User Documentation
For version beta6

*** Work in progress ***

Last updated: November 4, 2007

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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.

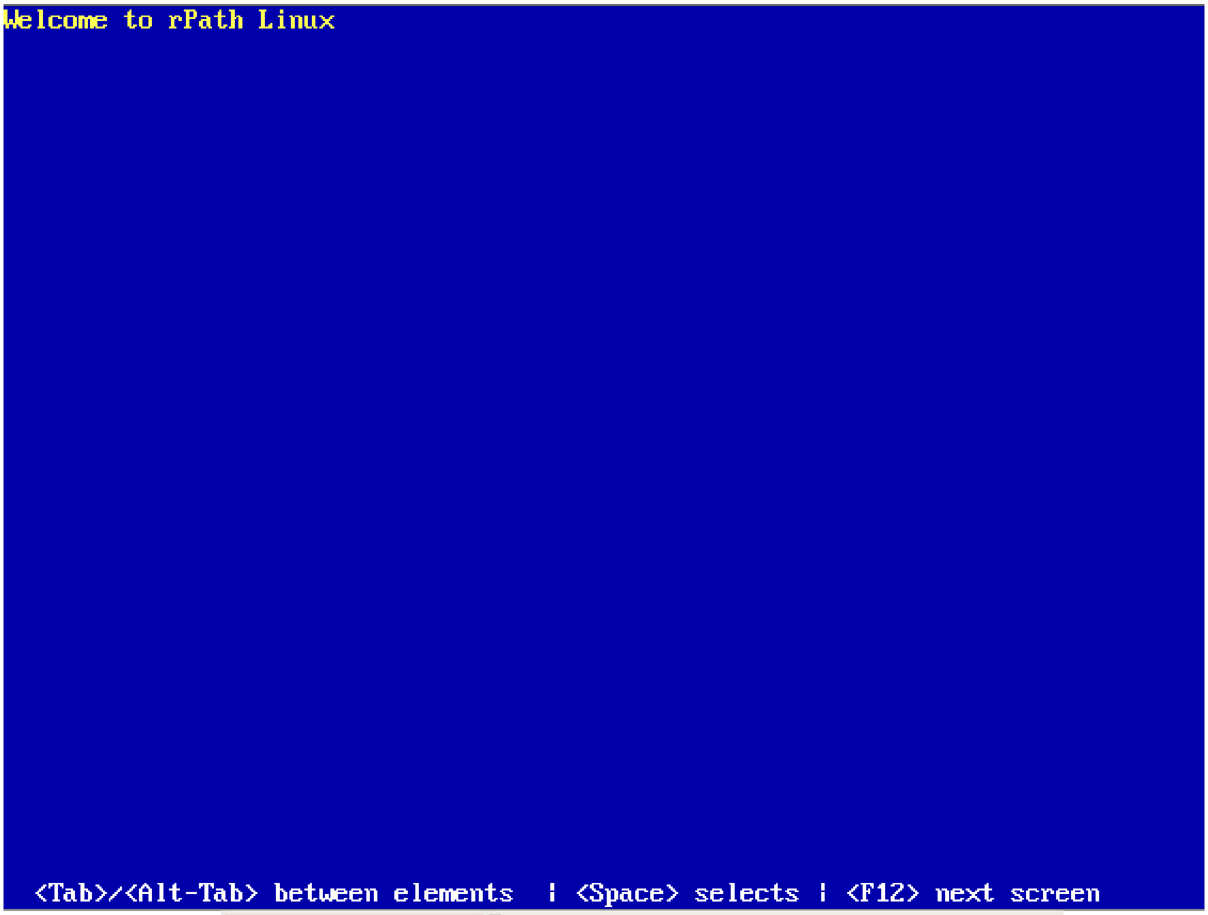
5. Pre-requisites:

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/

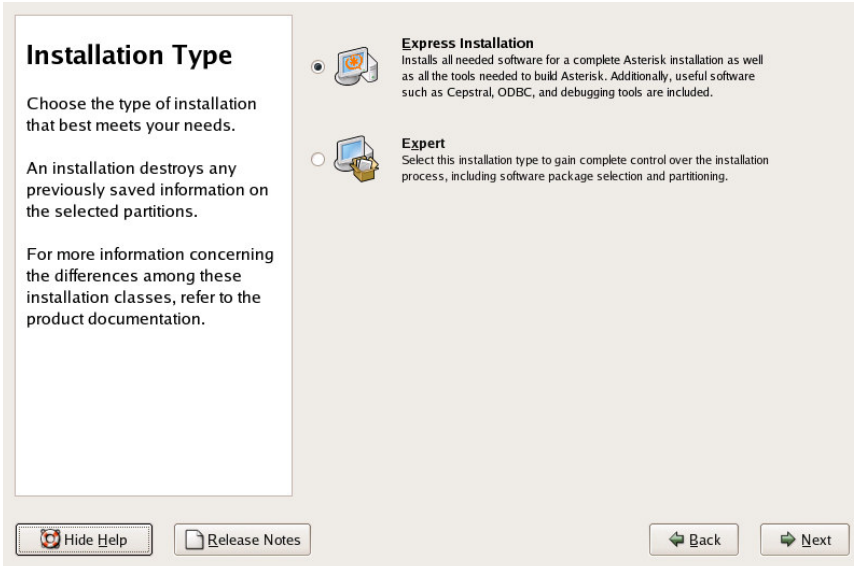
6. AsteriskNOW Installation:

- 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.

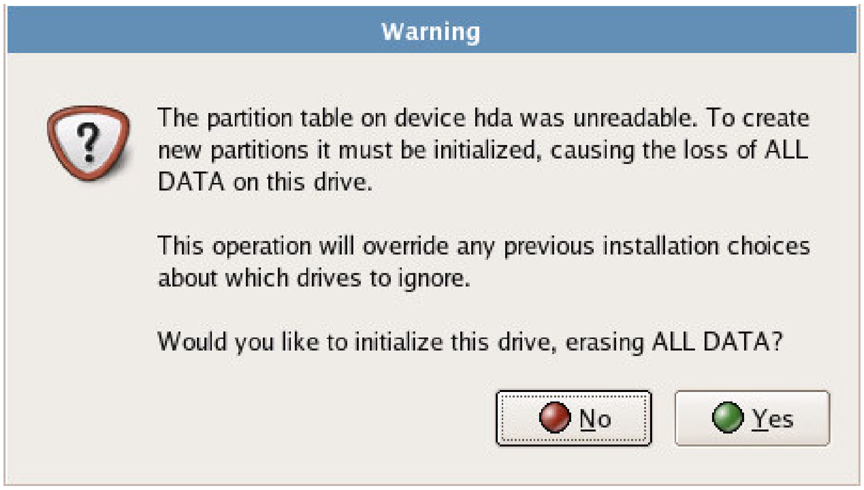


- Click next to continue



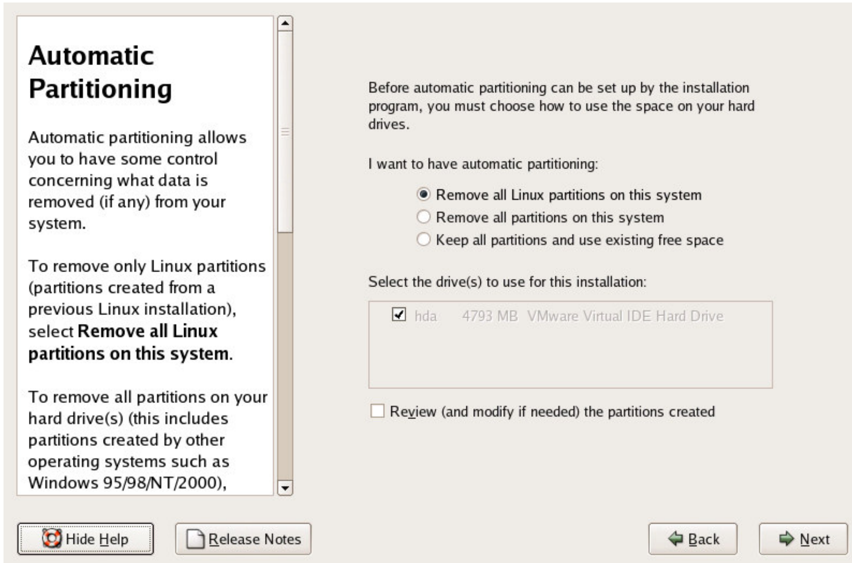
- Choose Express installation and click next.

- 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 loose all of your data you have on this hard drive if you proceed further by pressing the yes button.

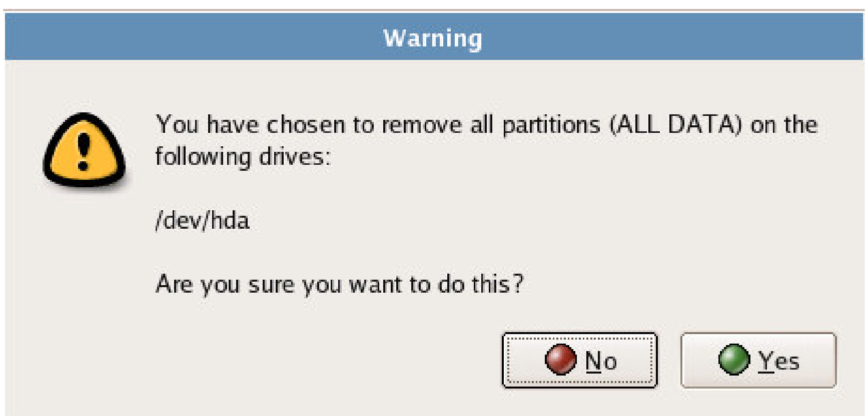


- Click on Yes to continue the setup.

6.1. Partitioning:

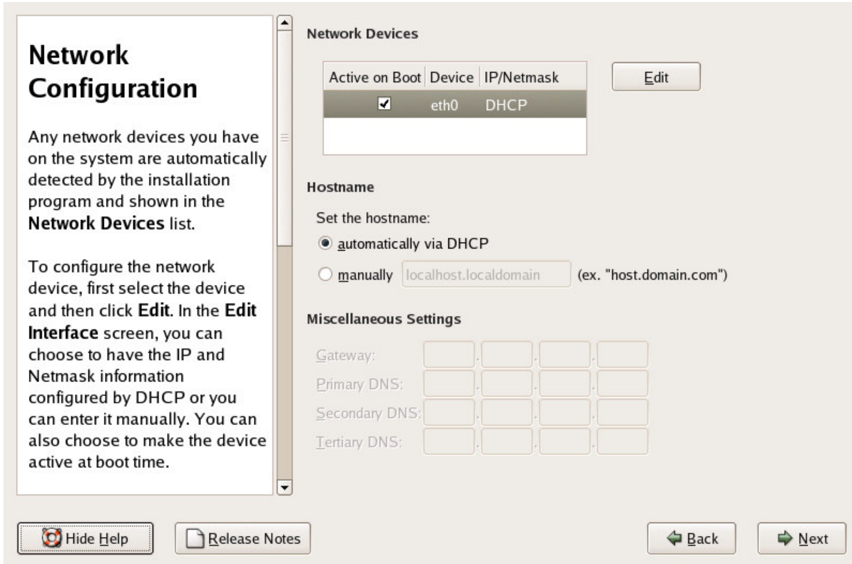


- Choose remove all partitions on this system
- Click Next to continue setup



- This is you final warning. You will loose 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

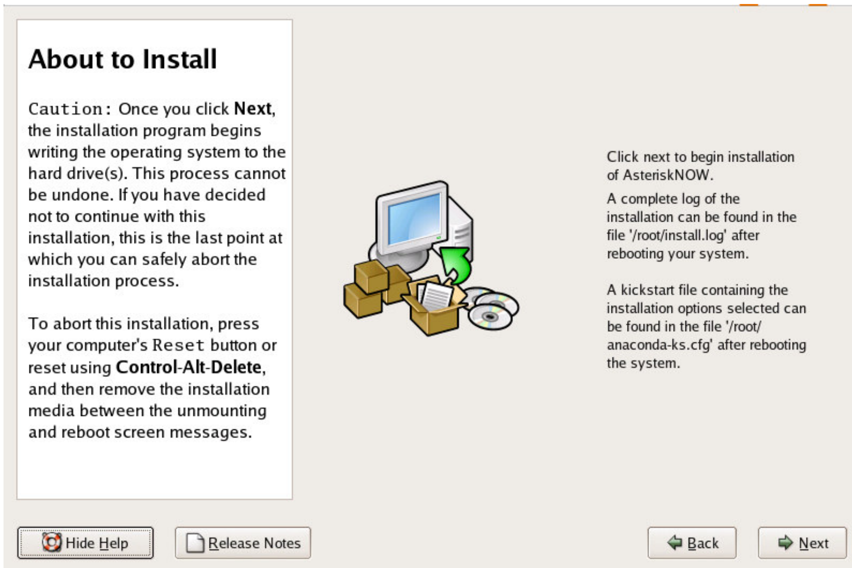
6.3. Time Zone Selection:

- 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:

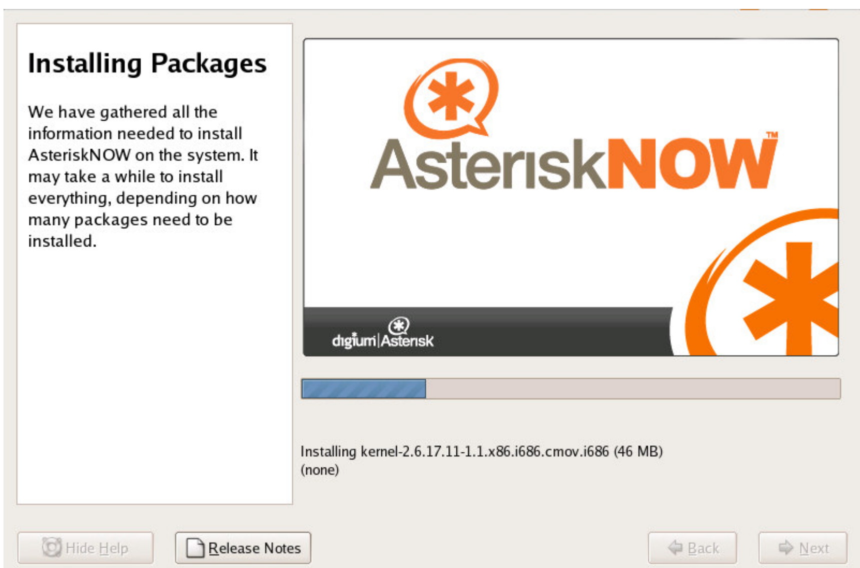


- Click Next to continue setup

6.6. Formatting the System:

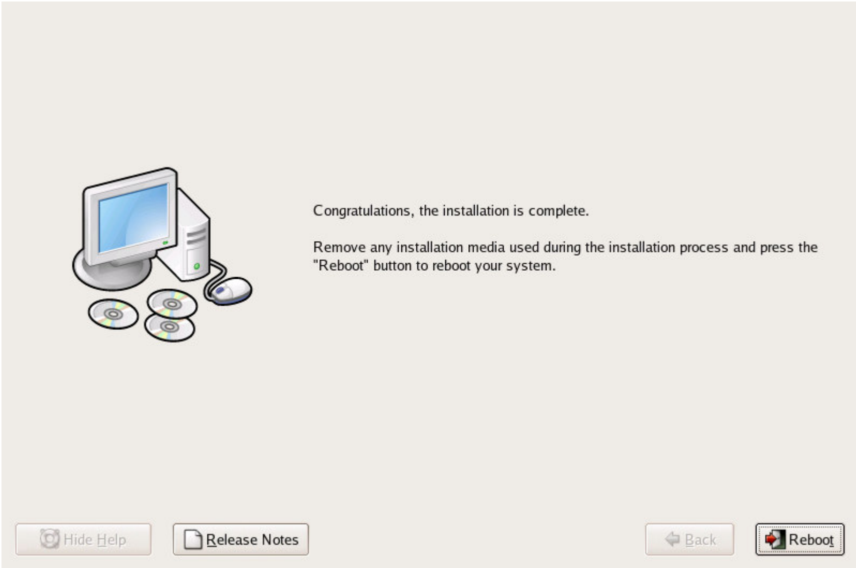
- Click Next to continue setup

6.7. Installing the Packages:



6.8. Running post Installation scripts:

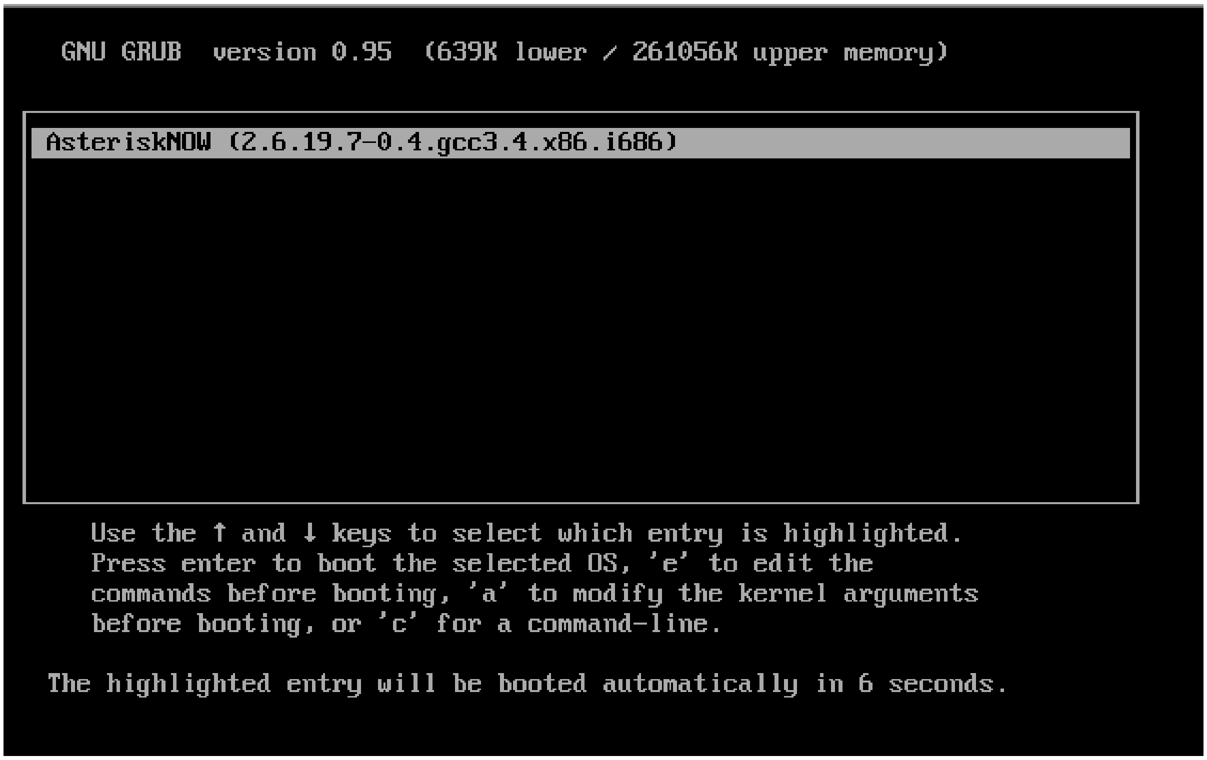
6.9. Finishing installation:



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

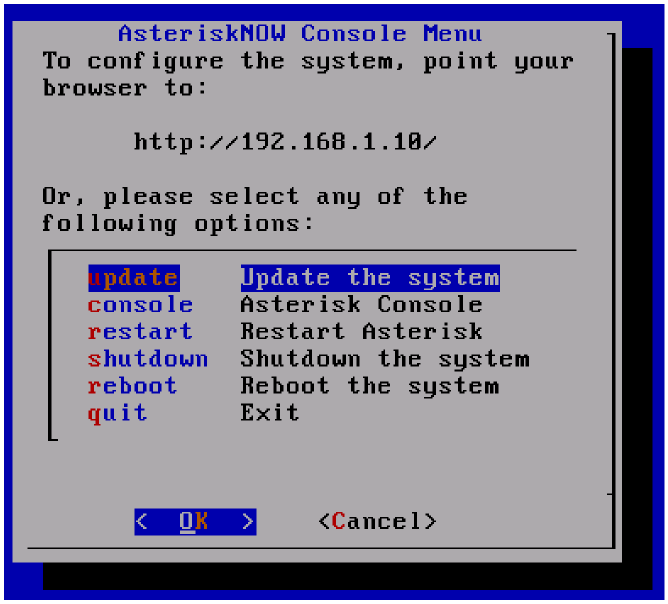
7. Starting AsteriskNOW:

- 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:

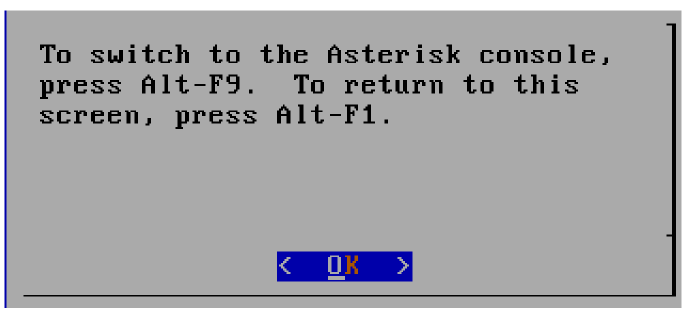


- Make a note of the Ip address displayed in the AsteriskNOW console menu. You need to use it to access the web menu later.

7.1.1. Update menu:

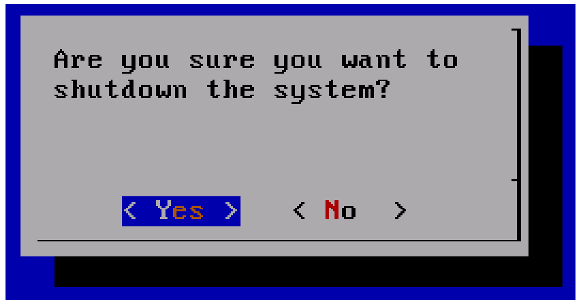
Use the update menu to update your system manually.

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:

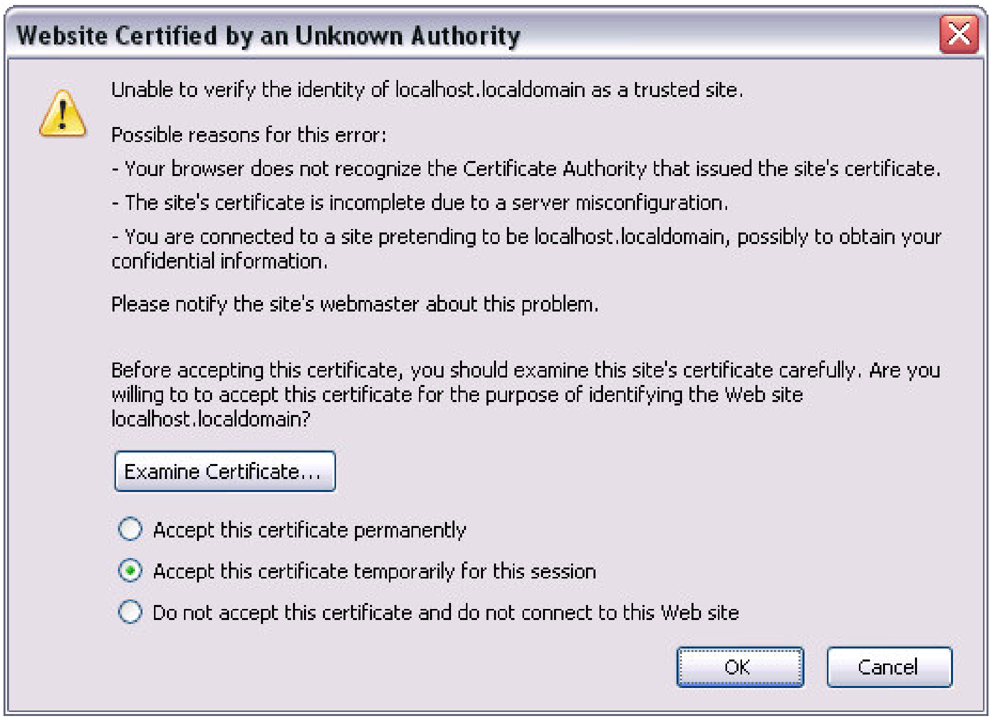
8. Configuring AsteriskNOW:

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.

