:AF:A7:75

DHCP  Configure manually

**IP Address** 192.168.1.10

**Netmask** 255.255.255.0

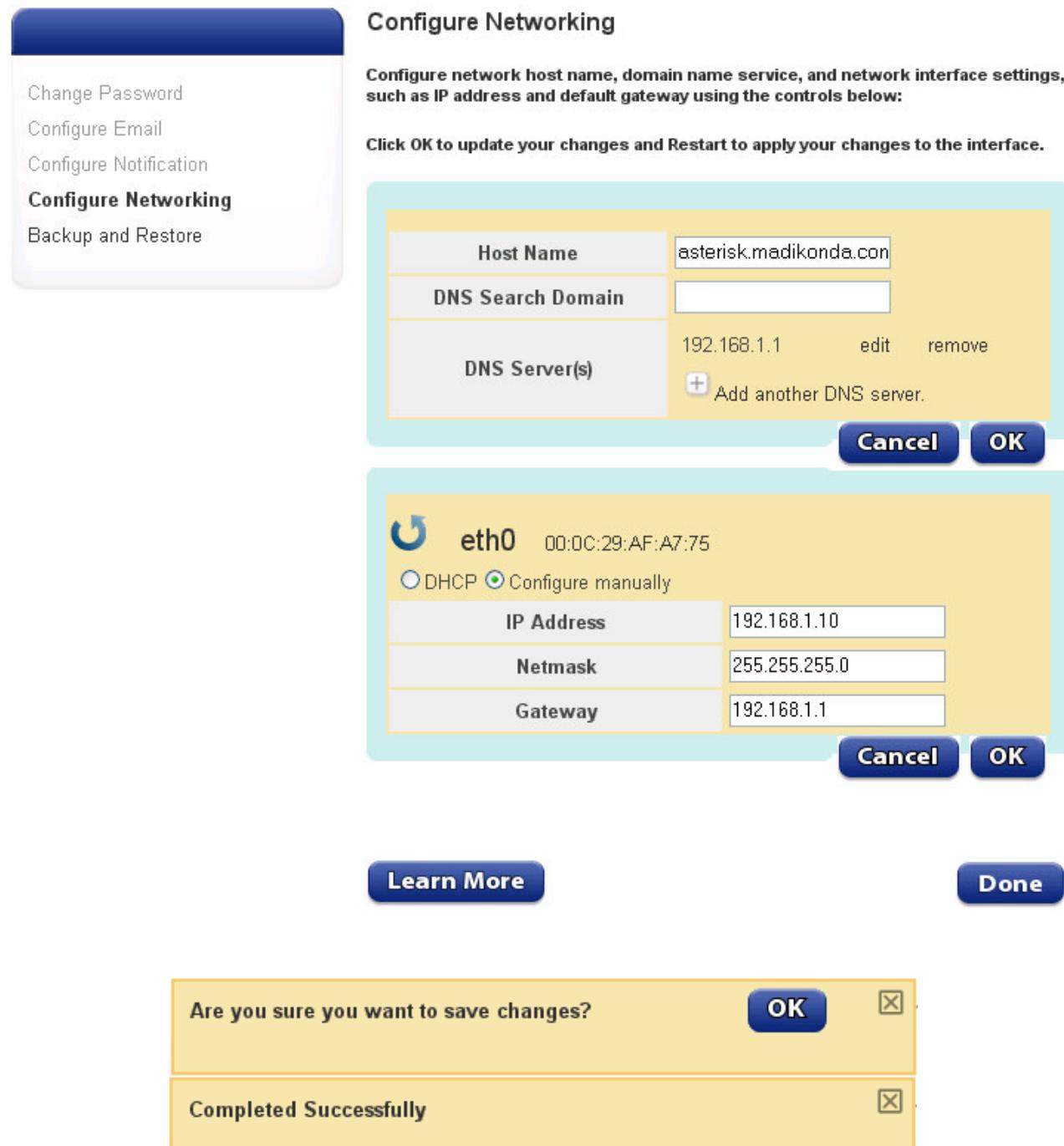
**Gateway** 192.168.1.1

**Cancel** **OK**

**Learn More** **Done**

Are you sure you want to save changes? **OK**

Completed Successfully



You can also configure the networking manually, by logging on as root using the system console.