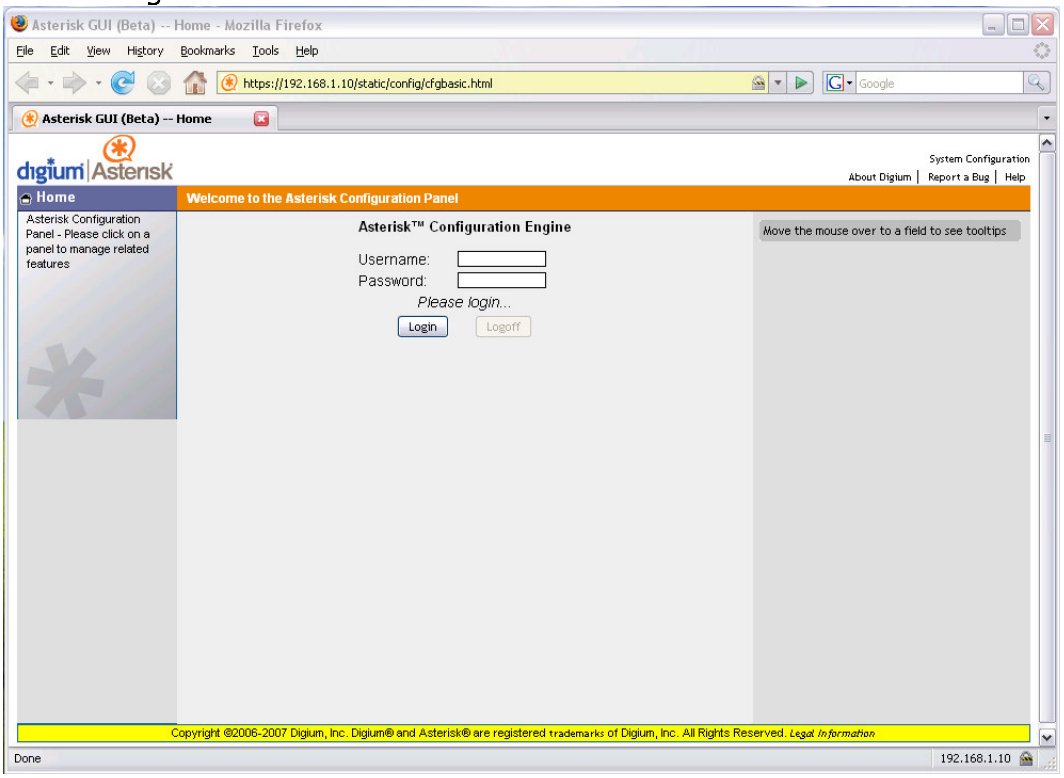
You can get the IP address by looking at the AsteriskNow console menu.

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.

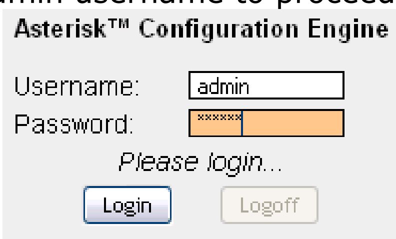


Then only you will be able to access the web based administration interface of AsteriskNOW.

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.

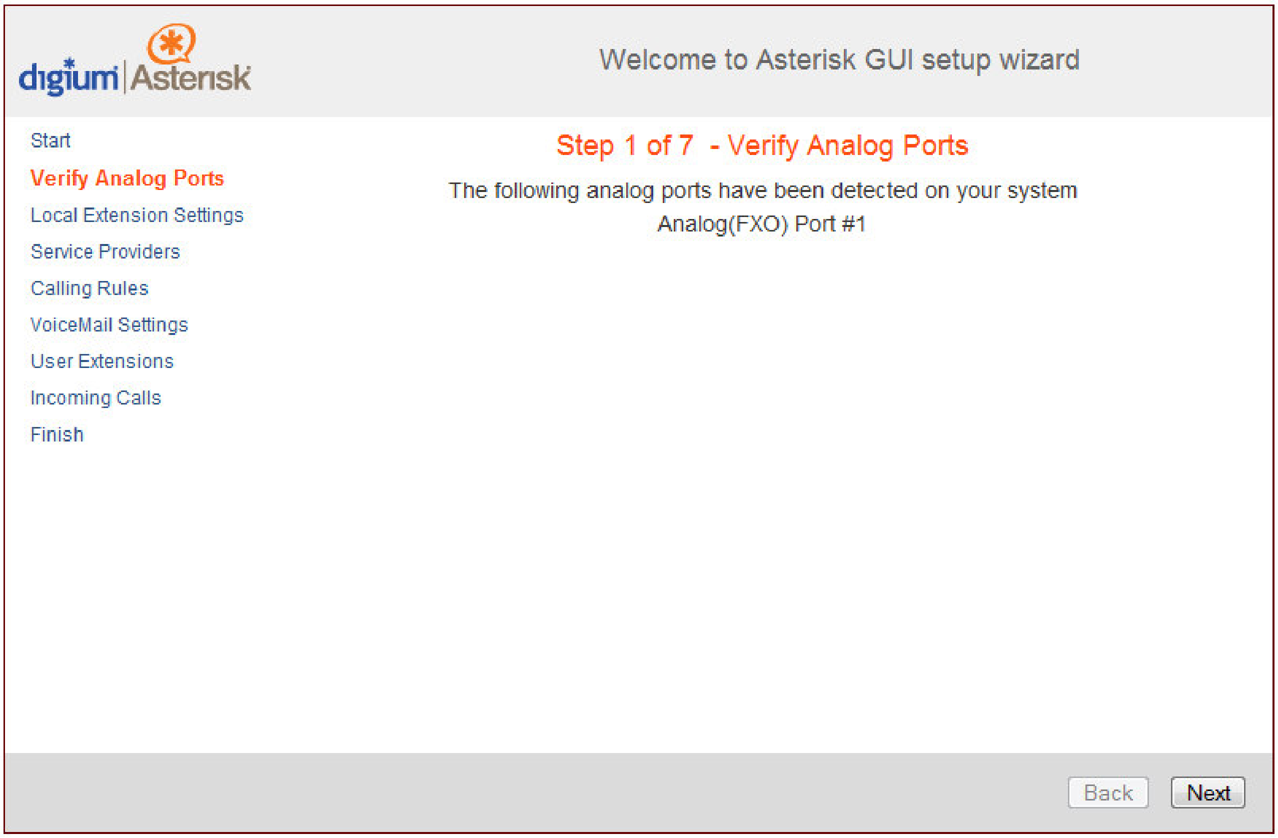


Username: admin
Password: xxxxx (the password you set when installing the system)

8.1. Setup Wizard:

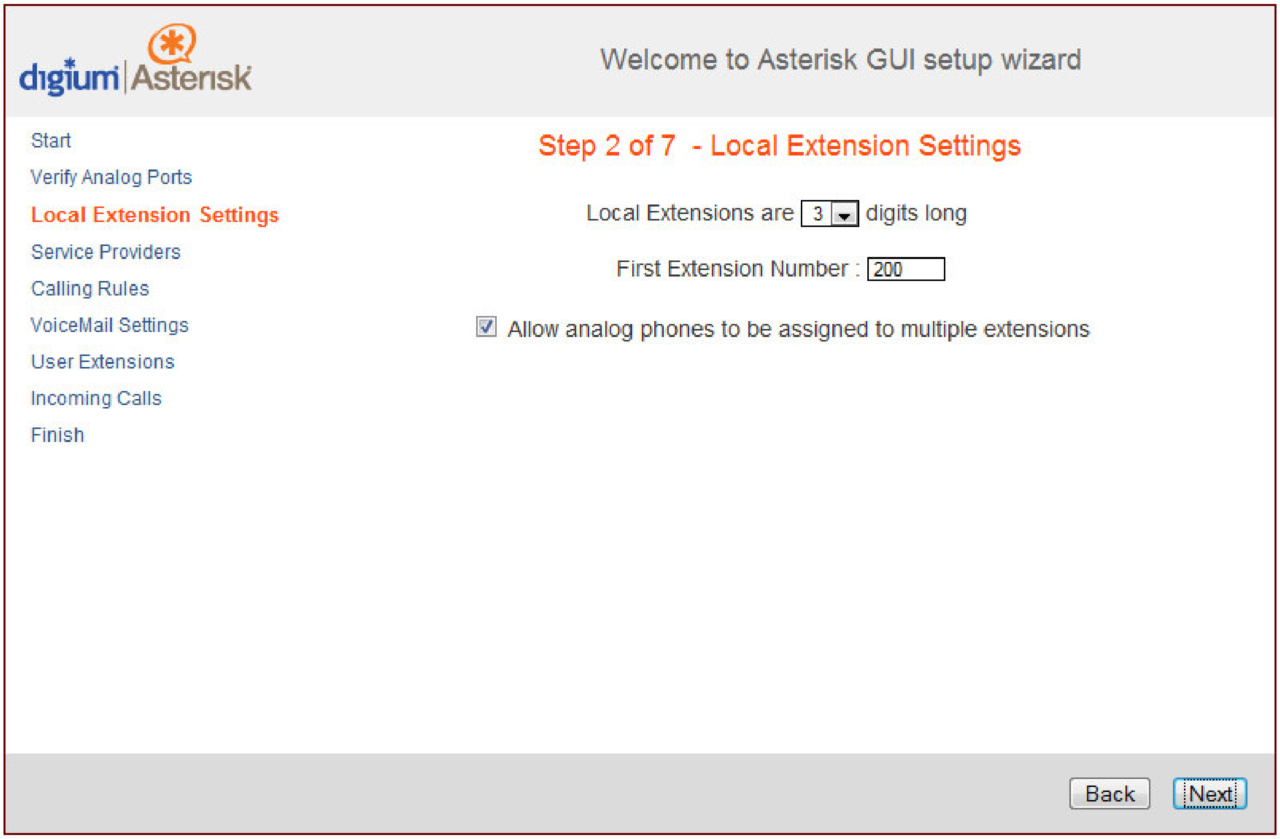
At first login, the Wizard is started automatically for you to walk through the steps. Just follow the prompts and finish the wizard's task.

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.

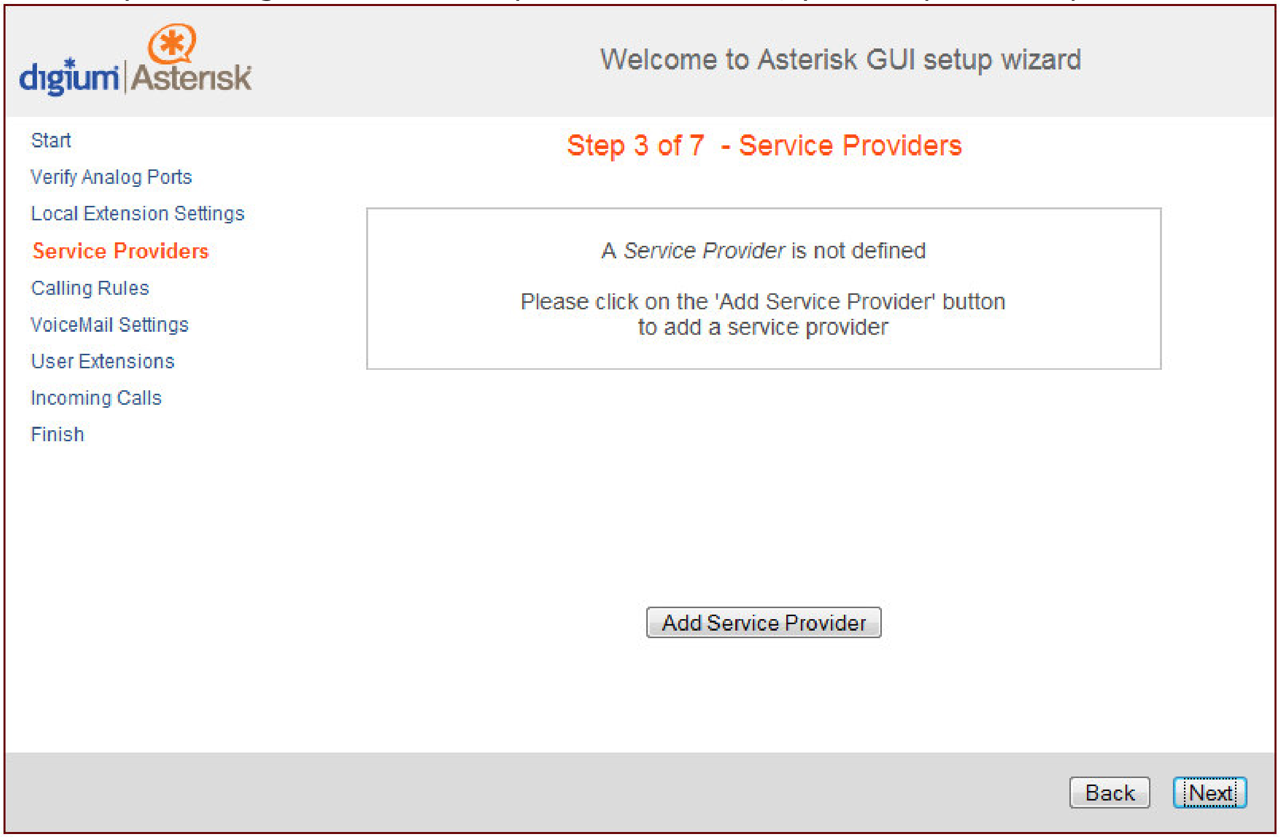
8.1.2. Local Extension Settings:



- 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.

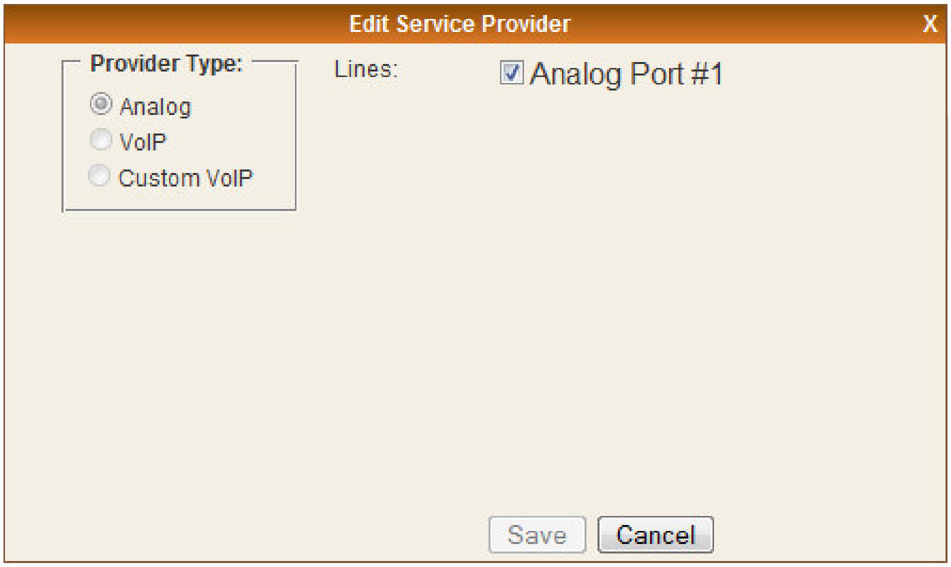


If you want to dial out to the rest of the world then you need to have at least one provider, who will carry your calls.

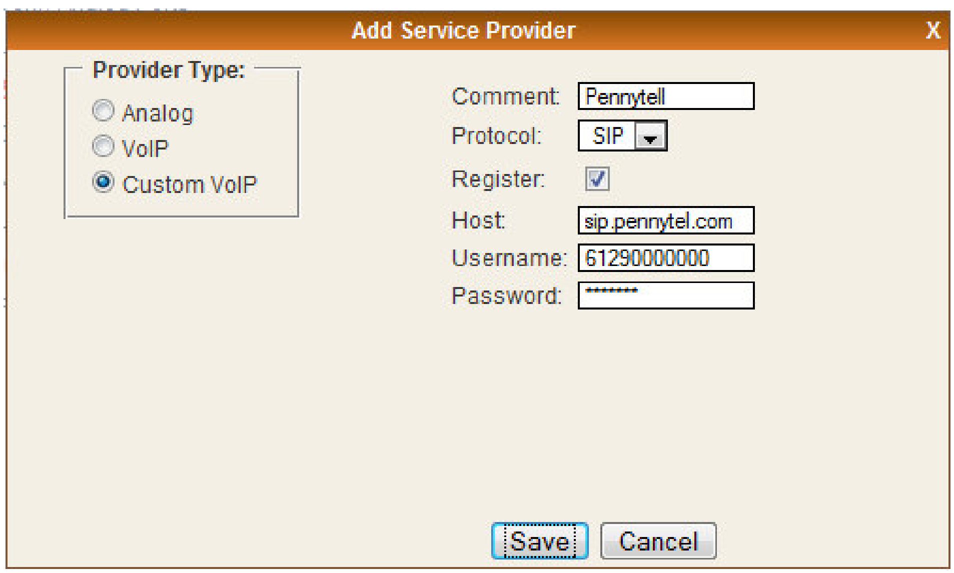
When you click on the Add Service Provider the following screen appears.
You have a choice of selecting three types of service providers.

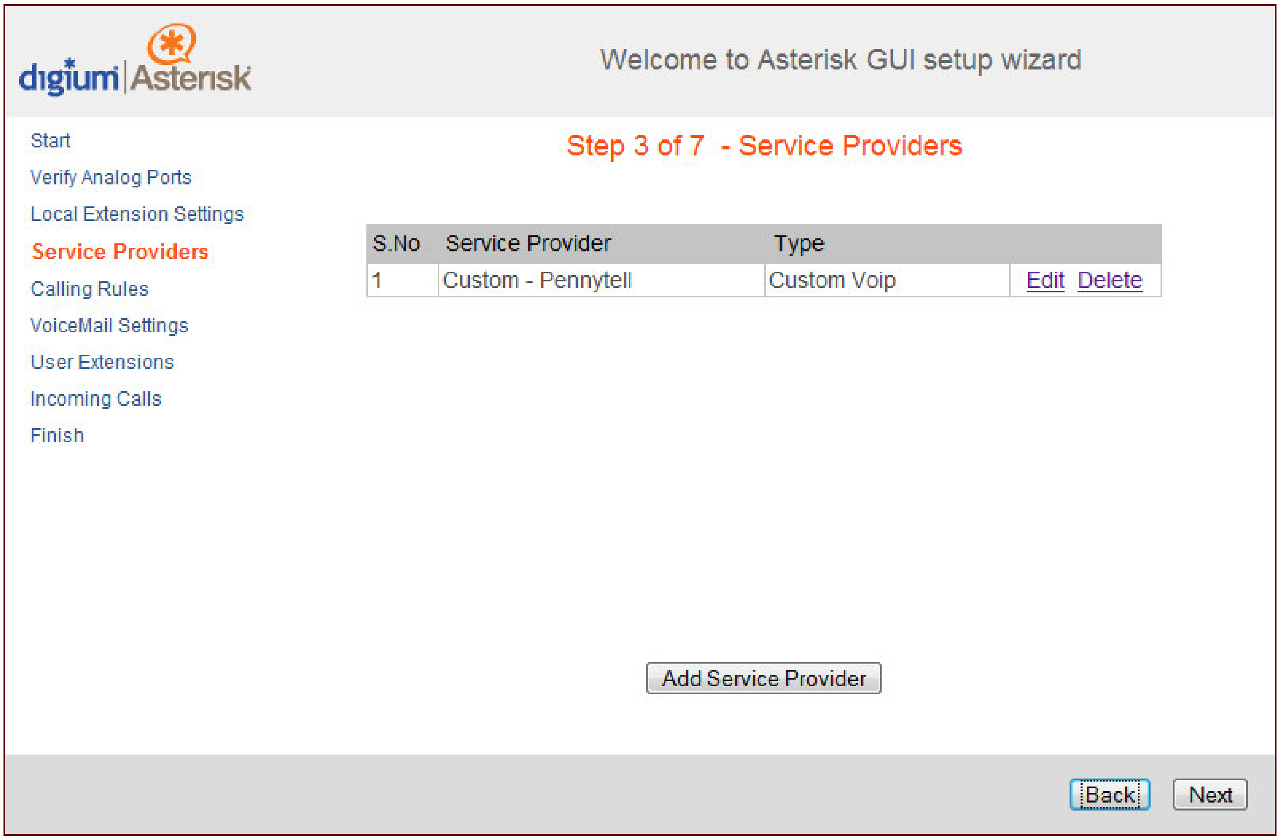
1) Analog
2) Voip
3) Custom Voip

Adding an Analog Port:



Adding a Custom Voip Service provider:



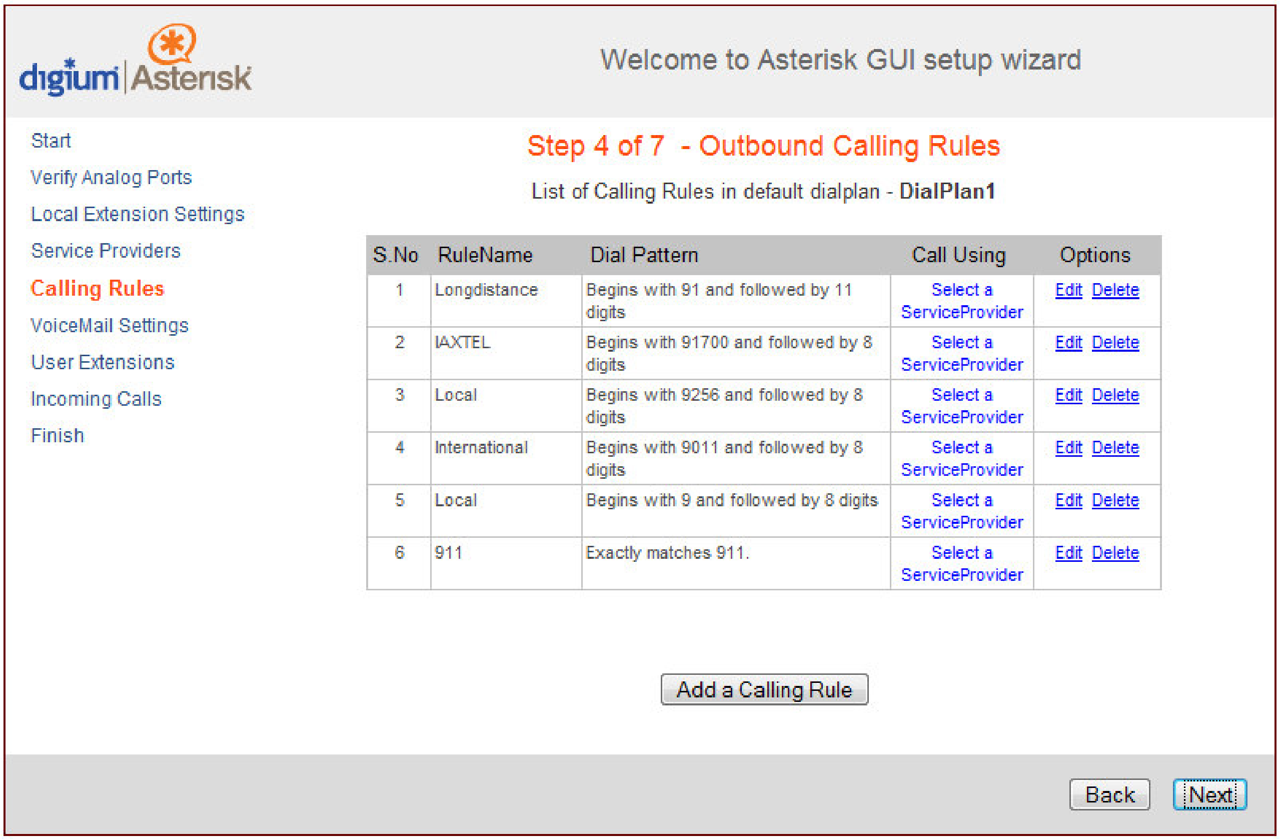

digium Asterisk
Welcome to Asterisk GUI setup wizard
Start
Verify Analog Ports
Local Extension Settings
Service Providers
Calling Rules
VoiceMail Settings
User Extensions
Incoming Calls
Finish
Step 3 of 7 - Service Providers
S.No
Service Provider
Type
1
Custom - Pennytell
Custom Voip
Edit
Delete
Add 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.



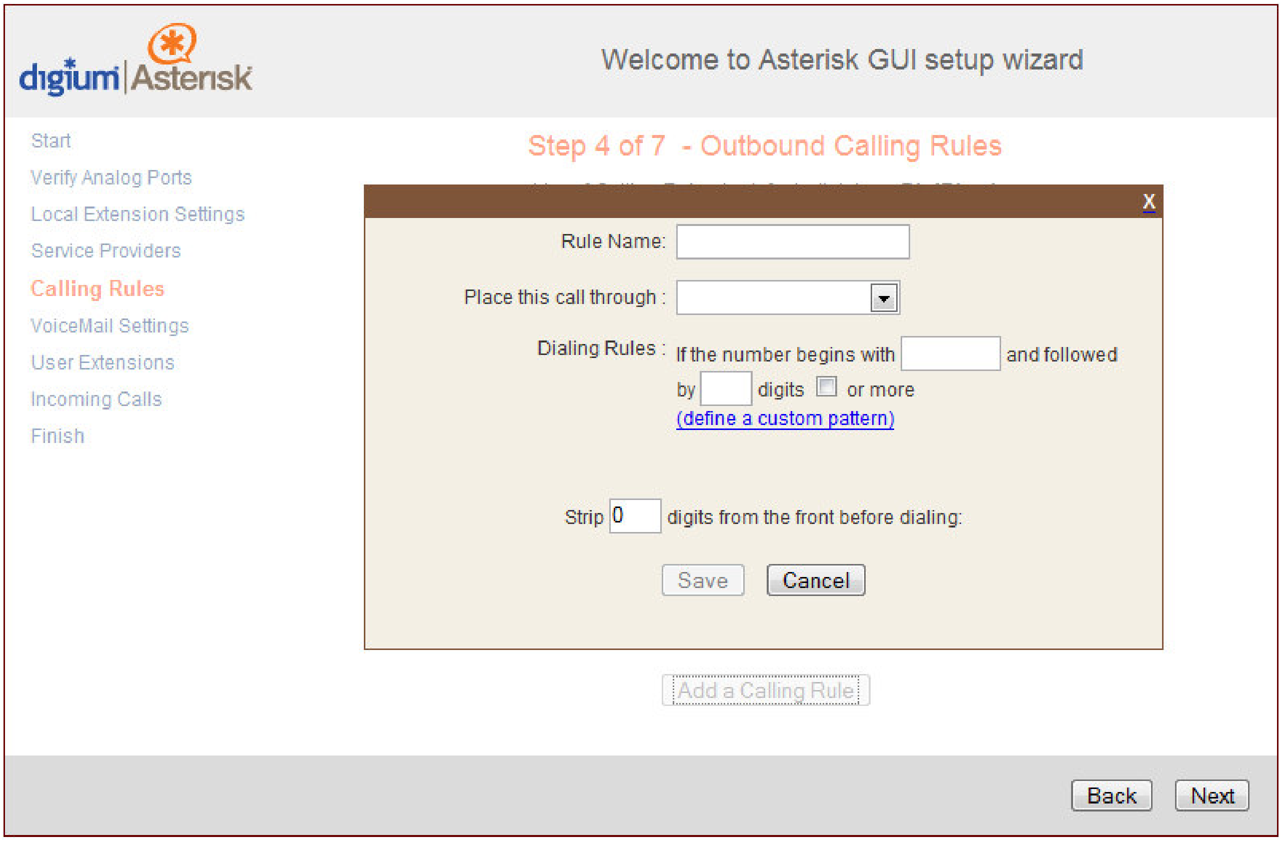
Australia:

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 XXXXXXXXXXXX)

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.

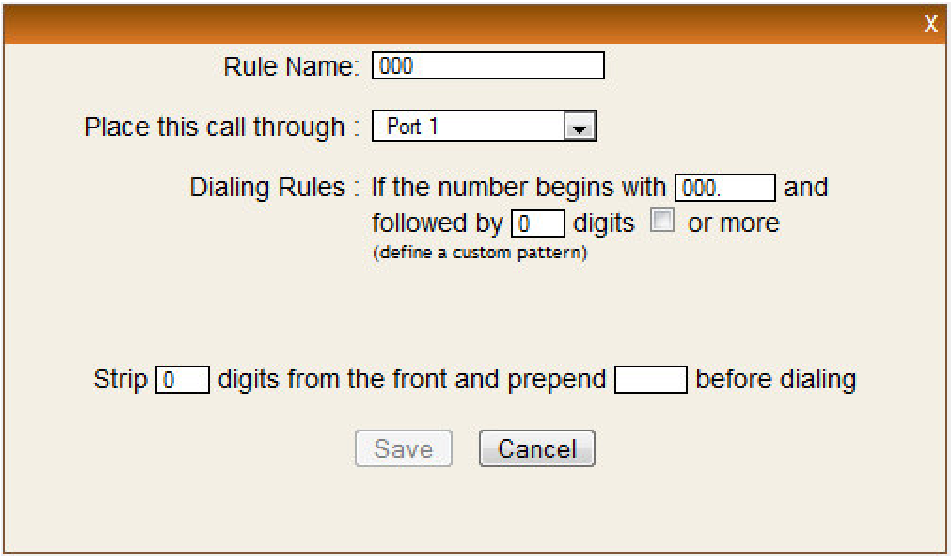


Rule for local Numbers: (For Starting with 9)

Rule for local Numbers: (For Starting with 8)

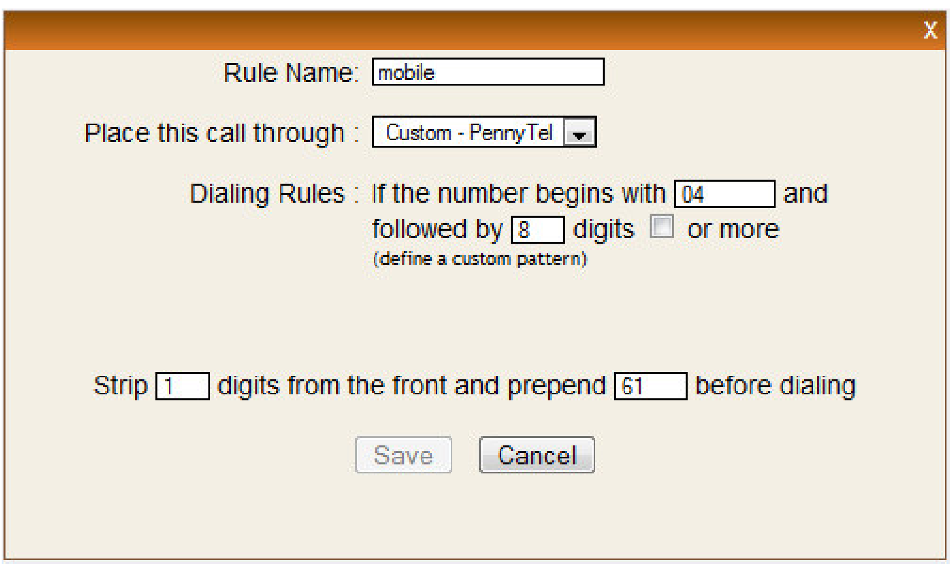
Rule for Long-distance National Call:

Rule for Emergency calls:



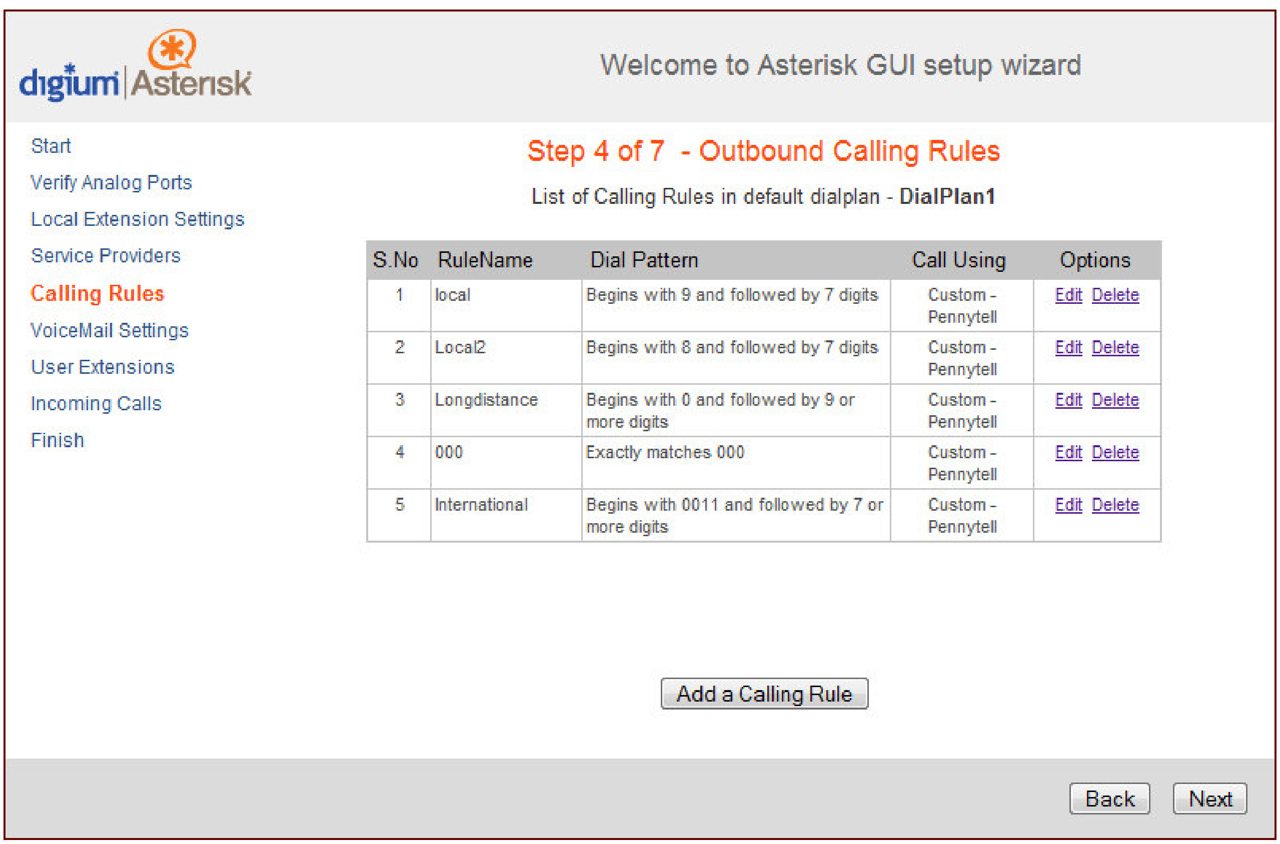
Remember it is always recommended to use your local landline to route your Emergency calls. Here I have chosen my analog Port 1 to route all of my emergency calls that start with 000. My analog port is an Xp100 card a very old version, but it still works.

Rule for Mobile Calls:



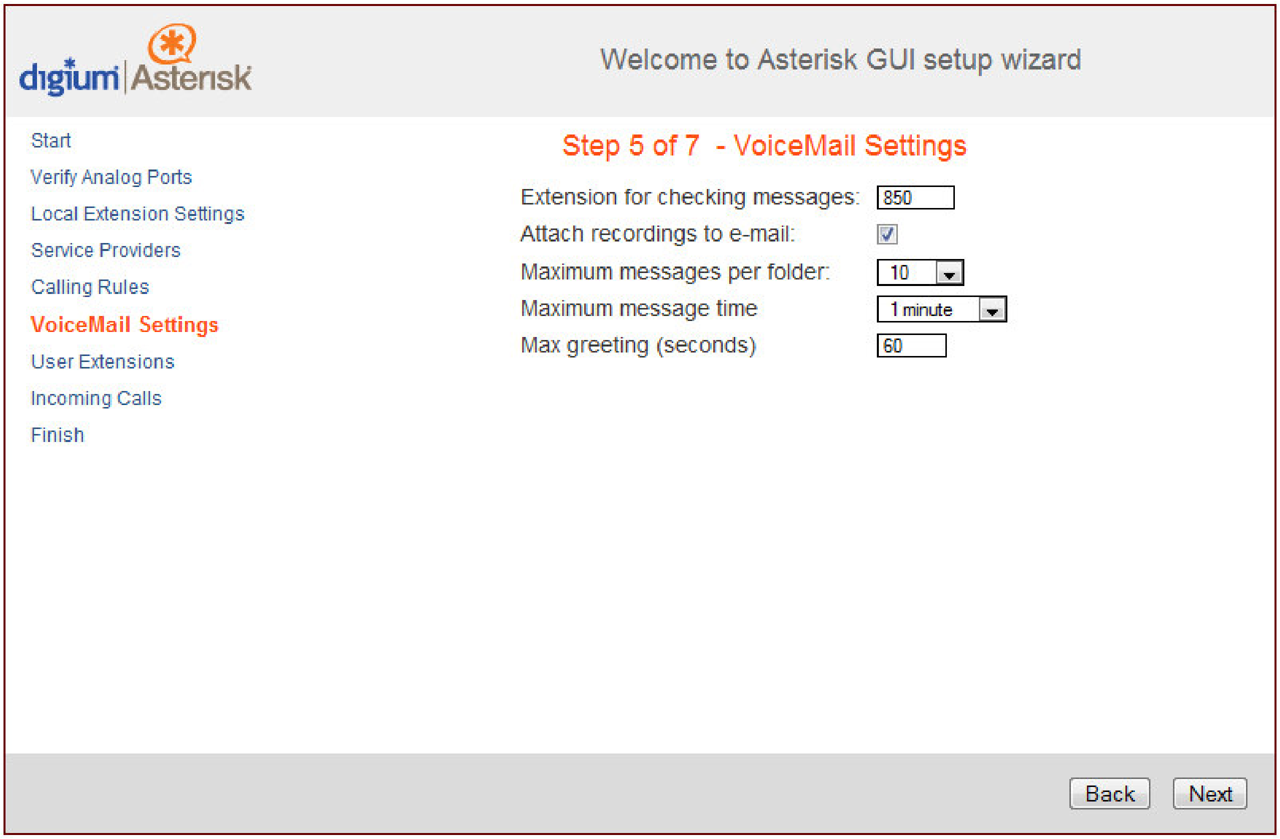
Rule for International Calls:

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



Click on next to continue.

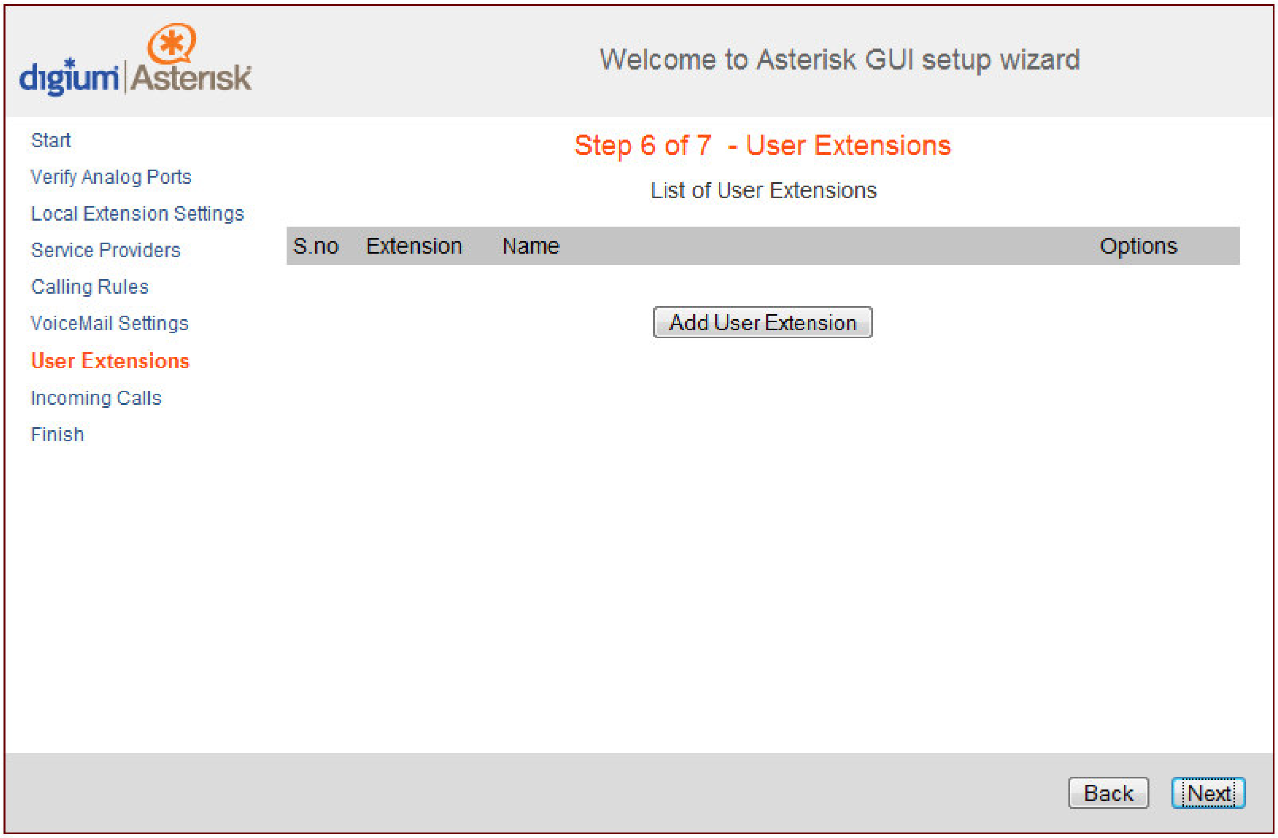
8.1.5. Voicemail:



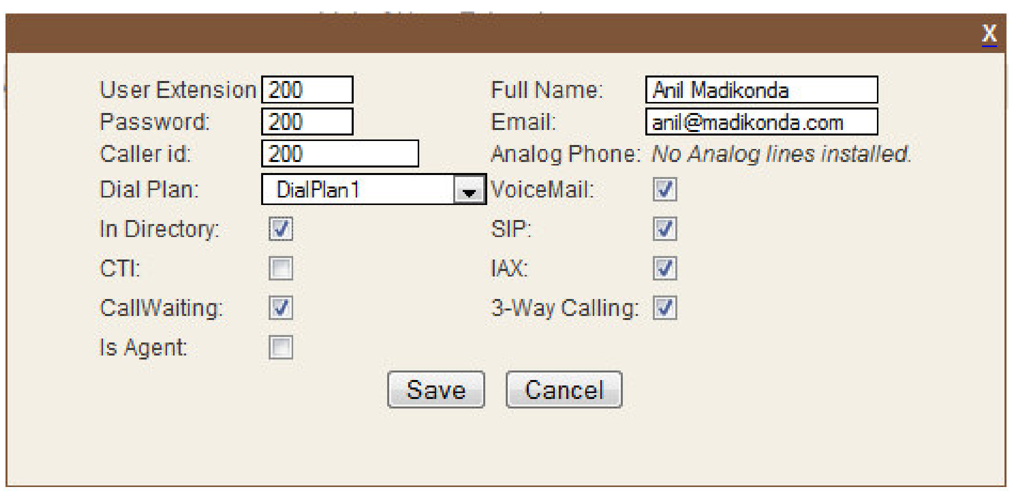
In this section you need to specify your voicemail settings.
Take the time to setup these as well and then click on next to continue.

8.1.6. Extensions:

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

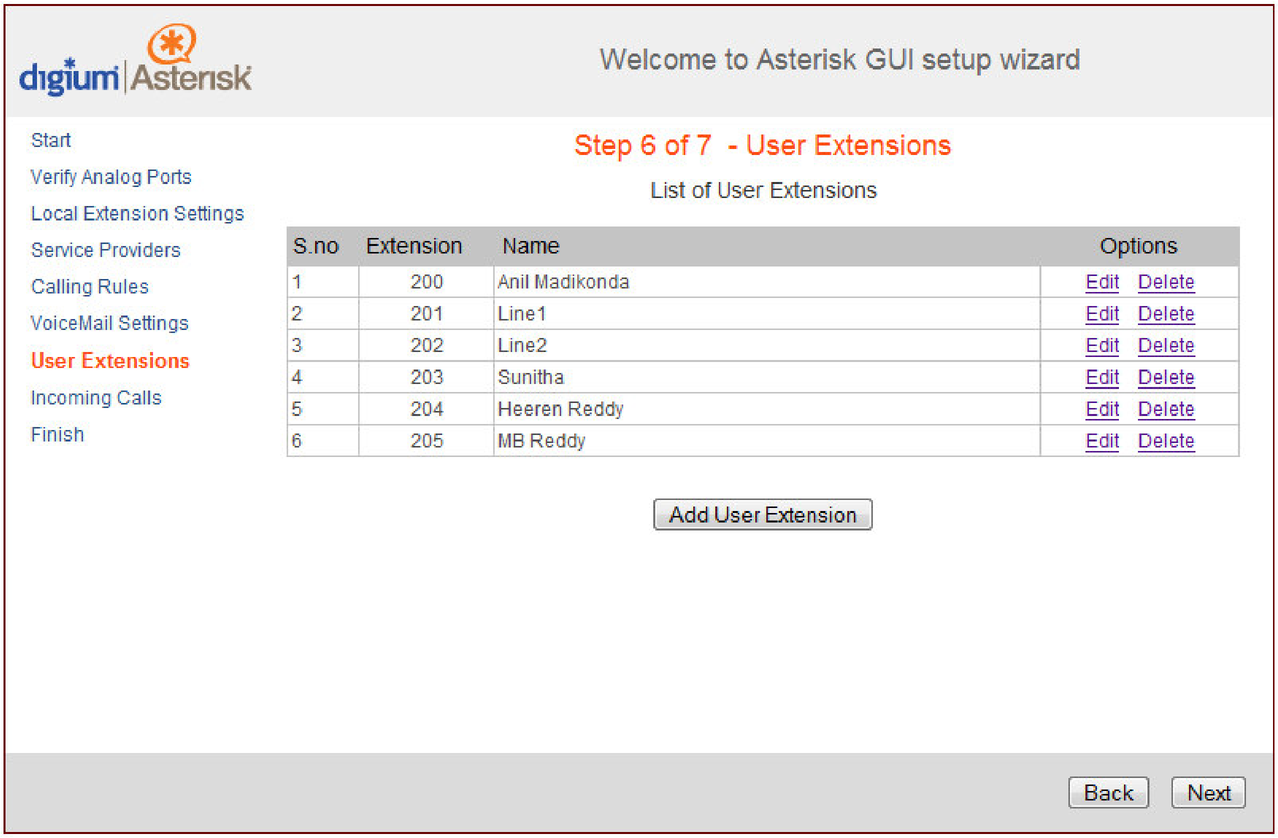


Click on Add User Extension to start the form to create new extensions as below.



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.

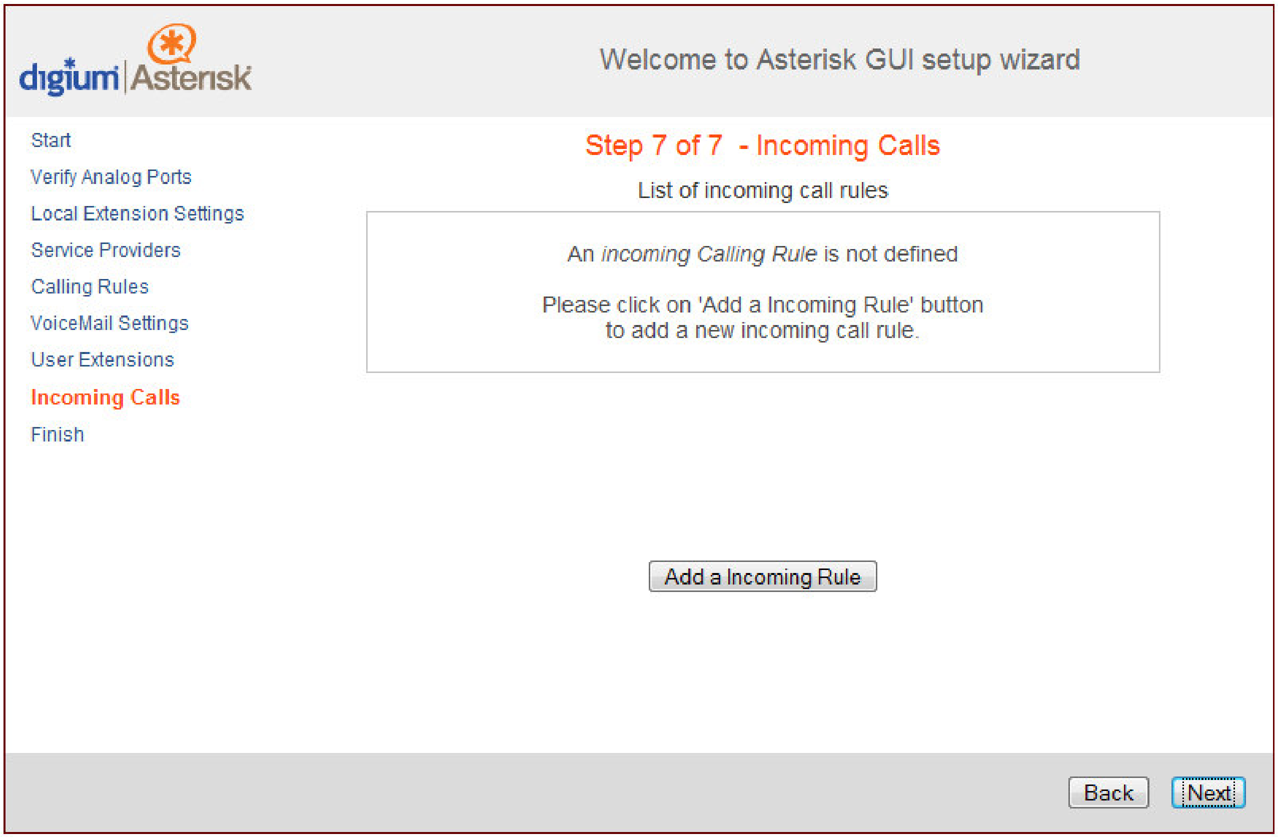
Once you have created all the extensions that you need, then your screen should look similar to this.



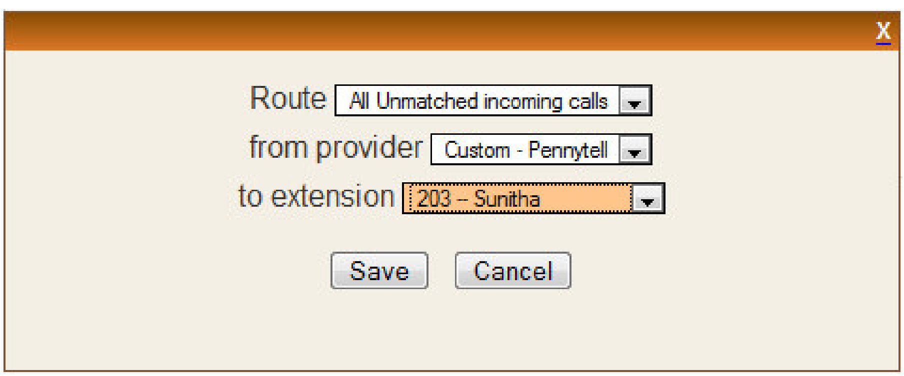
Once you are finished entering all of your user extensions, then click on next to continue.

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.

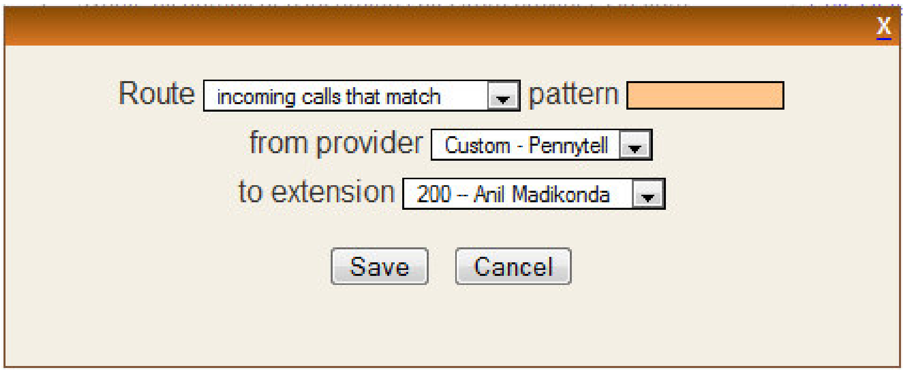


I have setup a default rule that applies to all unmatched calls. These call are routed to the extension 203. You can also setup a rule to match a certain incoming number and then route it to a different number.