Backup:

The screenshot shows the 'Backup Configuration' tab selected in the top navigation bar. On the left, a sidebar lists 'Change Password', 'Configure Email', 'Configure Notification', 'Configure Networking', and 'Backup and Restore'. The 'Backup and Restore' option is highlighted. The main area is titled 'Configure Backups' with the sub-instruction 'Configure backup schedule, location and the number of backups to keep.' Below this is a 'Backup Settings' section. It includes an 'Enable Backup Schedule' toggle set to 'Yes' (radio button selected), a 'Backup Schedule' dropdown set to 'Weekly' with days of the week listed (Mon is selected), and a time dropdown set to '11 PM'. A 'Number of Backups' input field contains the value '3'. A note below it says 'How many backups should be stored at the location below.' The 'Backup Location' section follows, with 'Backup Type' set to 'NFS File Share', a 'Disk Label' input field, 'Connection Host' and 'Connection Path' input fields, and notes about remote shares and mountpoints. Below these are 'Connection User Name' and 'Connection Password' input fields. A large blue 'Save' button is at the bottom right. A progress bar at the bottom indicates 'Performing backup...' and 'Please wait ...'.

Change Password  
Configure Email  
Configure Notification  
Configure Networking  
**Backup and Restore**

**Backups** **Backup Configuration**

**Configure Backups**

Configure backup schedule, location and the number of backups to keep.

**Backup Settings**

Enable Backup Schedule  Yes  No

Backup Schedule    
 Mon  Tue  Wed  Thu  Fri  Sat  Sun

At specific time:

Number of Backups   
*How many backups should be stored at the location below.*

**Backup Location**

Backup Type

Disk Label

Connection Host

Connection Path   
*For remote shares, this is the full path on the share where the backup should be stored. For Label mounts, this is the path relative to the mountpoint where the backups are to be stored.*

Connection User Name

Connection Password

**Save**

Performing backup...  
Please wait ...

## rPath Appliance Agent

[Backup and Restore](#)  
[Change Password](#)  
[Configuration](#)  
[Manage Services](#)  
[Schedule Reboot](#)  
[System Updates](#)  
[View Conary Log](#)  
[View Log](#)

System information:

- Total memory: 250 MB
- Memory free: 15 MB
- Total swap: 258 MB
- Swap free: 258 MB
- 1 CPU(s):
  - ■ CPU model: Intel(R) Pentium(R) M processor 1.70GHz
  - CPU frequency: 598.261
- Uptime: 0 days, 2 hours, 7 minutes

Please wait while the installed software list is being retrieved ...

## **9.4. System Information:**

**rPath Appliance Agent**

Backup and Restore  
Change Password  
Configuration  
Manage Services  
Schedule Reboot  
System Updates  
View Conary Log  
View Log

System information:

- Total memory: 250 MB
- Memory free: 15 MB
- Total swap: 258 MB
- Swap free: 258 MB
- 1 CPU(s):
  - CPU model: Intel(R) Pentium(R) M processor 1.70GHz
  - CPU frequency: 598.261
- Uptime: 0 days, 2 hours, 7 minutes

Please wait while the installed software list is being retrieved ...

## **9.5. System Updates:**

The System update panel gives you access to updating your entire system, including the latest releases of asterisk software. You can schedule a check.

When the system finds any updates, it will email you the details about the updates. Then you can use this panel again to install the updates.

This screenshot shows the 'System Updates' panel. On the left, a sidebar lists various system management options: Backup and Restore, Change Password, Configuration, Manage Services, Schedule Reboot, System Updates (which is bolded), View Conary Log, and View Log. At the top right, there are two tabs: 'System Updates' (highlighted in orange) and 'Preferences'. Below the tabs, a message reads: 'To check for available system updates, click Check Now. If updates are found to be available, they may be examined by expanding System Update and applied by clicking Apply Now.' A link 'Click Preferences for recurring updates and update preferences.' is also present. At the bottom are two buttons: 'Learn More' and 'Check Now'.

The following shows the update function searching for updates.

This screenshot shows the 'System Updates' panel during a search. The sidebar and tabs are identical to the previous screenshot. In the main area, a message box contains the text 'Looking for available updates ...' followed by 'Please wait ...'. There is a small decorative icon of a speaker with sound waves to the right of the message box. At the bottom are 'Learn More' and 'Check Now' buttons.

**System Updates**   **Preferences**

Backup and Restore  
Change Password  
Configuration  
Manage Services  
Schedule Reboot  
**System Updates**  
View Conary Log  
View Log

To check for available system updates, click Check Now. If updates are found to be available, they may be examined by expanding System Update and applied by clicking Apply Now.