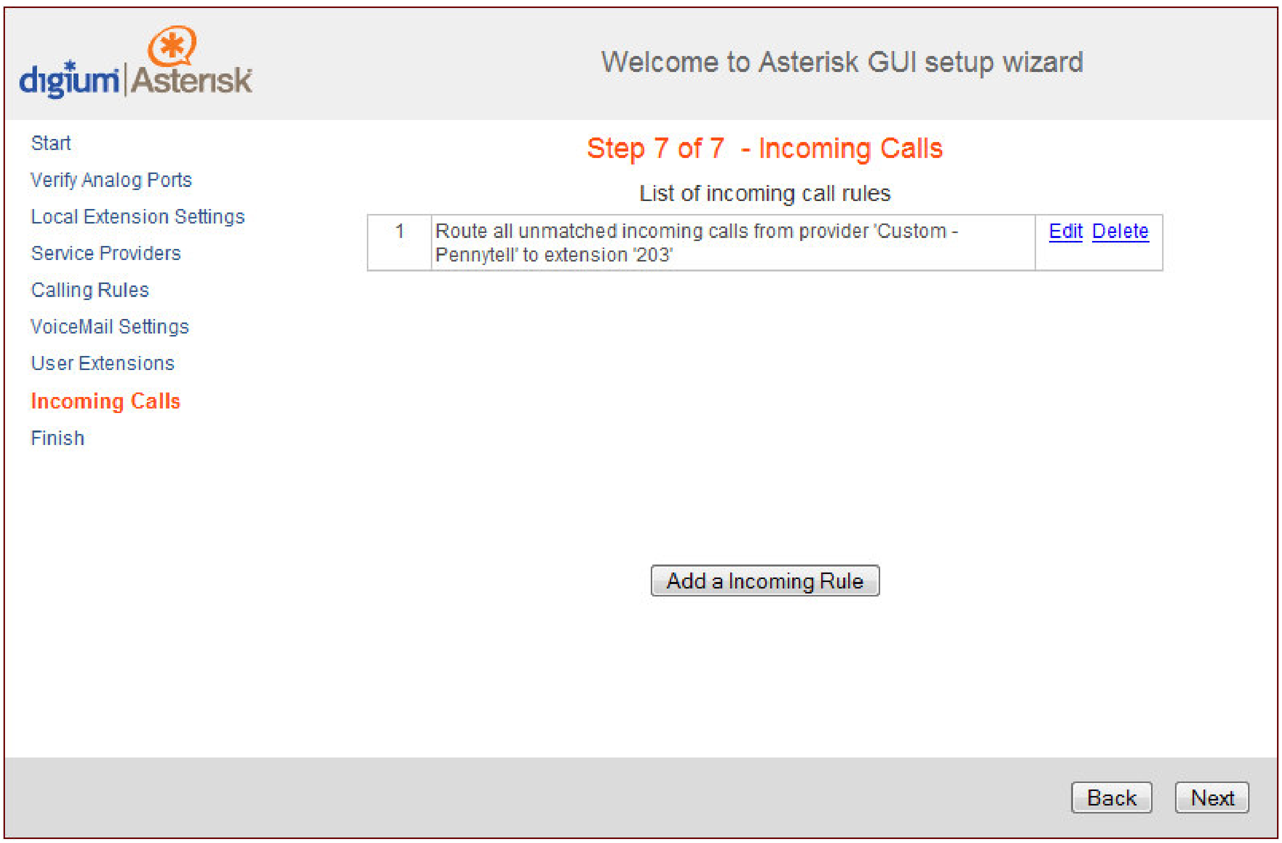


Click on save button to save the rule.

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

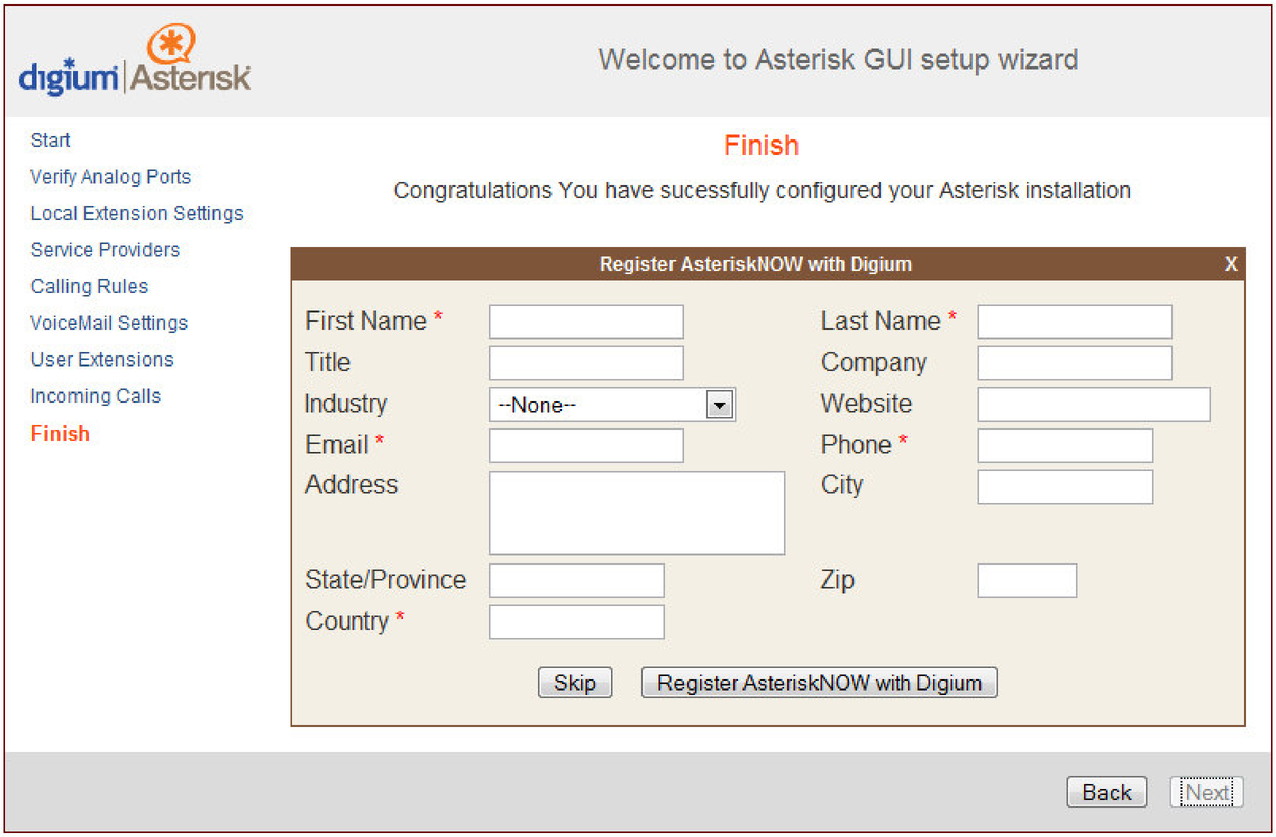


Click on save to save the rule.

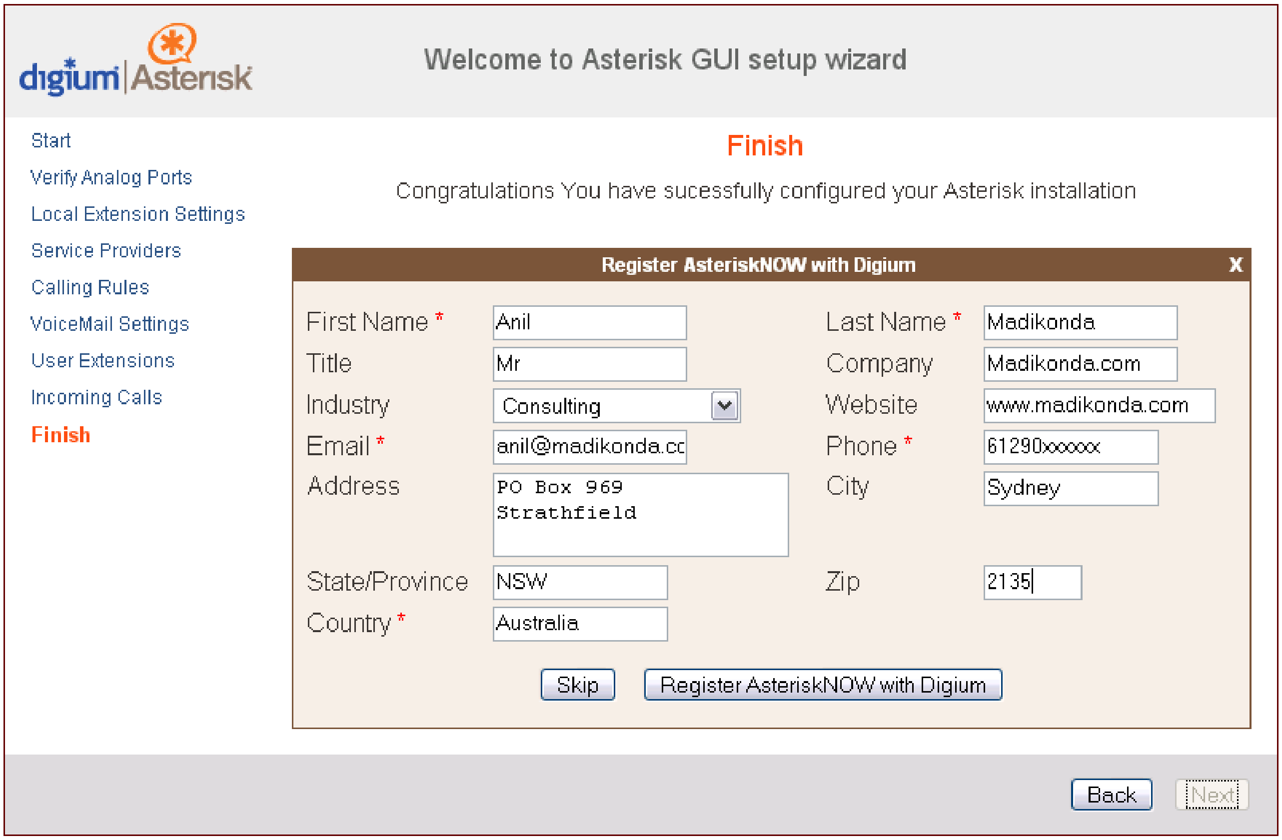


After you have created your incoming rules, then click on next to continue.

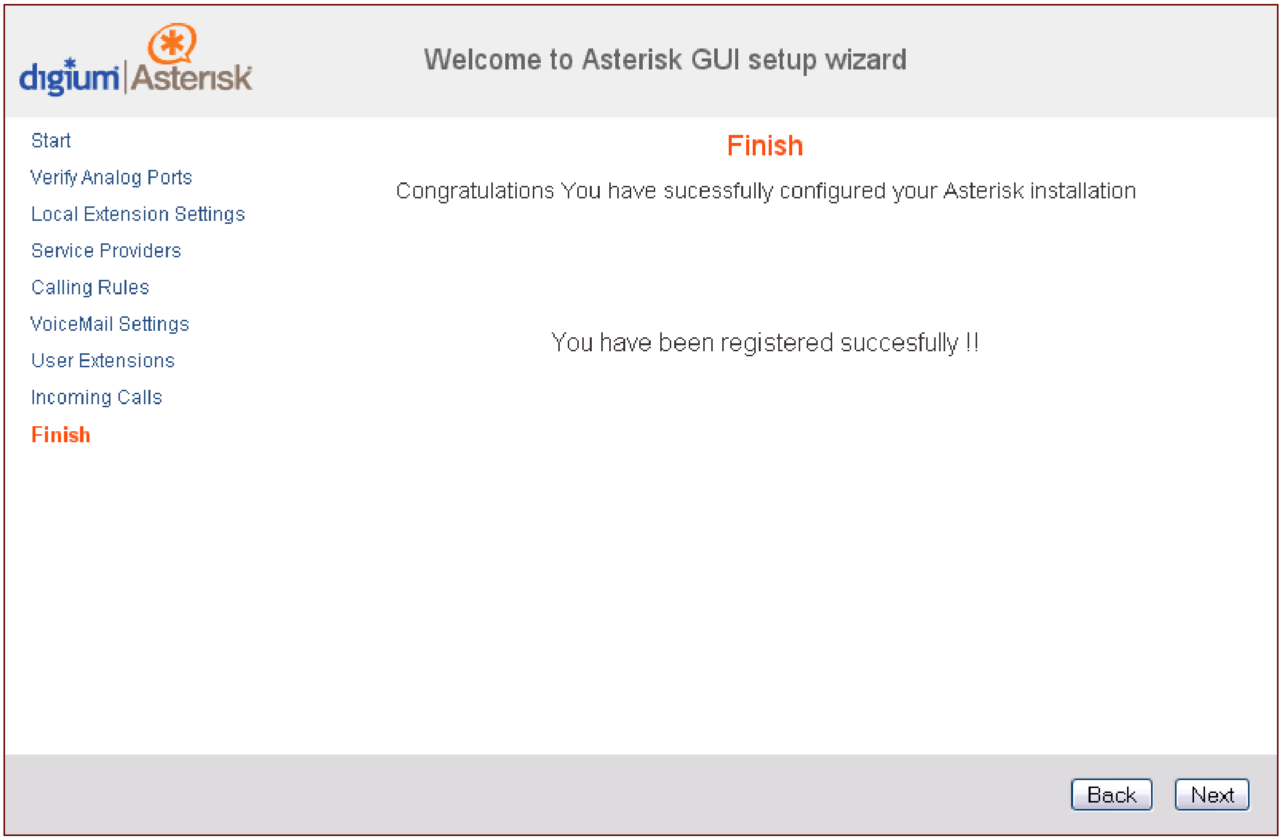
8.1.8. Register your copy of AsteriskNOW:



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

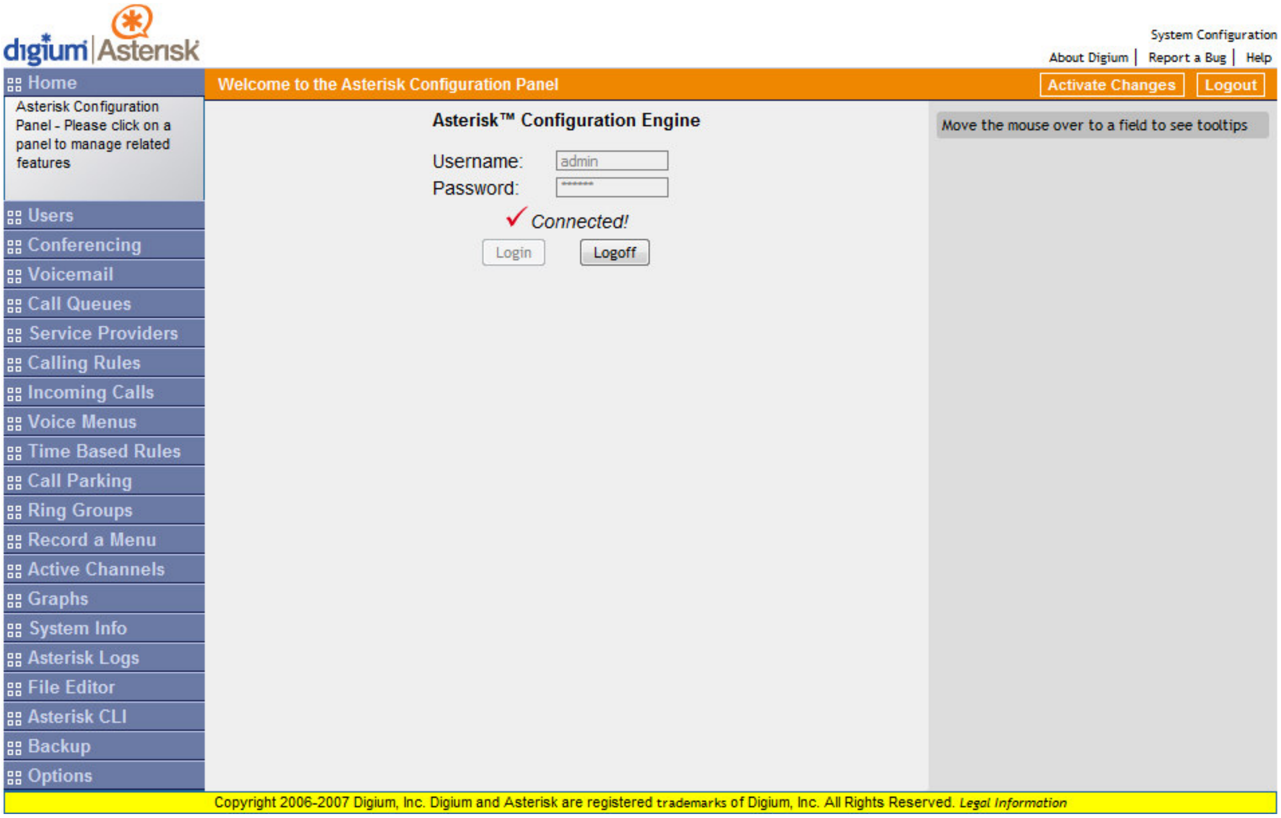


Click on register AsteriskNow with Digium. This will bring up a small window as below, while your details are sent to digium.



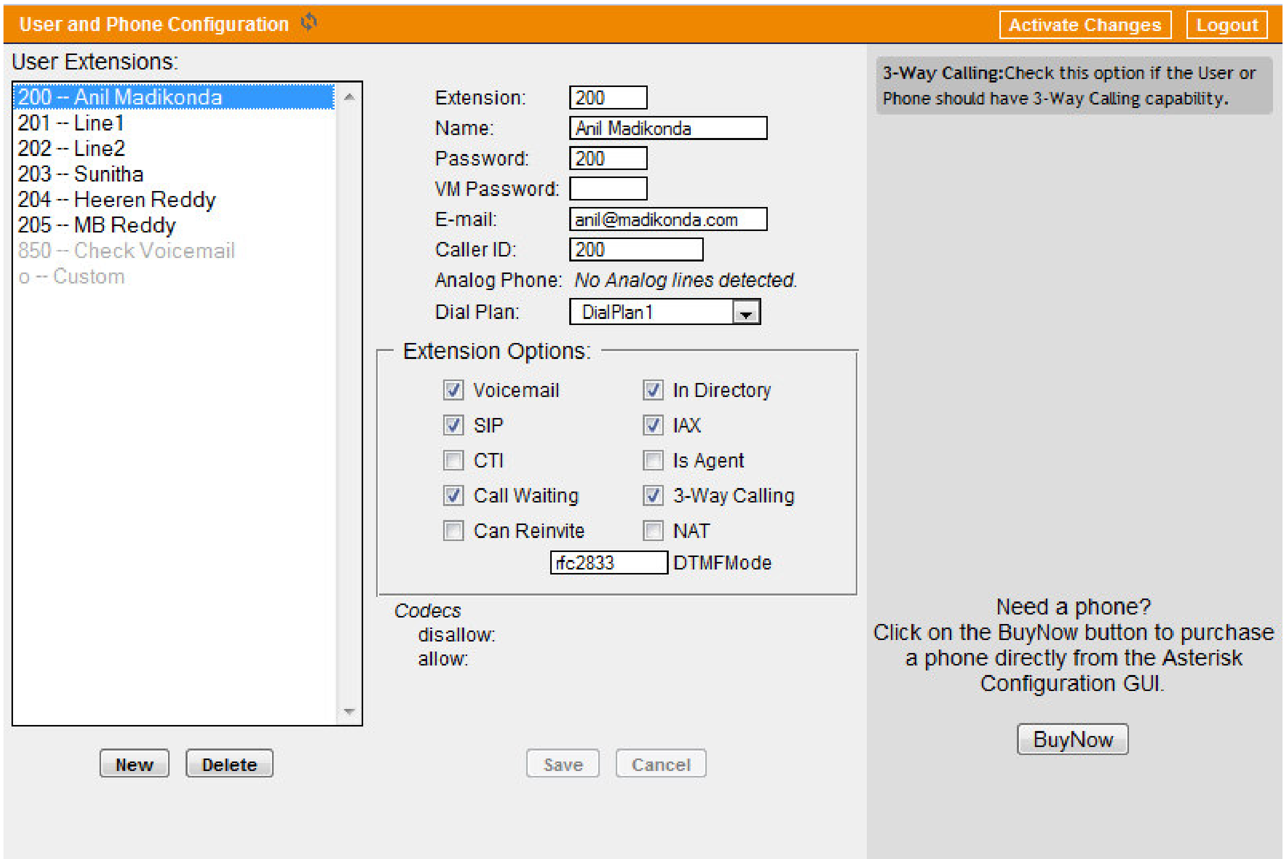
Click next to finish the Wizard setup process.

8.2. Asterisk Configuration Panel:



8.3. User & Phone Configuration:

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



General:

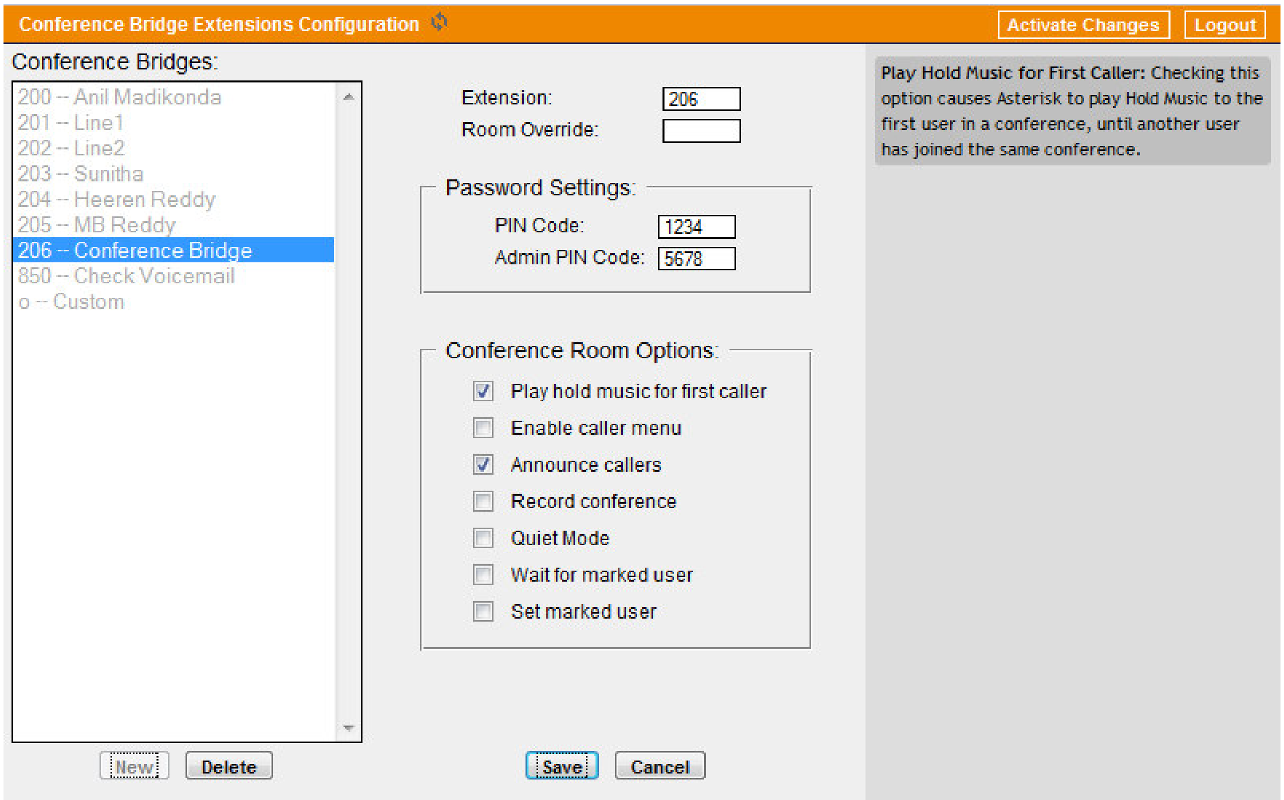
Extension: The numbered extension, i.e. 1234, that will be associated with this particular User / Phone.
Name: A character-based name for this user, i.e. "Bob Jones"
Password: The password for the user's sip/iax account , Ex: "12u3b6"
VM Password: Voicemail Password for this user, Ex: "1234".
E-Mail: The e-mail address for this user, i.e. bobjones@bobjones.null
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.



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.

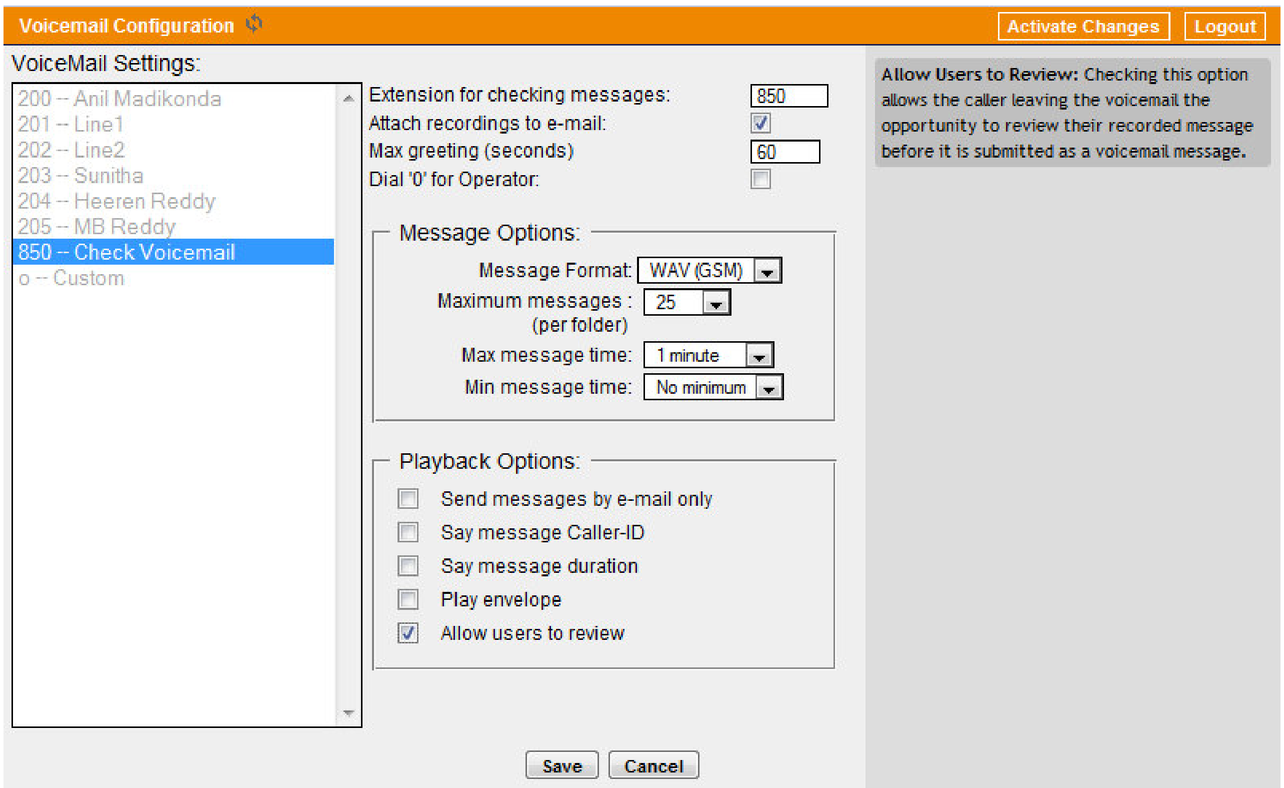
Conference Room Options:

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



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 voicemenu or operator.

Message Options:

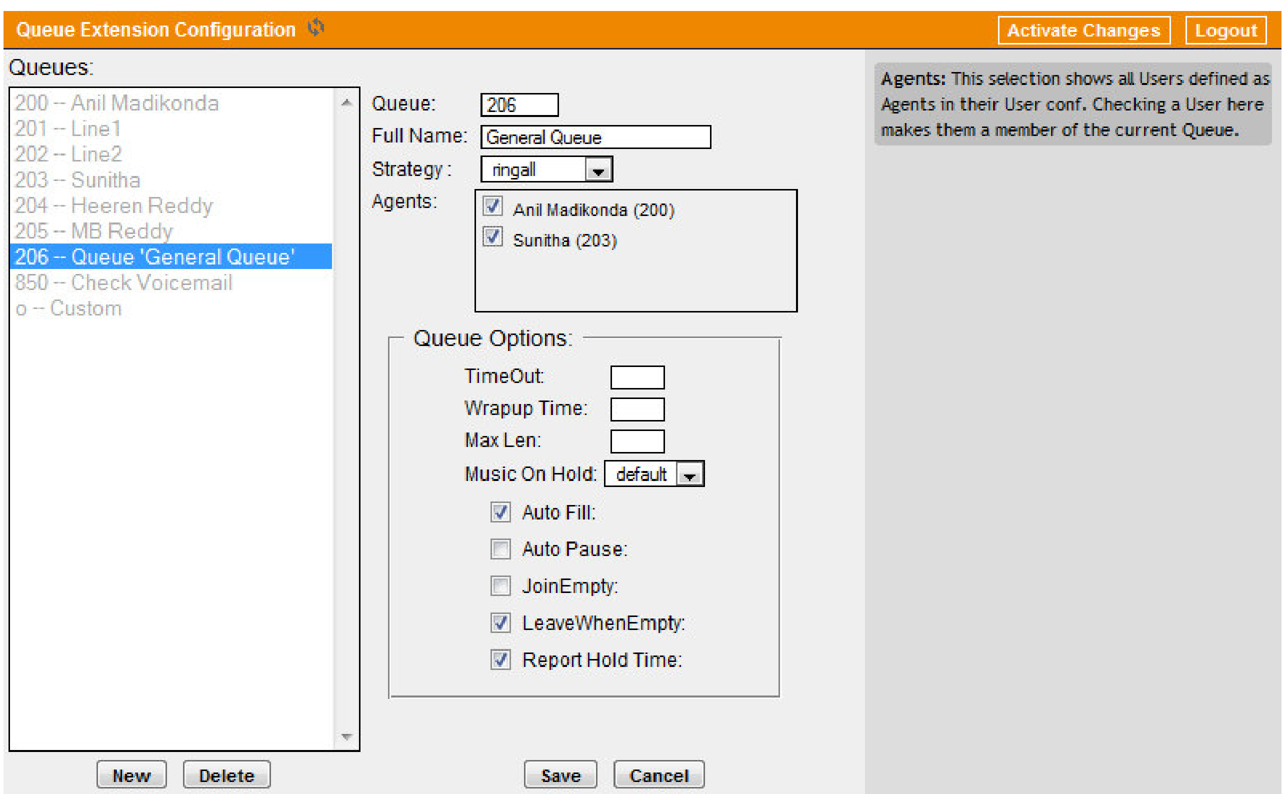
Message Format: This selection box controls the format in which messages are stored on the system and delivered by e-mail.
Maximum messages per folder: This select box sets the maximum number of messages that a user may have in any of their folders.
Maximum Message Time: This select box sets the maximum 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 optino 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: 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

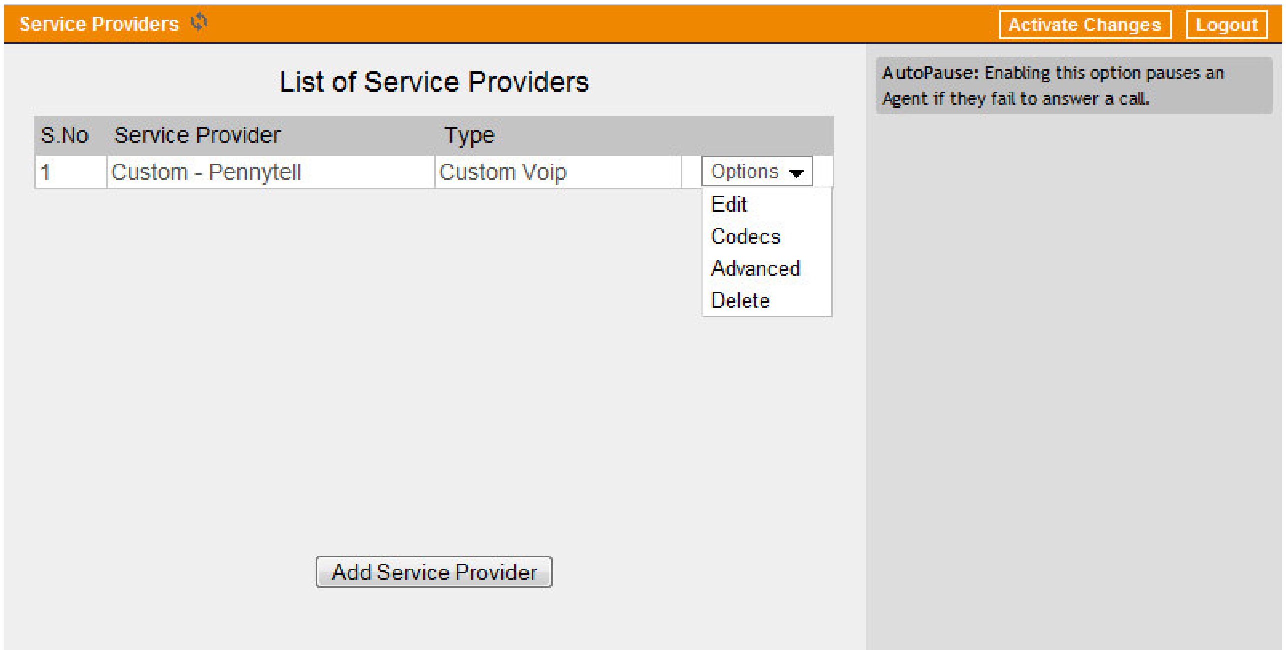
Agents: This selection shows all Users defined as Agents in their User conf. Checking a User here makes them a member of the current Queue.

Queue Options:

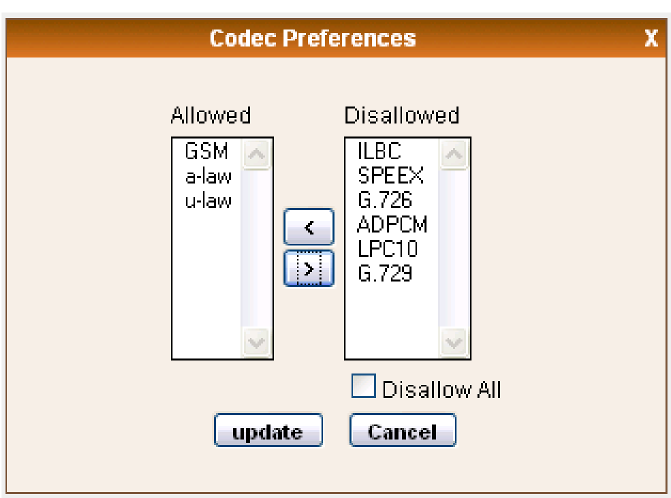
Timeout: This option defines the time in seconds that an Agent's phone rings before the next Agent is rung, i.e. "15"
Wrapup Time: After a successful call, time time in seconds that an Agent remains free before another call is sent to them. Default is 0, which is No Delay.
MaxLen: This option sets the 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.

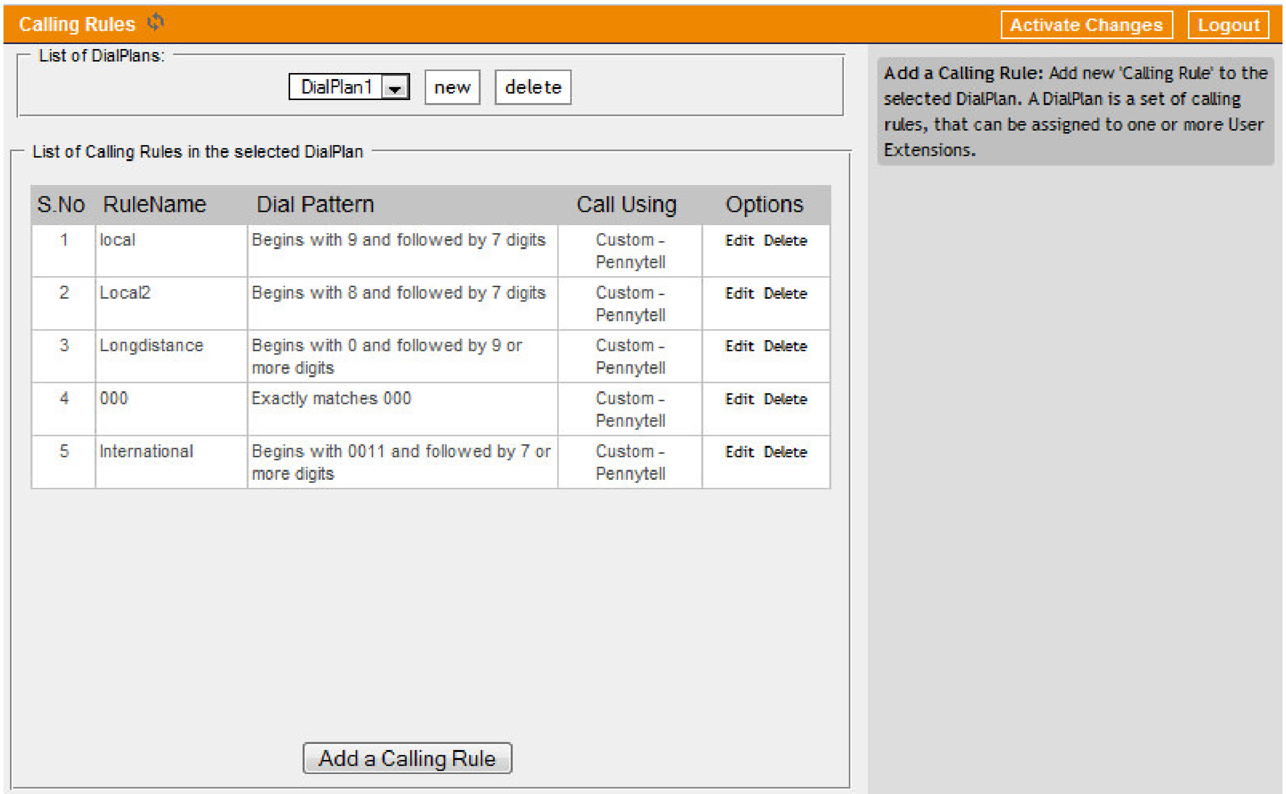


Analog/Voip Trunks: Analog lines are attached to analog interfaces of the PBX using FXO cards. Voice over IP (VoIP) connections are provided by an Internet Telephony Service Provider (ITSP).

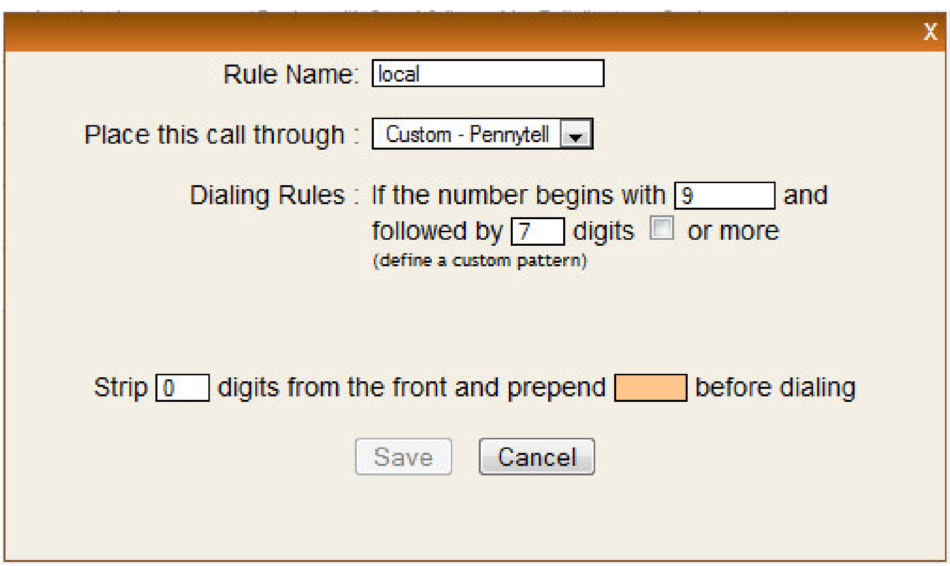


8.8. Calling Rules:

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

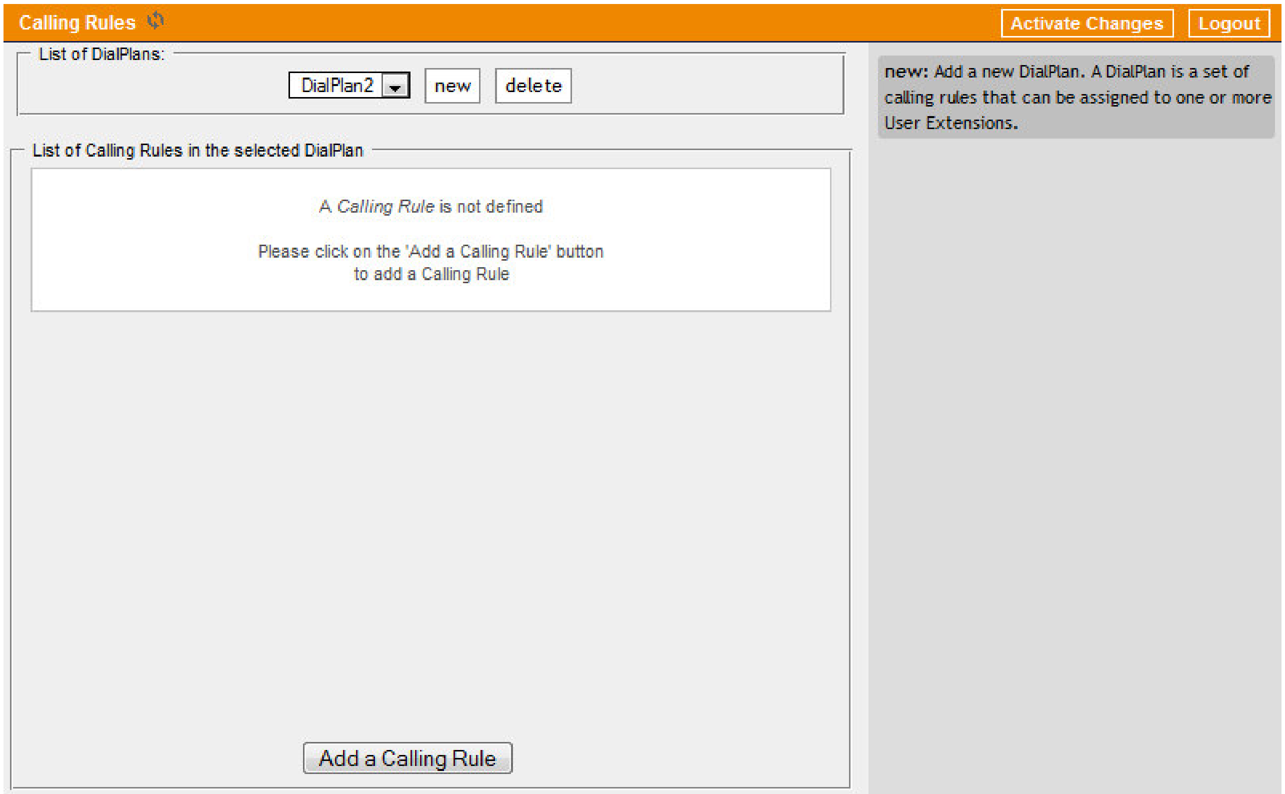


S.No	RuleName	Dial Pattern	Call Using	Options
1	local	Begins with 9 and followed by 7 digits	Custom - Pennytell	Edit Delete
2	Local2	Begins with 8 and followed by 7 digits	Custom - Pennytell	Edit Delete
3	Longdistance	Begins with 0 and followed by 9 or more digits	Custom - Pennytell	Edit Delete
4	000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



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

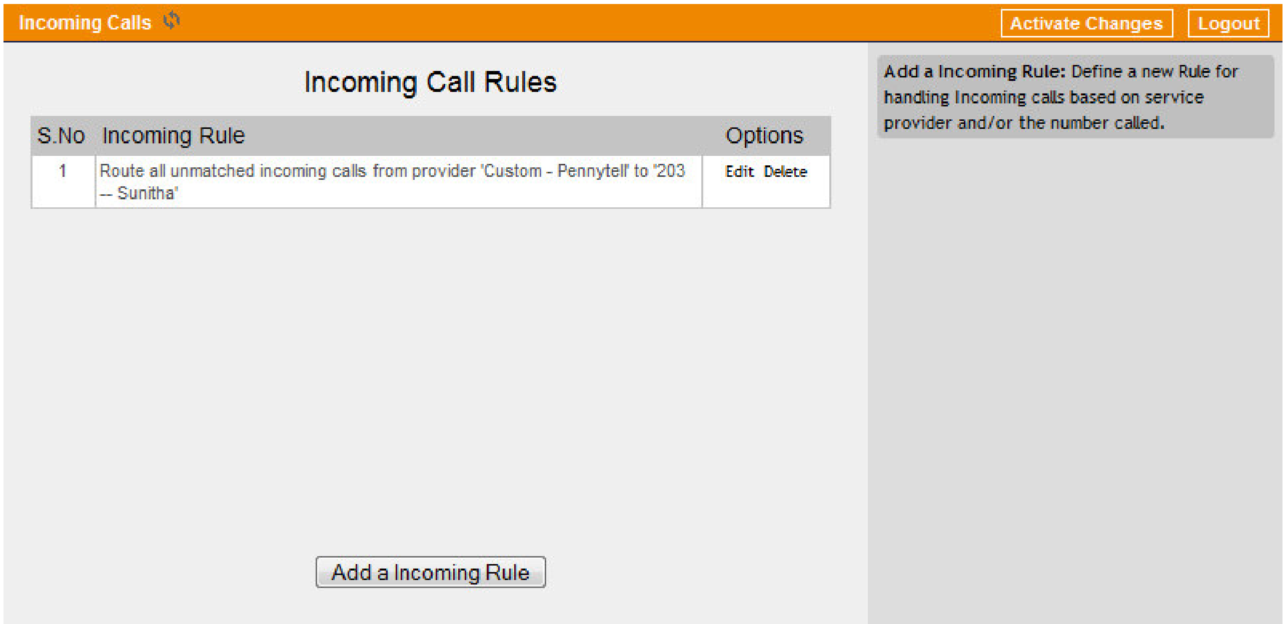
Dial Plan 2:



8.9. Incoming Calling Rules:

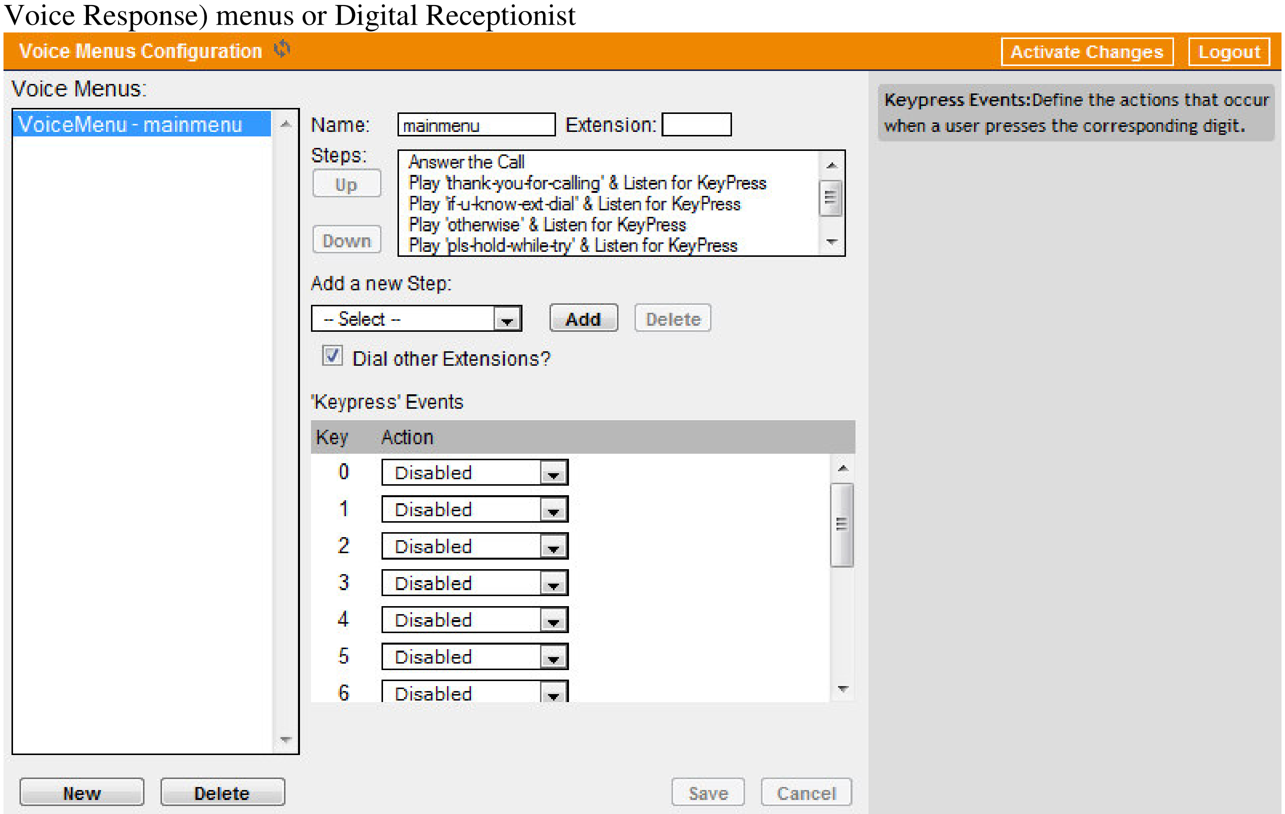
Define how your incoming calls should be handled & configure DID (Direct inward Dialing)

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



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

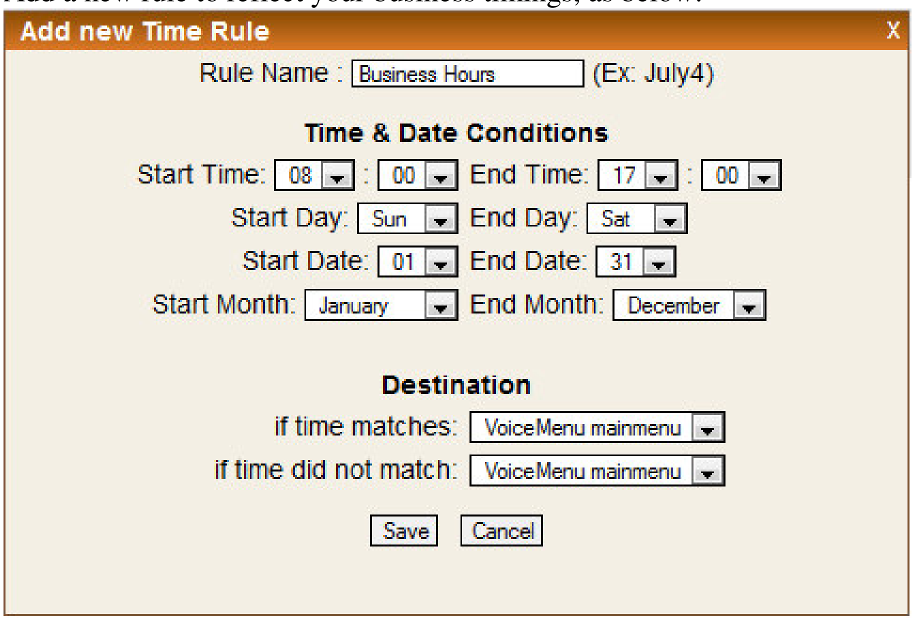


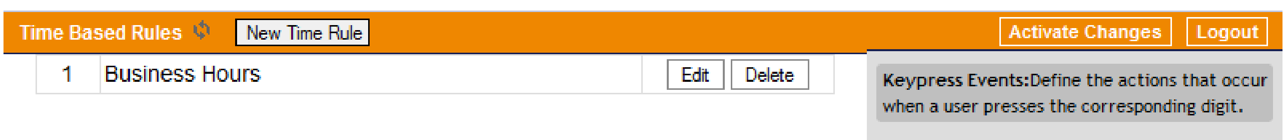
This is the main program setup of AsteriskNOW. How you setup your configuration here affects your whole pbx operations.