Click Preferences for recurring updates and update preferences.

**Looking for available updates ...**

Scheduled

**Learn More**   **OK**

## Schedule your updates:

Using this feature you can schedule your server to retrieve and apply update them automatically.

**System Updates**   **Preferences**

Backup and Restore  
Change Password  
Configuration  
Manage Services  
Schedule Reboot  
**System Updates**  
View Conary Log  
View Log

**System Updates Preferences**

System Updates checks for updated versions of your software based on information about your system and current software. Use the options below to specify when you wish the update checks to be performed and whether updates should be automatically installed.

NOTE: Update checks are performed between the hour selected and the following hour. the precise time an update check will occur will be shown in a message dialog when the schedule is saved.

**Check for updates**    
 Mon  Tue  Wed  Thu  Fri  Sat  Sun  
At specific time:

**Automatically install updates**  
If you do not select this option, you will be notified when updates are ready to be installed.

**Save**

No check has finished yet. 

Next scheduled check at: 04/16/2007 04:16 AM

**Learn More**

## **9.6. Conary Configuration:**

You do not need to update the configuration here. But if you want to check for updates from a different group and install, then you can change and update the configuration.

The screenshot shows a user interface titled "Conary Configuration". On the left, a sidebar lists links: Backup and Restore, Change Password, Configuration (which is highlighted), Manage Services, Schedule Reboot, System Updates, View Conary Log, and View Log. The main panel has a title "Enter Conary configuration details, such as the Install Label Path into the appropriate text input fields below." Below this is a note "Click OK to save the configuration." A text input field contains the value "asterisk.rpath.org@pk:1-4 asterisk.rpath.org@pk:devel cc". At the bottom right are two buttons: "Learn More" and "OK".

## **9.7. Time Zone Configuration:**

Set and update your system times using this control panel. Alternatively you can also tick the synchronize tick box and set the server to update the time automatically.

The screenshot shows a user interface titled "Current system information:" with the date "Mon, 09 Apr 2007 18:36:43 EST" displayed. On the left, a sidebar lists links: Backup and Restore, Change Password, Configuration, Configure Conary, Configure Email, Configure Networking, Configure Notification, Configure Proxy, Root Password, Time Zone and Time (which is highlighted), Upload SSL Certificate, Manage Services, Schedule Reboot, System Updates, View Conary Log, and View Log. The main panel has a title "Set the time zone and time". It includes a "Timezone:" dropdown set to "Australia/Sydney", a "Date and Time:" section with dropdowns for Hour (18), Minute (36), Second (44), Month (04), Day (09), and Year (2007), and a checkbox "Synchronize clock with internet servers" which is unchecked. Below this is an "NTP Server:" field containing the value "0.rpath.pool.ntp.org,1.rpath.pool.ntp.org,2.rpath.pool.ntp.or". At the bottom right are two buttons: "Learn More" and "Save".

## 9.8. Upload SSL Certificate:

**Upload SSL Certificate**

To replace the SSL certificate used by rAA, use the file upload dialog below to upload a new certificate.

.PEM File  [Browse...](#) **OK**

**Current Certificate**

Issued To	
Country	AU
Common Name	localhost.localdomain
Organization	Internet Widgits Pty Ltd
State	Some-State
Email Address	root@localhost.localdomain

Issued By	
Country	AU
Common Name	localhost.localdomain
Organization	Internet Widgits Pty Ltd
State	Some-State
Email Address	root@localhost.localdomain

Validity	
Issued On	Apr 9 12:26:09 2007 GMT
Expires On	Apr 8 12:26:09 2008 GMT

SHA1 Fingerprint	
36:63:0D:48:95:4F:88:3D:6B:4C:13:33:6B:FE:6E:5F:6C:53:85:AE	

**Learn More**

If you want to use your own certificate then upload the certificate here and then reboot the server. From next login you will see the new certificate being used.

## 9.9. Services:

Manage any services that you want to use from this control panel.

Services to start and stop.				
	Service	Current	Actions	
Backup and Restore	asterisk	<input checked="" type="checkbox"/>		 
Change Password	crond	<input checked="" type="checkbox"/>		 
Configuration	gpm	<input checked="" type="checkbox"/>		 
<b>Manage Services</b>	gui-lighttpd	<input checked="" type="checkbox"/>		 
Schedule Reboot	ip6tables	<input checked="" type="checkbox"/>		 
System Updates	iptables	<input checked="" type="checkbox"/>		 
View Conary Log	keytable	<input checked="" type="checkbox"/>		 
View Log	kudzu	<input checked="" type="checkbox"/>		 
	lighttpd	<input type="checkbox"/>		 
	mysqld	<input checked="" type="checkbox"/>		 
	netfs	<input checked="" type="checkbox"/>		 
	network	<input checked="" type="checkbox"/>		 
	ntpd	<input checked="" type="checkbox"/>		 
	postfix	<input checked="" type="checkbox"/>		 
	raa	<input checked="" type="checkbox"/>		 
	raa-lighttpd	<input checked="" type="checkbox"/>		 
	raa-restore	<input checked="" type="checkbox"/>		 
	sshd	<input checked="" type="checkbox"/>		 
	syslog	<input checked="" type="checkbox"/>		 
	vsftpd	<input checked="" type="checkbox"/>		 

## 9.10. Scheduled Reboot:

Reboot system

Click Schedule to access a calendar and time selector for choosing a date and time for scheduled reboot of the system. Click Cancel to cancel scheduling a reboot or click Remove to remove a previously scheduled reboot. To save and enable a scheduled reboot, click Save.

You can initiate a reboot of rPath Appliance Agent or shut it down by clicking Reboot or Shutdown.

**Shutdown** **Reboot** **Schedule**

**Learn More**

## 9.11. Conary Log:

Conary Log

Log information detailing Conary actions is available in the text area below or you can retrieve the raw text in a full-screen browser window by clicking Download.

```
[2007 Apr 09 22:28:23]    updated
asterisk-gui=/asterisk.rpath.org@pk:devel/0.9.6-5-1[~!asterisk-gui.livecd is:
x86]--/asterisk.rpath.org@pk:devel/0.9.8-4-1[~!asterisk-gui.livecd is: x86]
[2007 Apr 09 22:28:23]    updated
asterisk-gui:runtime=/asterisk.rpath.org@pk:devel/0.9.6-5-1[~!asterisk-gui.livecd is:
x86]--/asterisk.rpath.org@pk:devel/0.9.8-4-1[~!asterisk-gui.livecd is: x86]
[2007 Apr 09 22:28:23]    updated
asterisk-gui:doc=/asterisk.rpath.org@pk:devel/0.9.6-5-1[~!asterisk-gui.livecd is:
x86]--/asterisk.rpath.org@pk:devel/0.9.8-4-1[~!asterisk-gui.livecd is: x86]
[2007 Apr 09 22:28:26]    installed
libusb=/conary.rpath.com@rpl:devel//1/0.1.10a-3-0.1[~!builddocs is: x86]
[2007 Apr 09 22:28:26]    installed
libusb:lib=/conary.rpath.com@rpl:devel//1/0.1.10a-3-0.1[~!builddocs is: x86]
[2007 Apr 09 22:28:28]    installed
mISDN:lib=/asterisk.rpath.org@pk:devel/1.1.1-1-2[~!dom0,~!domU,~!mISDN.debug,~!mISDN.r
is: x86]
[2007 Apr 09 22:28:28]    installed
mISDN=/asterisk.rpath.org@pk:devel/1.1.1-1-2[~!dom0,~!domU,~!mISDN.debug,~!mISDN.numa,
is: x86]
```

**Learn More** **Download**

## 9.12. View Log:

Backup and Restore  
Change Password  
Configuration  
Manage Services  
Schedule Reboot  
System Updates  
View Conary Log  
**View Log**

### Log Information

Detailed logging information consisting of messages, their source, and logging level are displayed sorted by date below. Click the Download button to retrieve the raw tab-delimited text.