8.11. Time Based Rules:

define call routing rules based on date and time

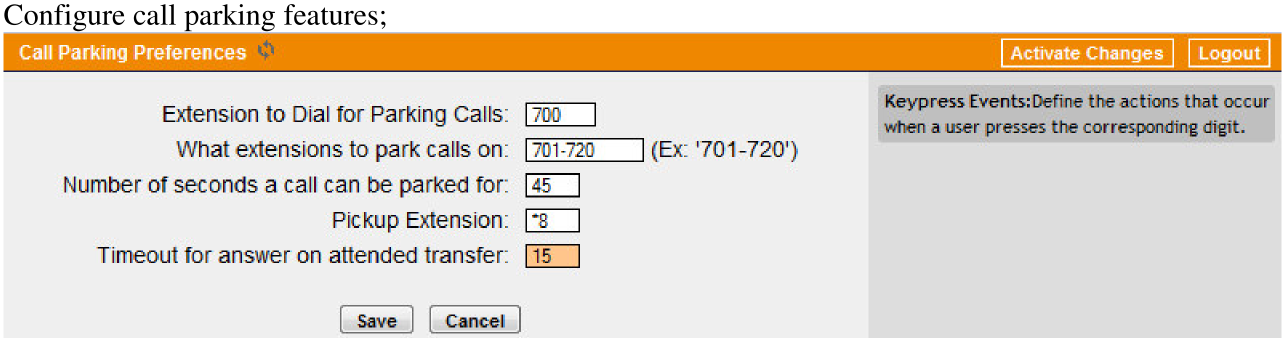
Add a new rule to reflect your business timings, as below:





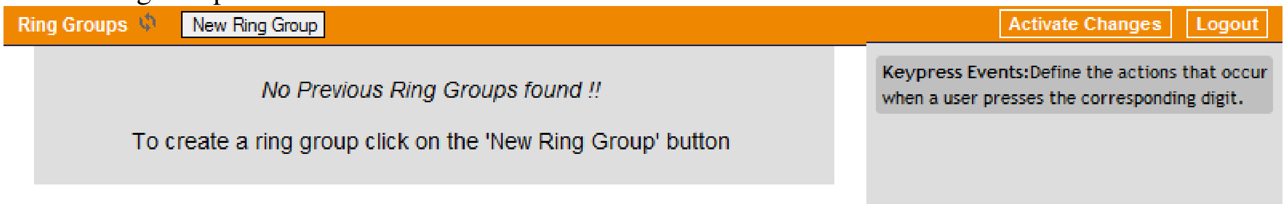
8.12. Call Parking:

Configure call parking features;

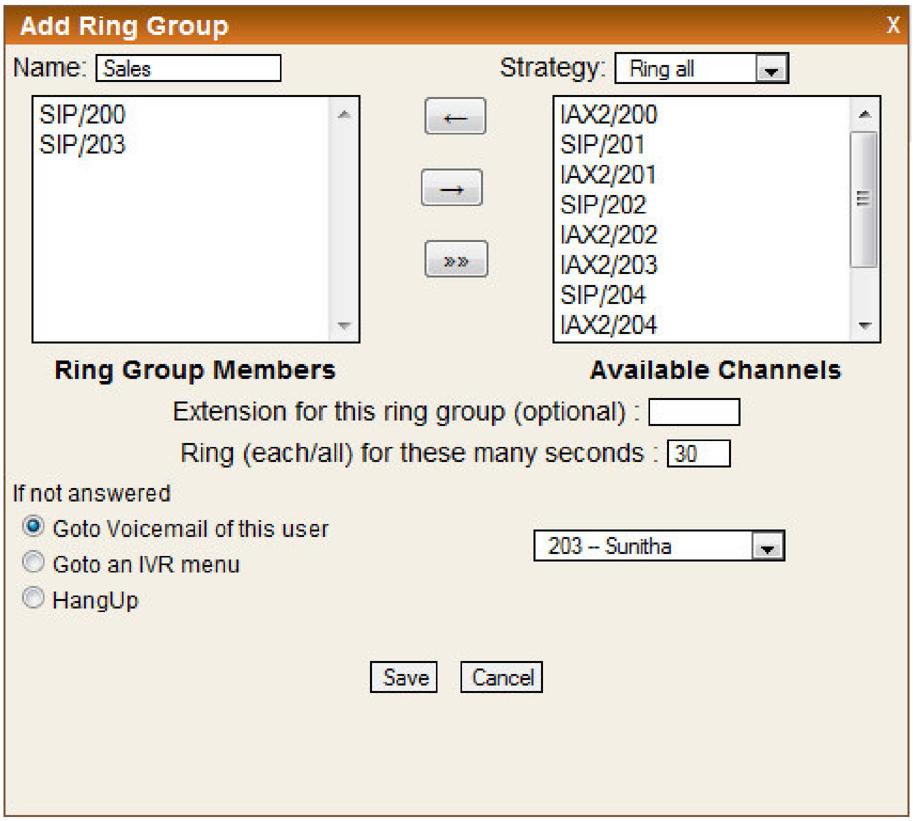


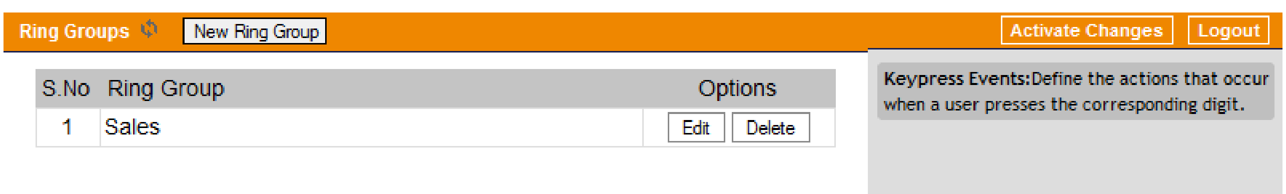
8.13. Ring Groups:

Define RingGroups to dial more than one extension



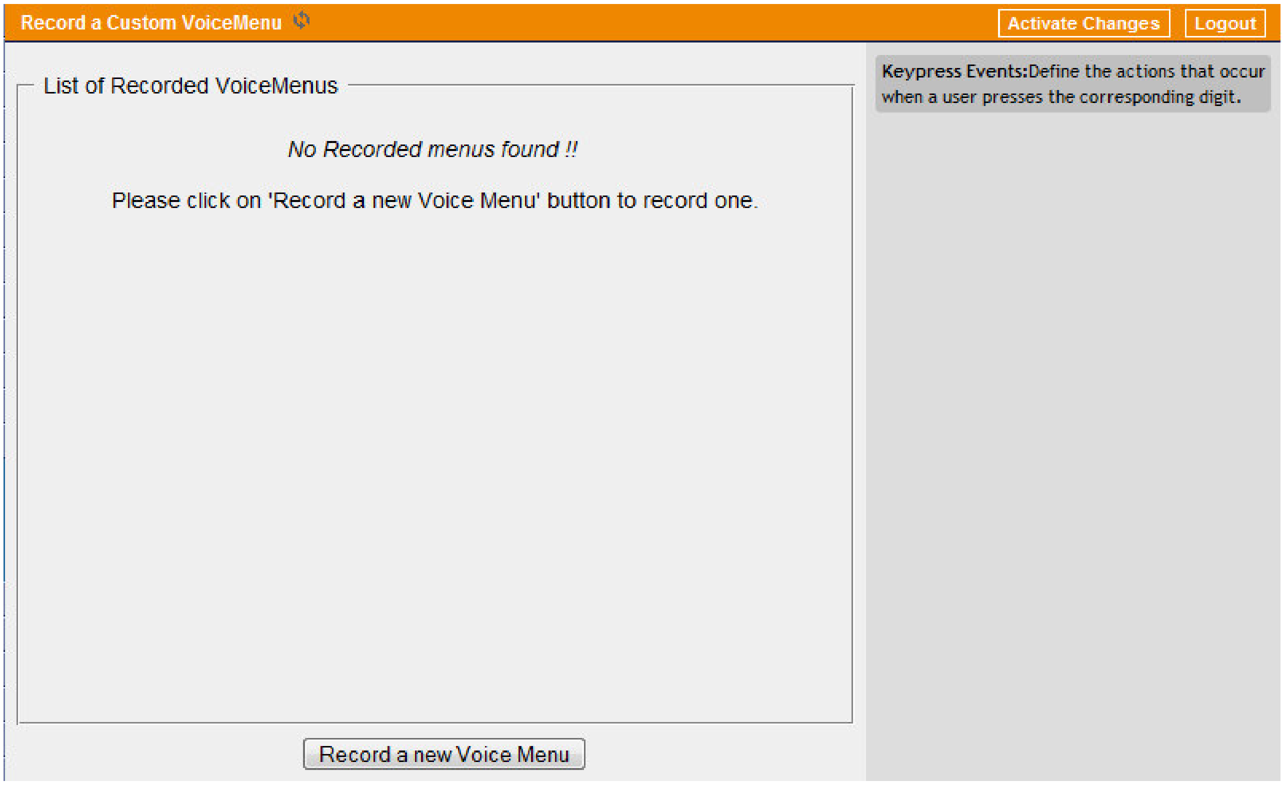
Click on New Ring Group to create your new ring Group.
Do not confuse this with the call queues.

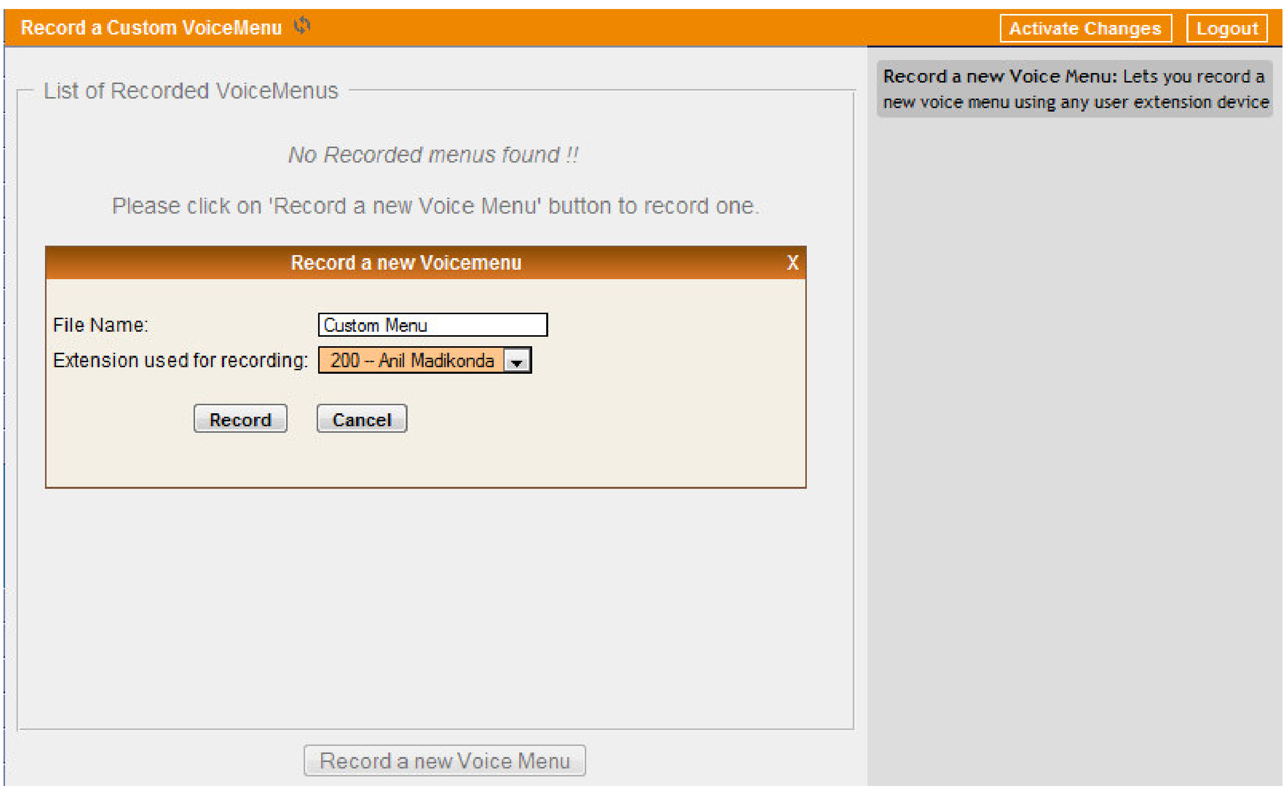


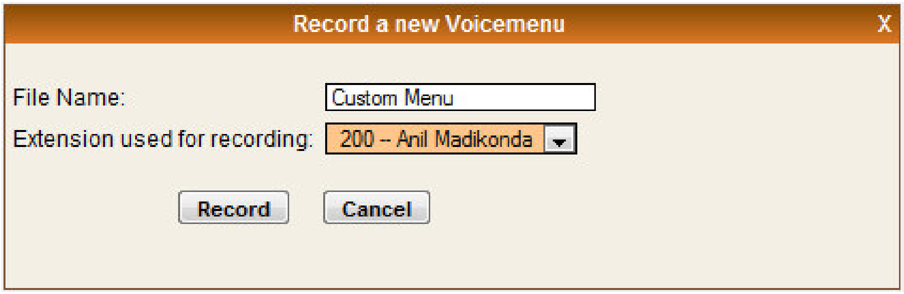


8.14. Record a Menu:

Allows you to record custom voicemenus over a phone


Record a Custom VoiceMenu
Activate Changes
Logout
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
Keypress Events:Define the actions that occur when a user presses the corresponding digit.

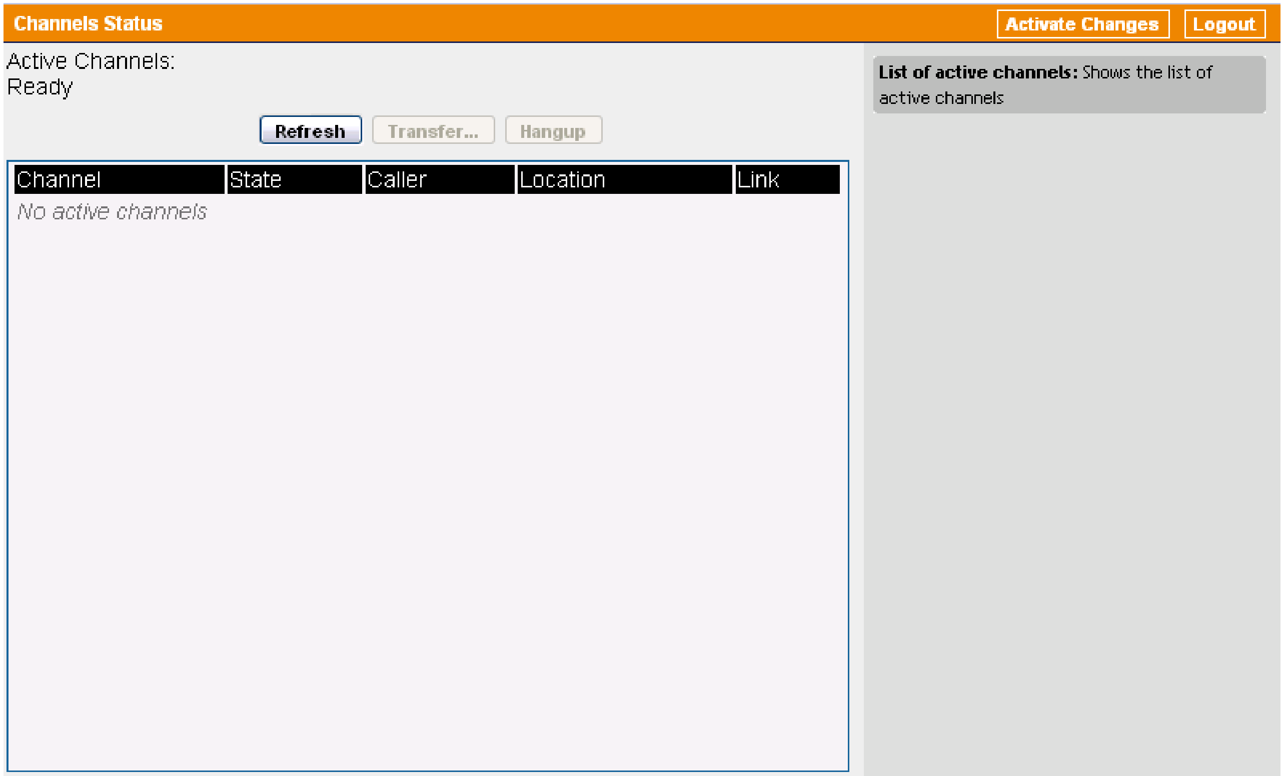

Record a Custom VoiceMenu
Activate Changes
Logout
List of Recorded VoiceMenus
No Recorded menus found !!
Please click on 'Record a new Voice Menu' button to record one.
Record a new Voicemenu
X
File Name:
Custom Menu
Extension used for recording:
200 -- Anil Madikonda
Record
Cancel
Record a new Voice Menu
Record a new Voice Menu: Lets you record a new voice menu using any user extension device



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.

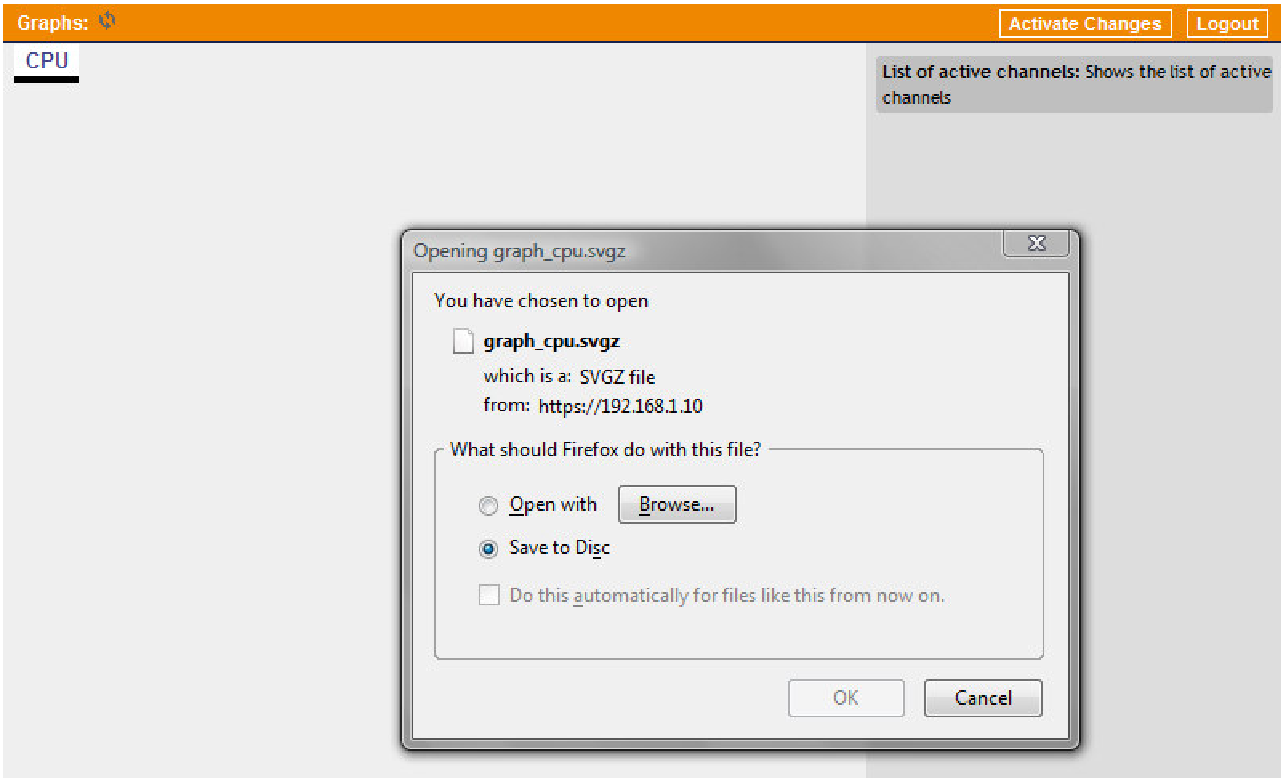
8.15. Active Channels:

Monitor and manage your active calls through this interface.

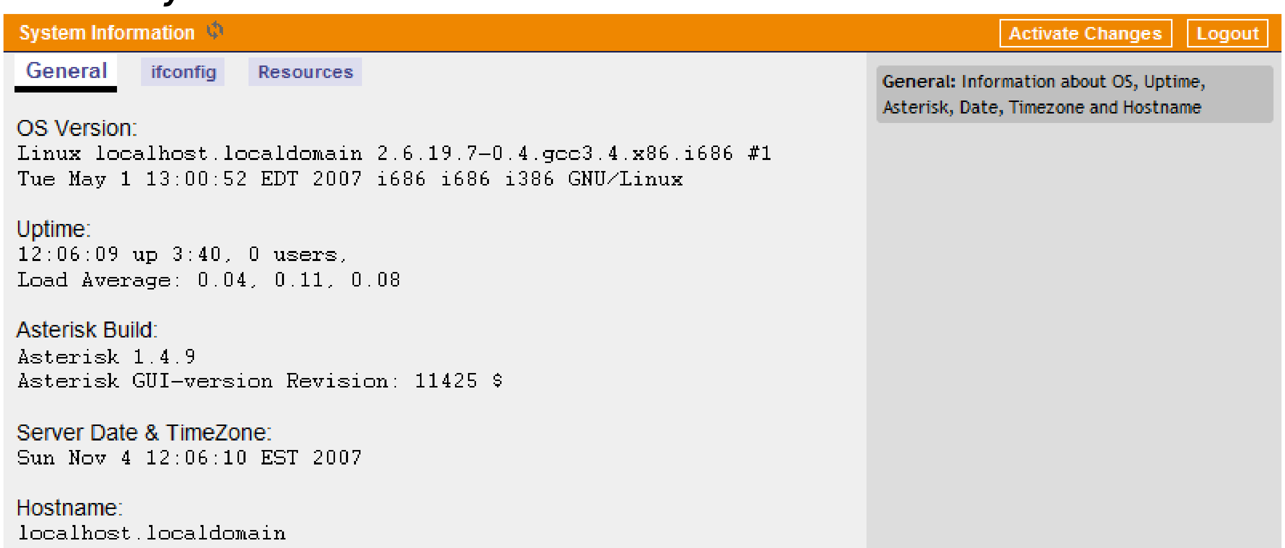


8.16. Graphs:

View Graphs of your System Information.