Date/Time	Message	Plugin	Level
Tue, 10 Apr 2007 00:22:15 EST	User 'admin' logged in from host '192.168.1.9'.	System	Information
Tue, 10 Apr 2007 00:29:51 EST	Completed Successfully	Configure Networking	Notice
Tue, 10 Apr 2007 00:31:57 EST	Failed: (Unknown Exception Type) Shell command "/bin/mount -t nfs -orw : /tmp/raa-backup-mount2WSzzYdir" exited with exit code 32	Backup and Restore	Error
Tue, 10 Apr 2007 00:54:45 EST	User 'admin' logged in from host '192.168.1.9'.	System	Information
Tue, 10 Apr 2007 00:56:34 EST	Notification email for type 'unknown' failed to send to 'anil@madikonda.com' because 'Connection unexpectedly closed'.	Backup and Restore	Debug
Tue, 10 Apr 2007 00:56:34 EST	Failed: A permanent failure has occurred: Shell command "/bin/mount -t nfs -orw : /tmp/raa-backup-mountgQbwrrdir" exited with exit code 32	Backup and Restore	Error
Tue, 10 Apr 2007 00:58:56 EST	Setting the timezone to 'Australia/Sydney', Message: "	Time Zone and Time	Information
Tue, 10 Apr 2007 00:58:56 EST	Adding timezone 'Australia/Sydney' to sysconfig/clock, Message: 'Completed'	Time Zone and Time	Information
Tue, 10 Apr 2007 00:58:57 EST	Failed: (Unknown Exception Type) Error running '/bin/date: invalid date '0000000000''	Time Zone and Time	Error
Tue, 10 Apr 2007 00:59:22 EST	Setting the timezone to 'Australia/Sydney', Message: "	Time Zone and Time	Information
Tue, 10 Apr 2007 00:59:23 EST	Adding timezone 'Australia/Sydney' to sysconfig/clock, Message: 'Completed'	Time Zone and Time	Information

[Learn More](#)

[Download](#)

## **10. Installing Other Programs:**

### ***10.1. Installing Mysql:***

Mysql version 5x is installed automatically as part of the AsteriskNow installation and can be maintained using the conary package maintenance system.

**Information below applies to only versions below AsteriskNow beta 4.**

To install mysql server on your AsteriskNow machine use the following command

```
Conary update info-mysql  
Conary update mysql-server
```

This will install Mysql Server version 5 from the asterisk group.

Then pin the software down so that the server is not un-installed when you update using AsteriskNOW update facility.

```
Conary pin info-mysql  
Conary pin mysql-server
```

### ***10.2. Install samba using conary:***

More conary information is available at:

<http://wiki.rpath.com/wiki/Conary:QuickReference>

To install samba you can use the following (as root) on the command line:

**Code:**

```
conary update samba=conary.rpath.com@rpl:1 --resolve
```

You can then start samba by executing the following command:

**Code:**

```
/etc/init.d/smb start
```

To install other packages, you need to search for them in rbuilder. It maybe as simple as replacing 'samba' with whatever package you are wanting to install, sometimes it's not

## **11. System Commands:**

- 1) "sudo su poweroff" to reboot from a ssh session.
- 2) "conary config --show-passwords" will show you the passwords (does not appear to work with this distribution)

## **12. To Get Root Access on Console:**

To get root access from the AsteriskNOW console Menu. Select the second option "console", then press Alt-F9 to get to the \*CLI> prompt. From there type **!** and hit enter.

You then have root access.

## 13. Advanced Configuration & User Tips:

### **To Install Asterisk + Gui on a fresh Operating System:**

Well, you can checkout asterisk 1.4 and the asterisk gui.

enter this command as typed, as root.

```
cd /usr/src ; for i in zaptel libpri asterisk asterisk-gui ; do mkdir $i ; svn co http://svn.digium.com/svn/\$i/trunk $i ; cd $i ; sh configure && make && make install && make samples; cd .. ; done ; clear ; echo "Installation Complete."
```

### **To allow for root login via ssh:**

The default is for the root account to be locked and root via sshd disabled.

To allow for root login via ssh you need to first login via ssh using the admin username. Windows users use putty and winscp to login.

Then enter "sudo su"

Then enter password and then press the enter key.

You are now logged in as root

Now enter the command "passwd root" to change the password for the root account.

Then enter the "new password" and press enter. System will prompt you to reenter the new password to confirm and then will change the password.

Next type "vi /etc/ssh/sshd\_config" and press enter.

In the text displayed use the down arrow to scroll to the line that reads "PermitRootLogin no".

Change the line to "PermitRootLogin yes"

Then go to the end of the file by using the down arrow key.

On an empty line type ":wq" and then press enter to quit vi editor and save the changes.

Then restart the sshd service or restart the server by rebooting it.

### **Asterisk addons:**

for now you have (at beta 3 stage)

<http://www.rpath.com/rbuilder/search?search=asterisk+addons&type=Packages>

which shows 2 people having it packaged, you can use this experimentally, and post your results, for I am not 100% sure it will work.

The command will look SOMETHING like

**Code:**

```
conary update asterisk-addons=starkey.rpath.org@rpl:devel/1.2.4-3-1
```

Do not quote me, not sure if its 100%, but try that :]

### ***Mysql Setup:***

At command line enter the below to login to mysql:  
mysql -u root -p

To enable access from remote machines use this command:

```
GRANT ALL PRIVILEGES ON *.* TO 'someuser'@'%' IDENTIFIED BY 'somepass';  
FLUSH PRIVILEGES;
```

## ***Updating Providers.conf:***

I've gotten my SipPhone account working, with DTMF, etc. I'd like to know what other providers are working in AsteriskNOW, and what you had to do to make it work. If you wouldn't mind, please post how you configured AsteriskNOW for your service provider, and I'll compile responses and try and create a HowTO somewhere.

Here's how I got SipPhone to work:

```
$ sudo vi /etc/asterisk/providers.conf
```

```
[SipPhone]
providername = SipPhone
provider = proxy01.sipphone.com
hassip = yes
hasiax = no
registeriax = no
registersip = yes
host = proxy01.sipphone.com
insecure = very
fromdomain = proxy01.sipphone.com

[voipcheap]
providername = VoipCheap
provider = sip.voipcheap.com
hassip = yes
hasiax = no
registeriax = no
registersip = yes
host = sip.voipcheap.com
insecure = invite
fromdomain = sip.voipcheap.com
```

Save the changes, and restart AsteriskNow. The New Providers should now show up as a service provider. Go ahead and add SipPhone as a service provider using the Asterisk GUI. Next, you'll need to fix users.conf. do the following.

```
$ sudo vi /etc/asterisk/users.conf
```

Find the trunk section for SipPhone, and add these lines to it:

```
username = 1747XXXXXXX
callerid = XXX-XXX-XXXX
fromuser = 1747XXXXXXX
authname=1747XXXXXXX
```

Where the X's are values that match your account with SipPhone, and your callerid.

## **Re-generate the GUI Certificate:**

If you want to use the VMware appliance of AsteriskNow and you change the default (localhost.localdomain) info to real names the GUI does not work anymore. This is probably due to the mismatch in the ssh certificate that is automatically build during the first startup!

- 1) The first time you start the appliance, press <space> at the boot prompt. Then add the option '**single**' to the boot command en press <enter>. Linux will now boot in single user mode.
- 2) Edit **/etc/hosts** with the correct host & domain names and static IP-address if you want that. For a static IP address also do 3 else proceed to 4 😊.
- 3) Edit **/etc/sysconfig/network-scripts/ifcfg-eth0** as follows (use your own IP-addresses 😊):  
**Code:**

```
BOOTPROTO=static
IPADDR=192.168.1.123
BROADCAST=192.168.1.255
NETMASK=255.255.255.0
GATEWAY=192.168.1.001
```

- 4) Don't know if this is necessary but I did it just to be sure:

type command '**netstat <hostname you want>**'

- 5) type command '**reboot**'

Now, during the first full boot the ssh certificates will be made with the correct host.domain names and the AsteriskNow GUI will work as a charm!

Success,

Willem

## 14. Client Connections:

Diax:

Diax is an Iax2 soft phone and easy to configure and use. Diax is available from <http://www.laser.com/dante/diax/diax.html>



Click on Line 1 to open up the registration settings screen as below.

**Alias:** Enter the Name of the server.

**Server:** Enter the IP address of your AsteriskNow Server or the Host address.

**Username:** Enter the username here.

**Password:** Enter your AsteriskNow User password & reconfirm in the second password field.

**Context:** It is best left blank, unless you are an advanced AsteriskNow user.

**Register:** Tick this box to register your soft phone with the AsteriskNow server.

Name: Enter your Name

Number: Enter your username again. This needs to be in a Numeric format.

Then click on the save button to save your changes. Once you conform the changes then DIAx will re-start and register with the AsteriskNow server.

When Diax Registers with an AsteriskNow server the Line number colour will change to a Green Coloured button. This indicates that Diax has registered successfully. It is Red, then it shows that Diax has not registered with the Server correctly.



You can now dial your local and other destination numbers.

## ***Vi Commands:***

```
To use vi: vi filename  
To exit vi and save changes: ZZ or :wq  
To exit vi without saving changes: :q!  
To enter vi command mode: [esc]
```