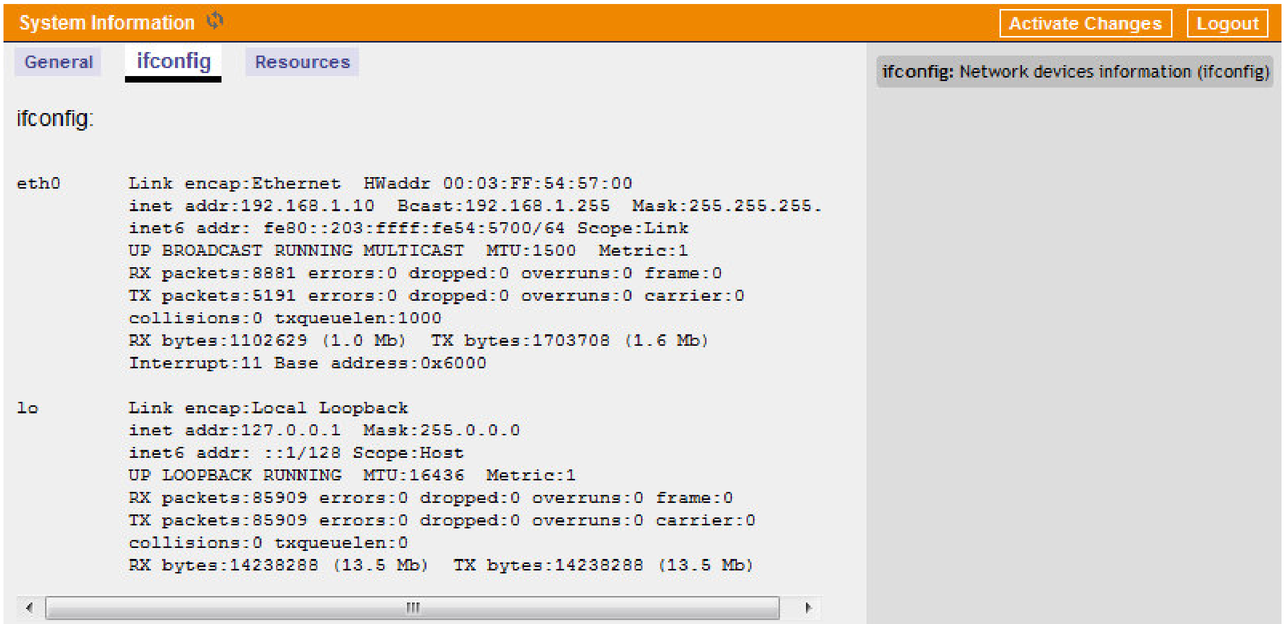


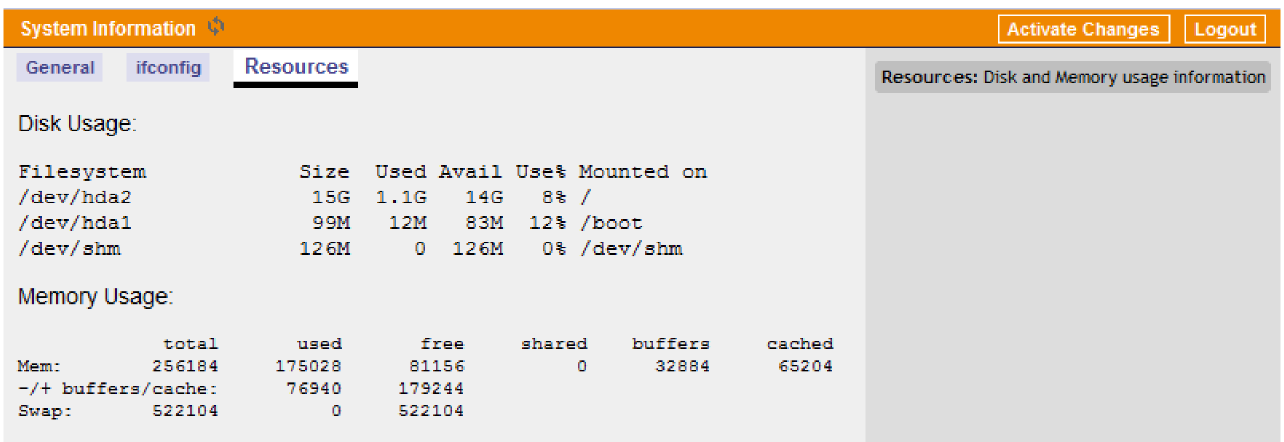
8.17. System Information:



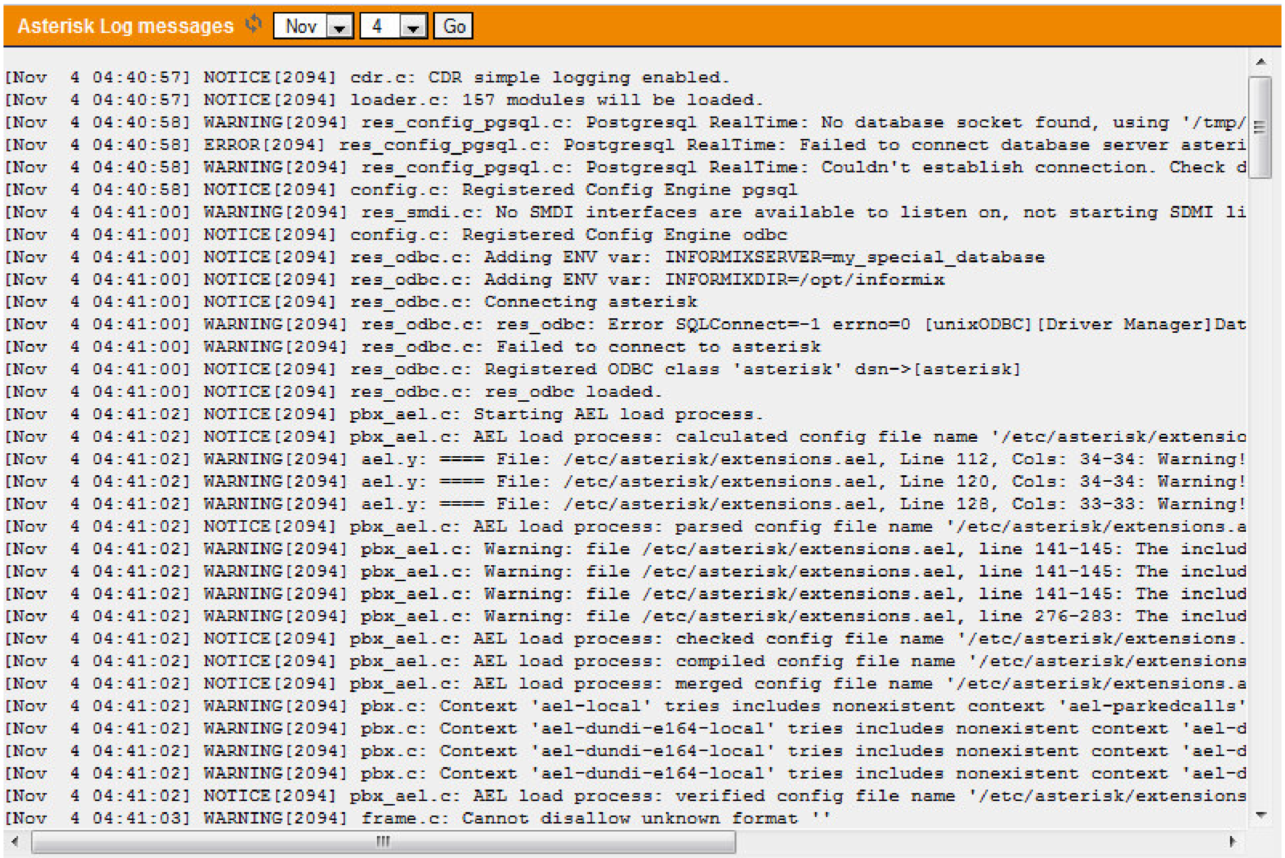
Use this function to:

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





8.18. Asterisk Logs:

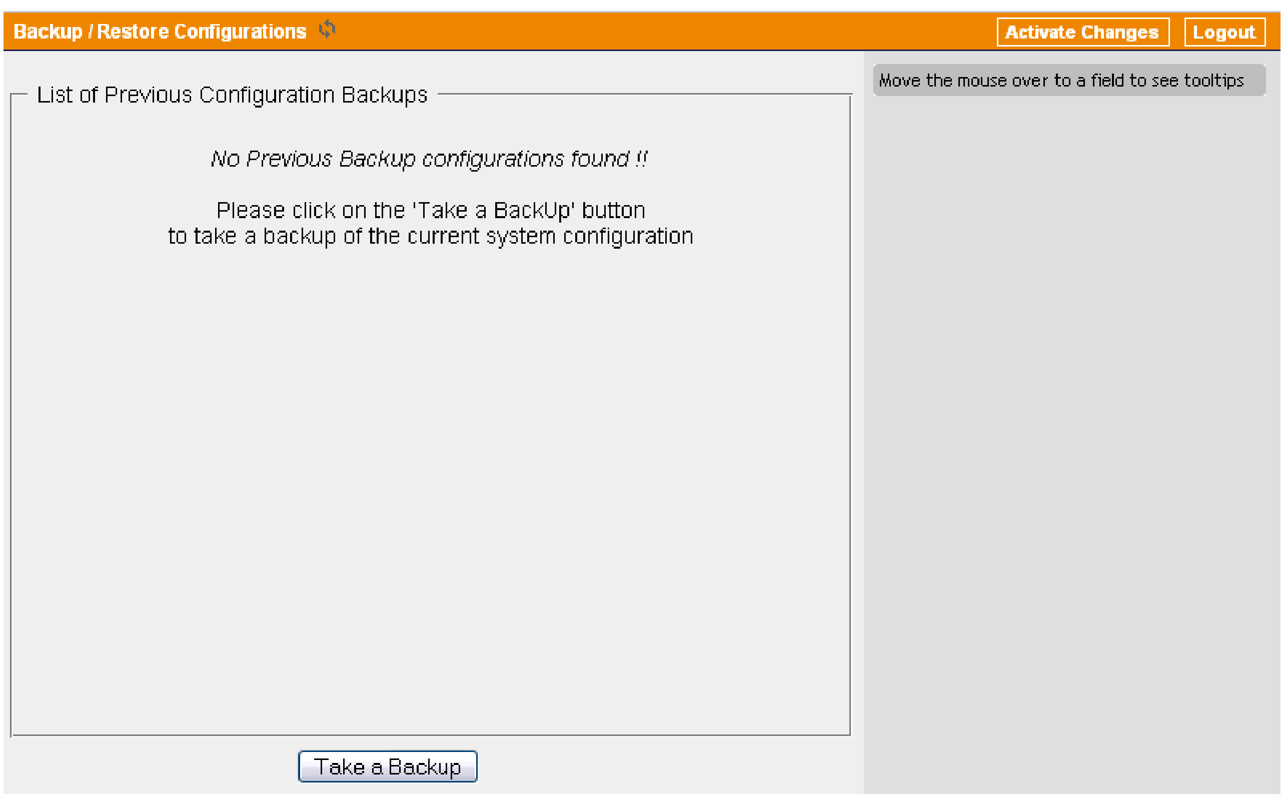


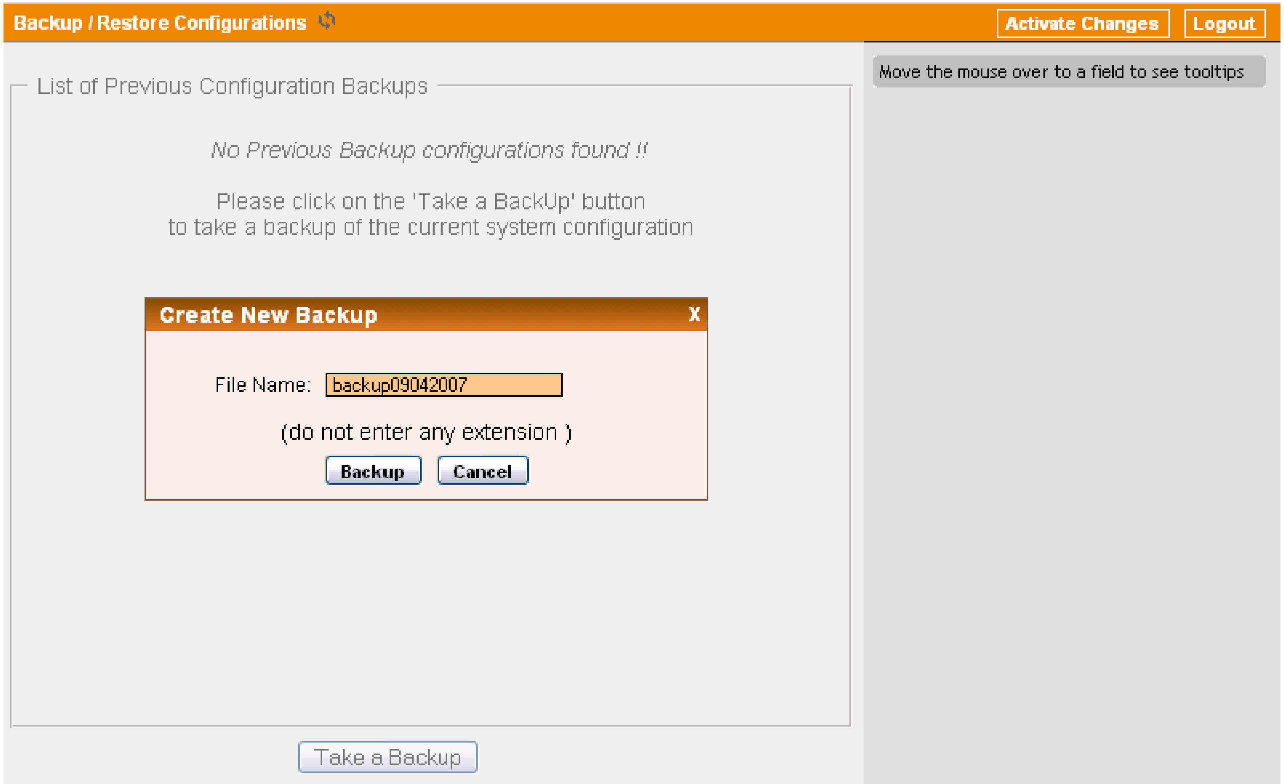
8.19. File Editor:

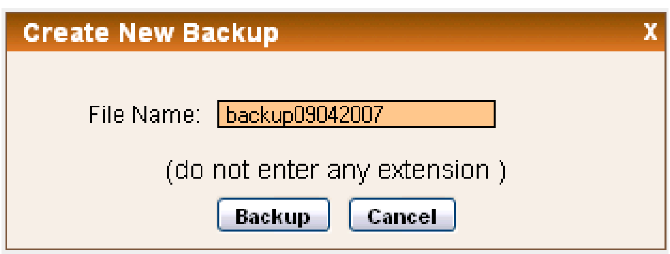
8.20. Asterisk CLI:

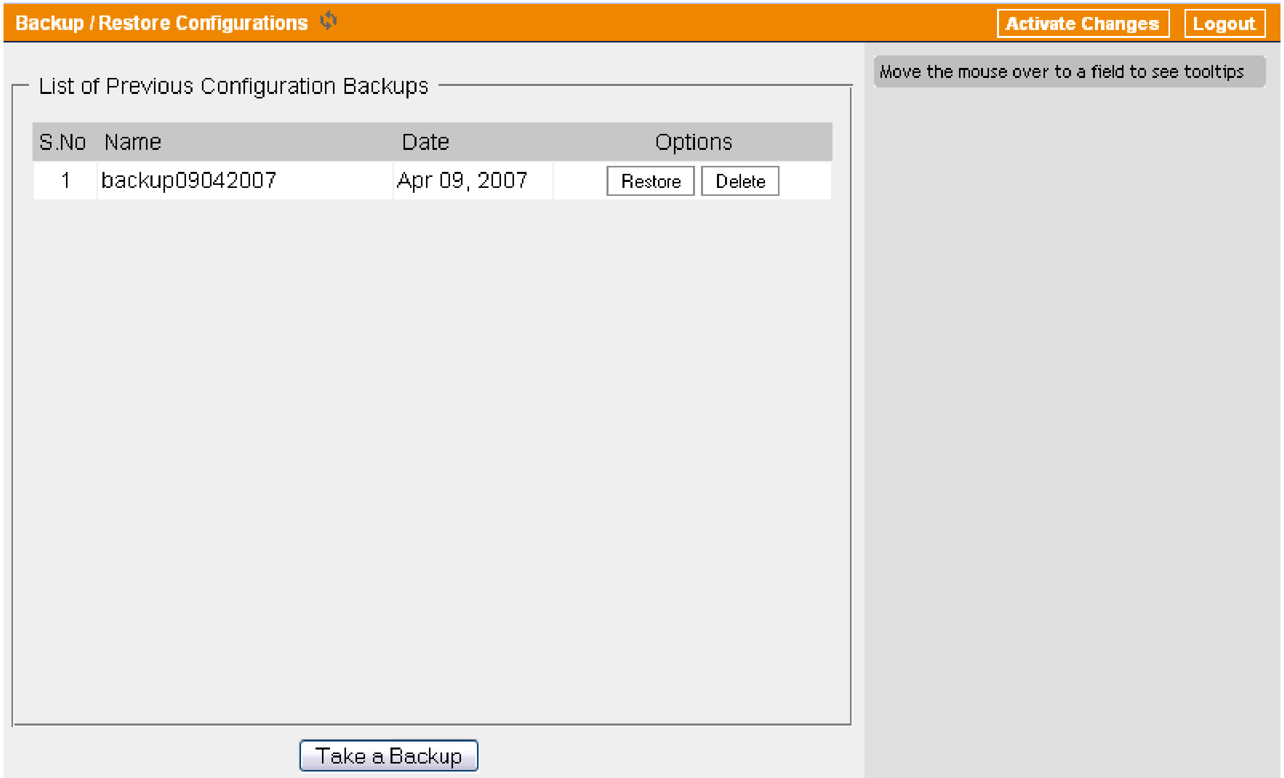

Asterisk Command Line Interface
Command> core show version
Asterisk 1.4.9 built by admin @ aomori on a i686 running Linux on 2007-07-25 21:01:48 UTC
Asterisk CLI>

Backup:

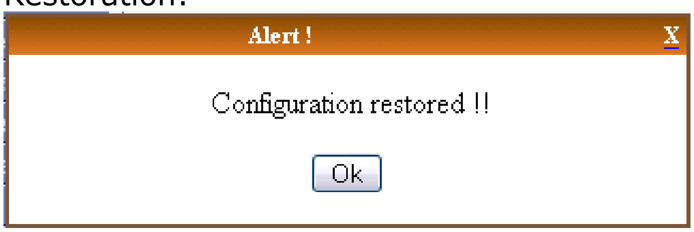

Backup / Restore Configurations
Activate Changes
Logout
List of Previous Configuration Backups
No Previous Backup configurations found !!
Please click on the 'Take a BackUp' button
to take a backup of the current system configuration
Take a Backup
Move the mouse over to a field to see tooltips


Backup / Restore Configurations
Activate Changes
Logout
List of Previous Configuration Backups
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
X
File Name:
backup09042007
(do not enter any extension)
Backup
Cancel
Take a Backup
Move the mouse over to a field to see tooltips

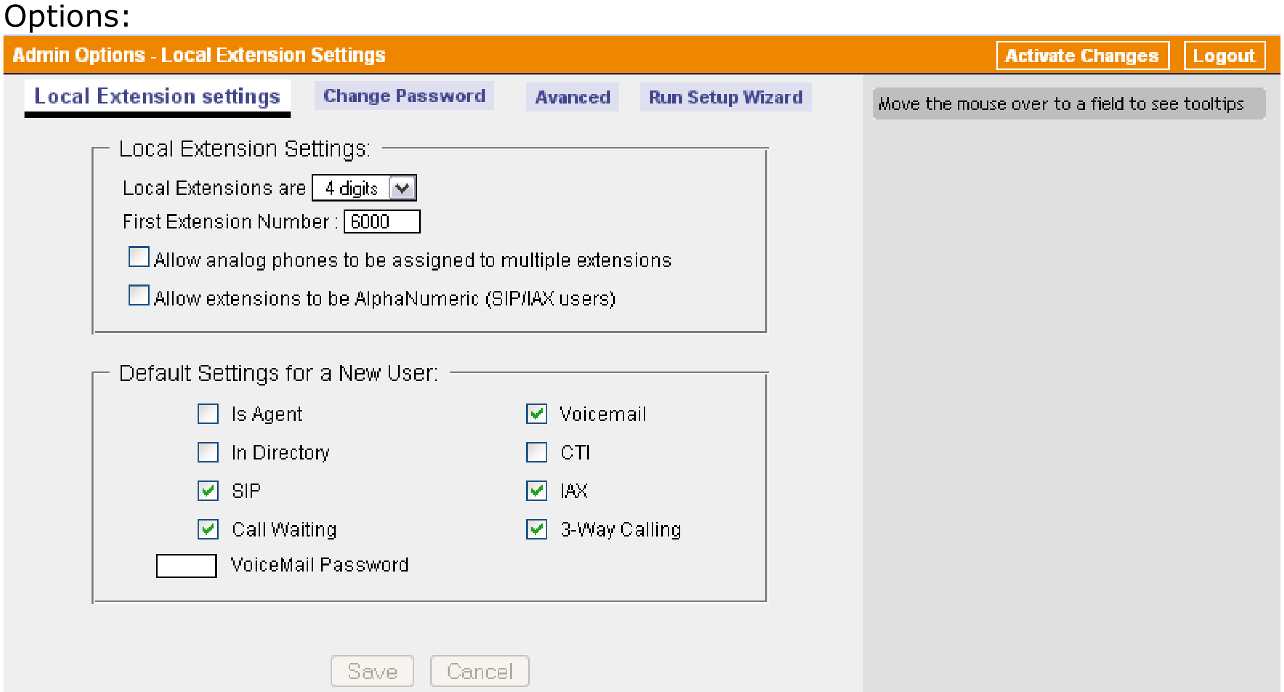

Create New Backup
X
File Name:
backup09042007
(do not enter any extension)
Backup
Cancel

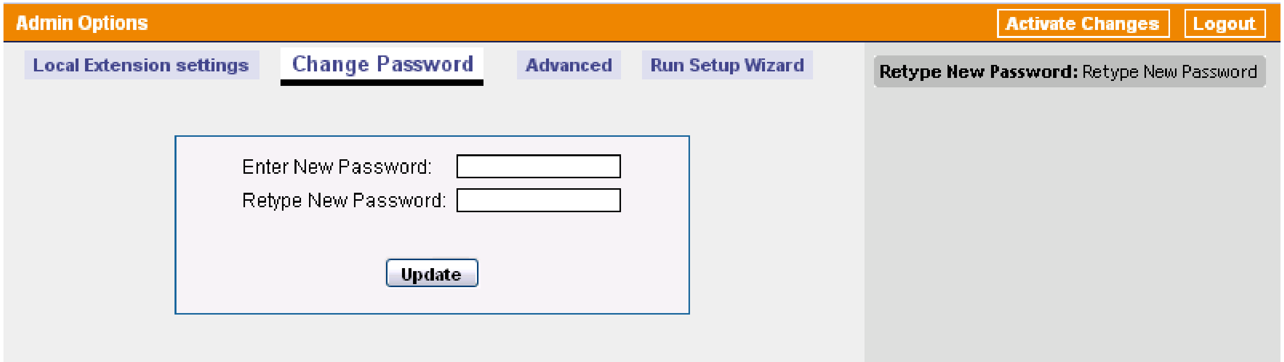


Restoration:

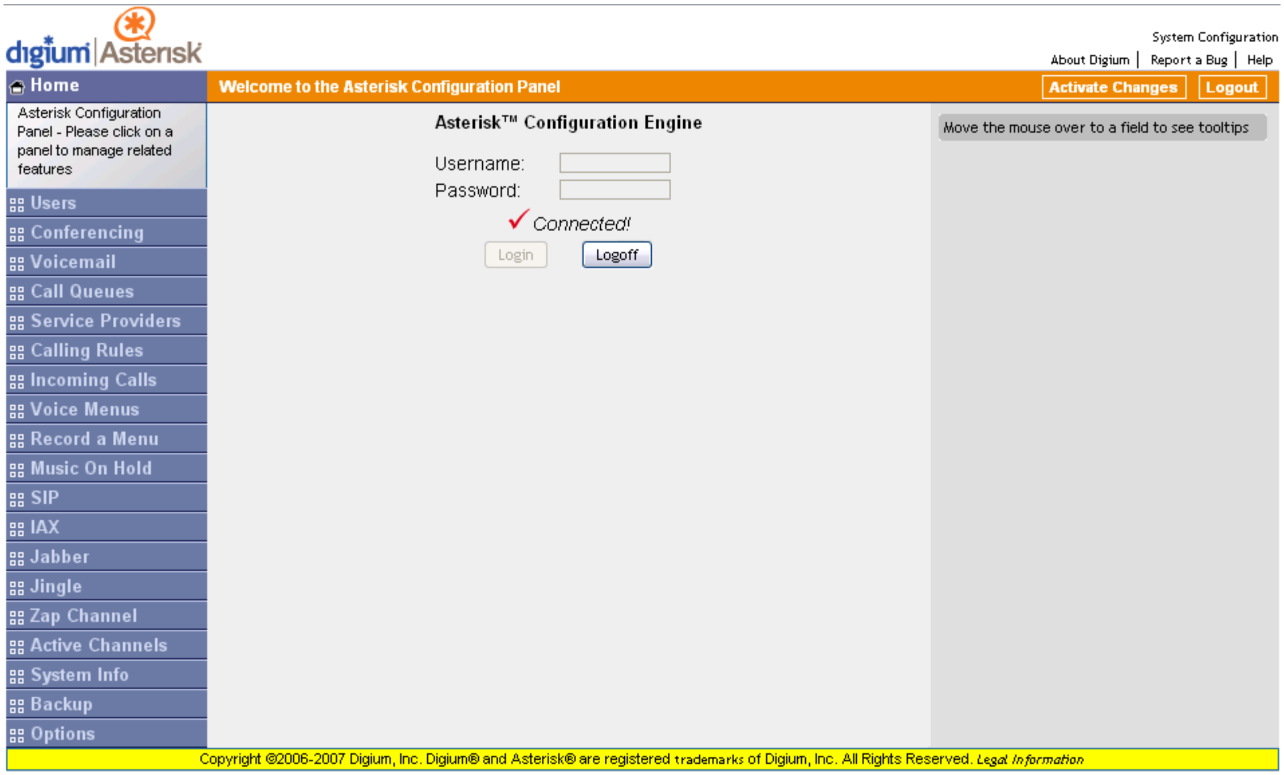


Options:



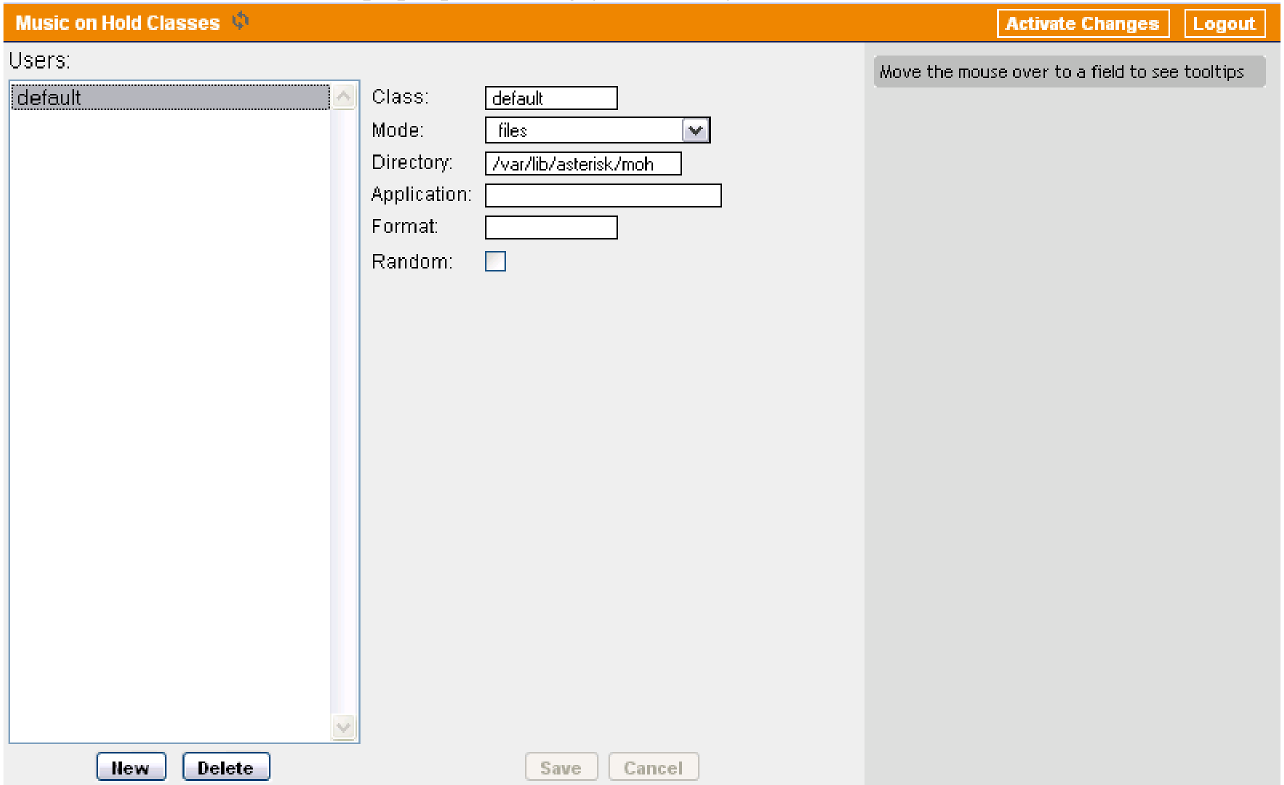


The Advanced Menu:



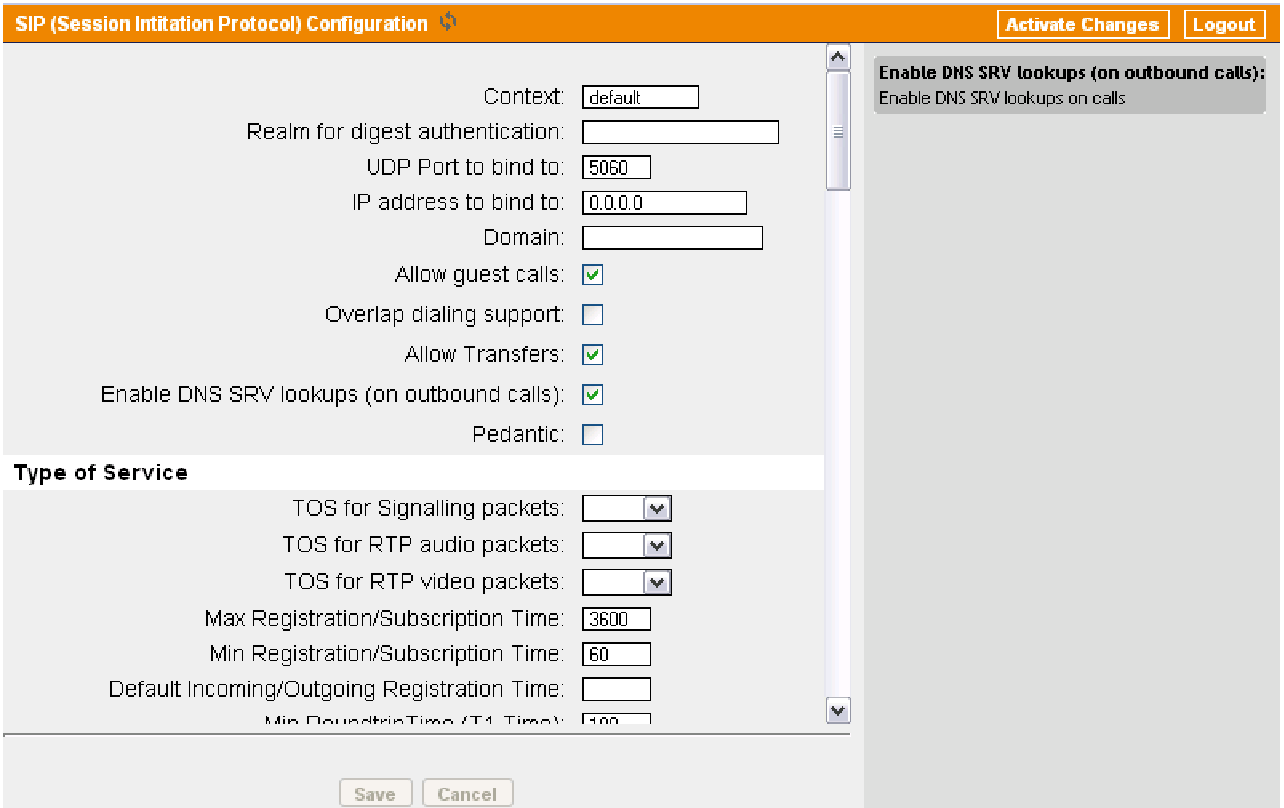
Music On Hold:

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



SIP:

SIP (Session Initiation Protocol) Configuration



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/outoing 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

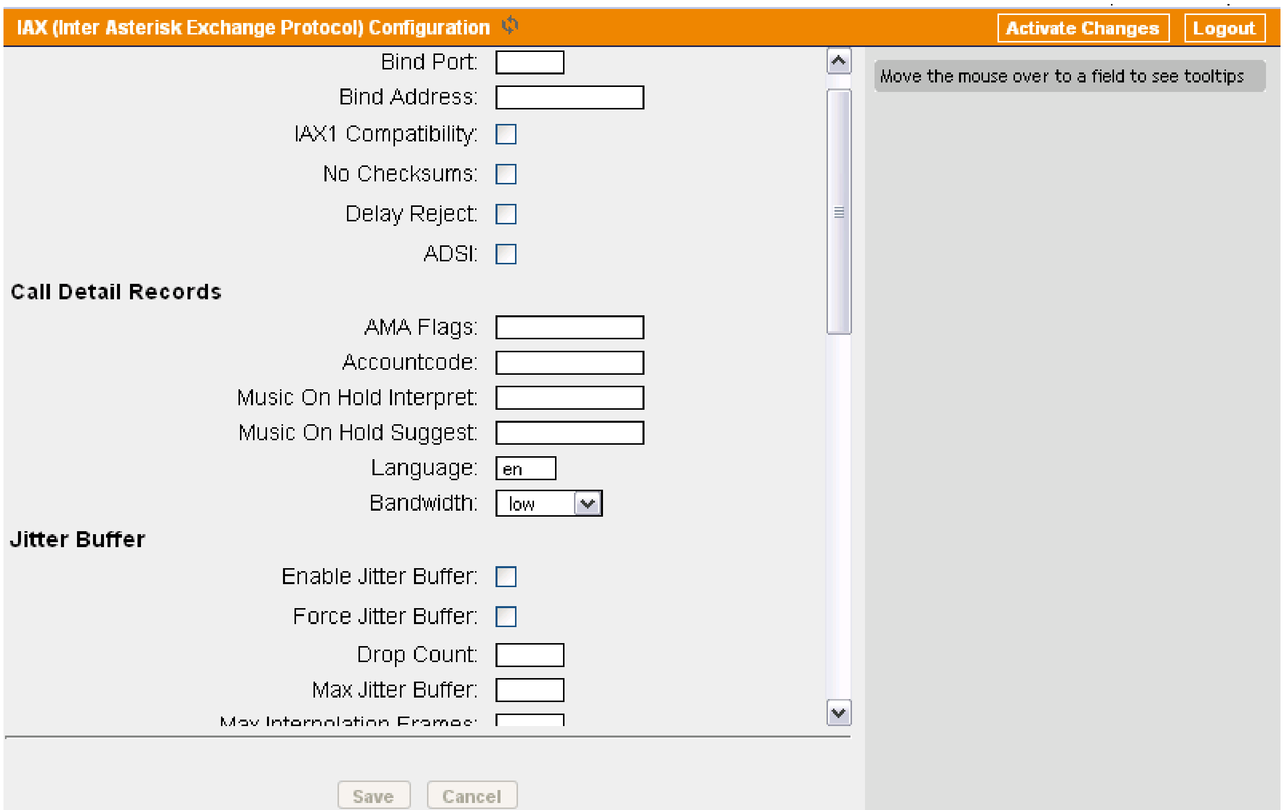
Outbound SIP Registrations

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

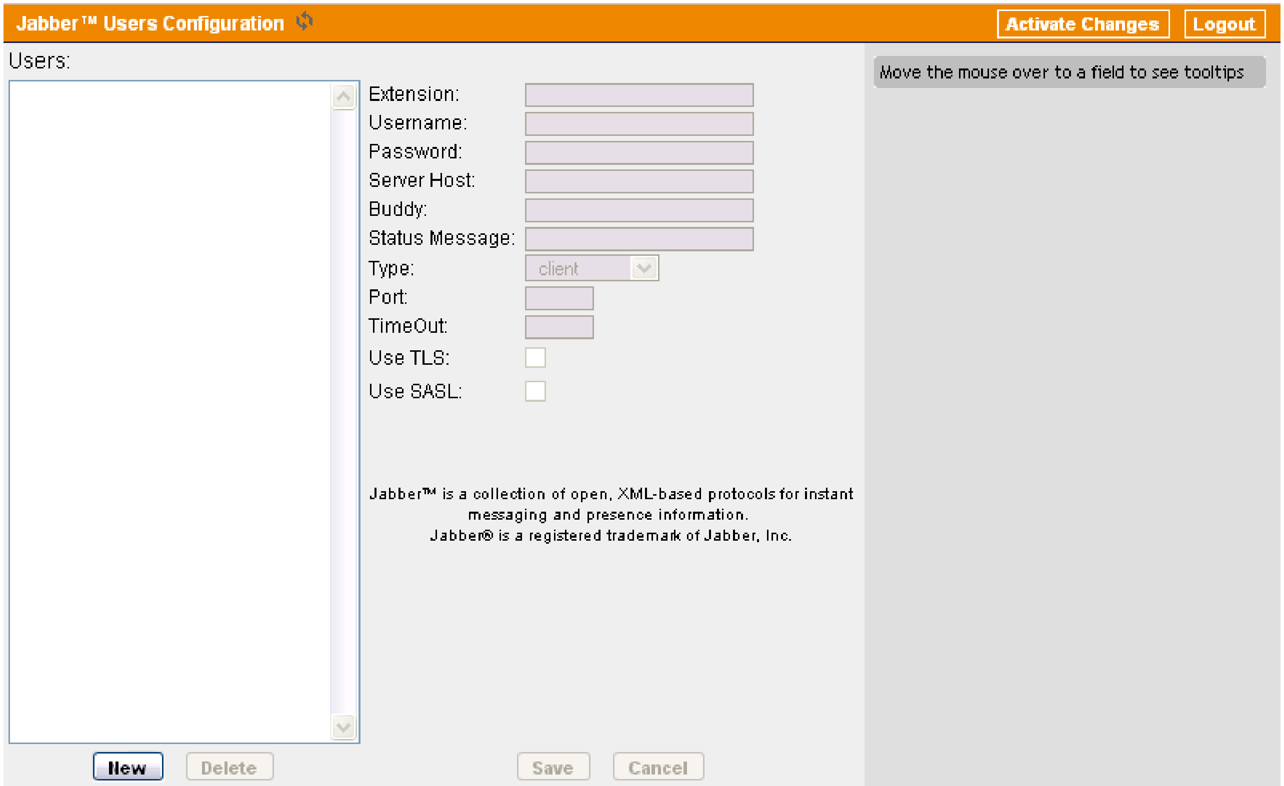
NAT Support

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:

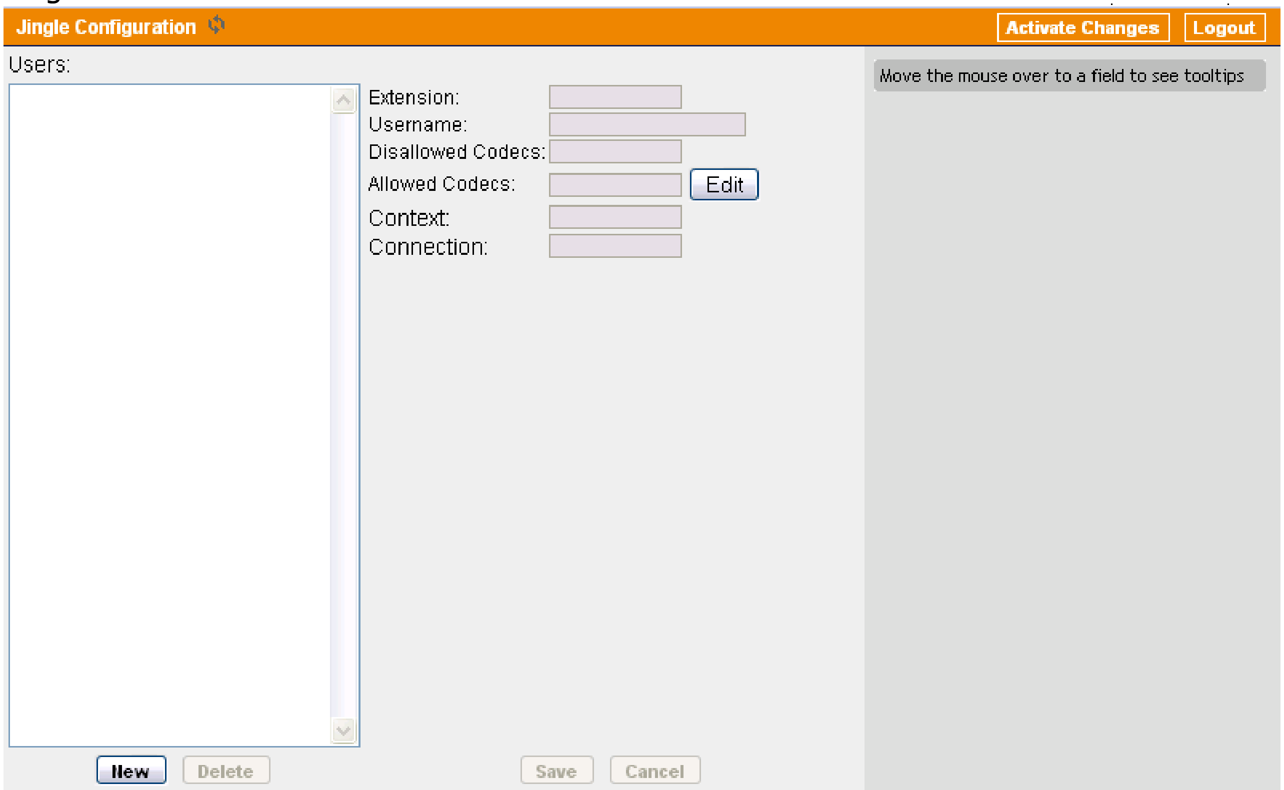

IAX (Inter Asterisk Exchange Protocol) Configuration
Activate Changes
Logout
Bind Port:
Bind Address:
IAX1 Compatibility:
No Checksums:
Delay Reject:
ADSI:
Call Detail Records
AMA Flags:
Accountcode:
Music On Hold Interpret:
Music On Hold Suggest:
Language:
en
Bandwidth:
low
Jitter Buffer
Enable Jitter Buffer:
Force Jitter Buffer:
Drop Count:
Max Jitter Buffer:
Save
Cancel
Move the mouse over to a field to see tooltips

Jabber:

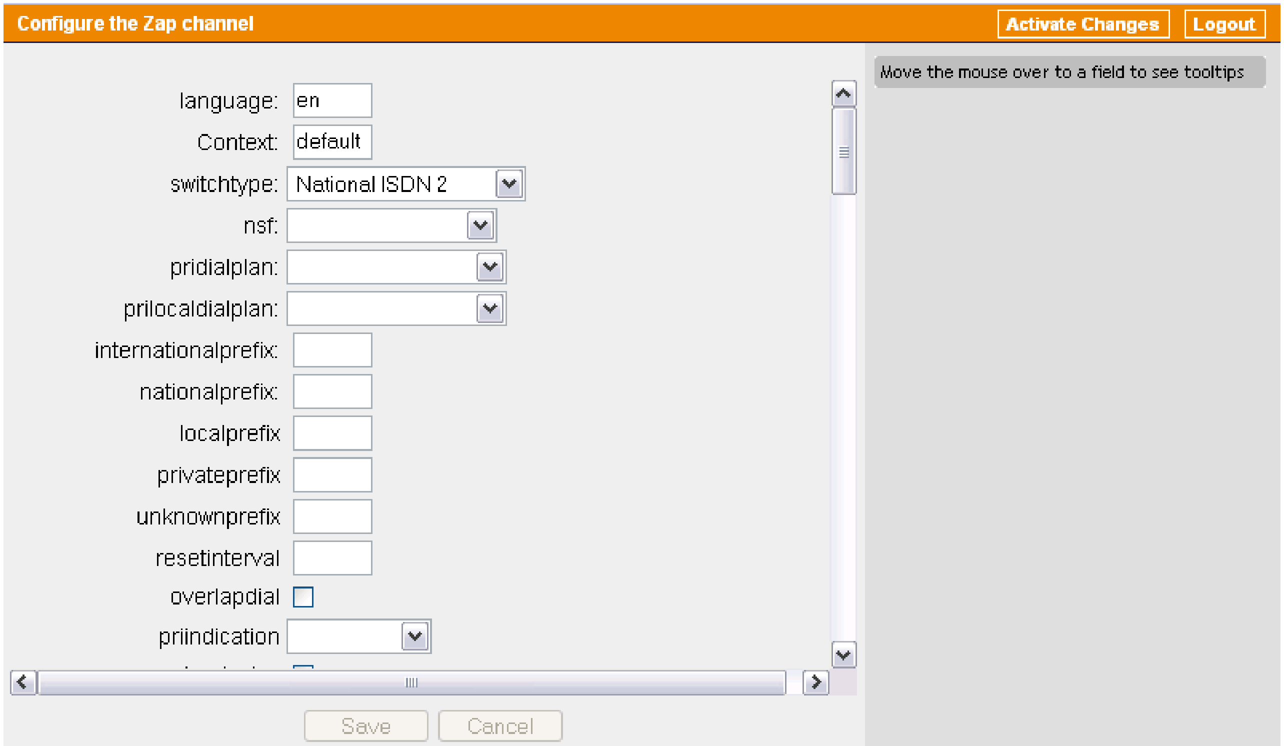


Jabber™ Users Configuration
Activate Changes
Logout
Users:
Extension:
Username:
Password:
Server Host:
Buddy:
Status Message:
Type:
client
Port:
TimeOut:
Use TLS:
Use SASL:
Jabber™ is a collection of open, XML-based protocols for instant messaging and presence information.
Jabber® is a registered trademark of Jabber, Inc.
Move the mouse over to a field to see tooltips
New
Delete
Save
Cancel

Jingle:

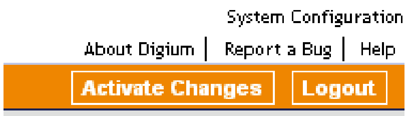

Jingle Configuration
Activate Changes
Logout
Users:
Extension:
Username:
Disallowed Codecs:
Allowed Codecs:
Edit
Context:
Connection:
Move the mouse over to a field to see tooltips
New
Delete
Save
Cancel

Zap Channels:

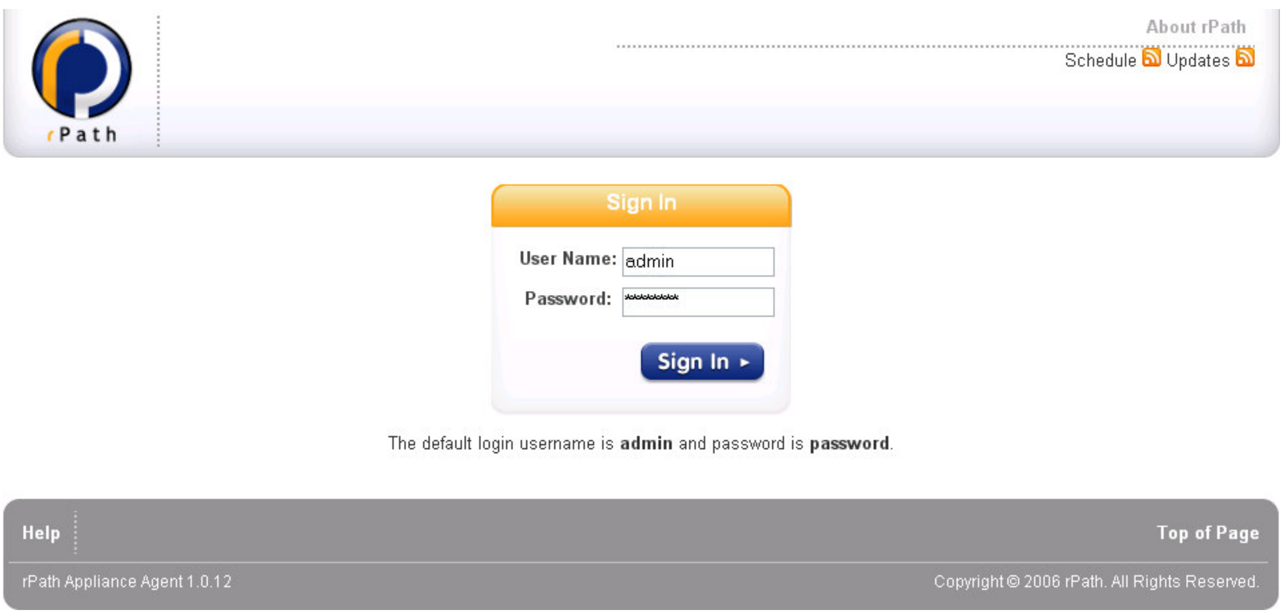


9. System Setup & Administration:

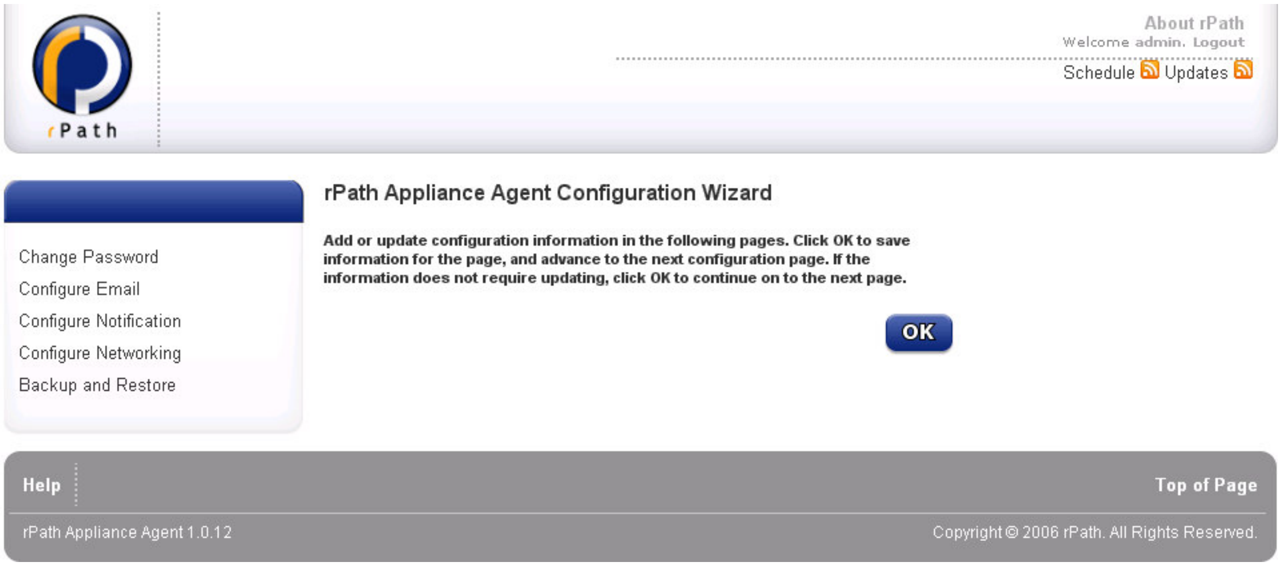
To enter into the system administration menu. Click on the System configuration link at the top right hand corner of the webpage.



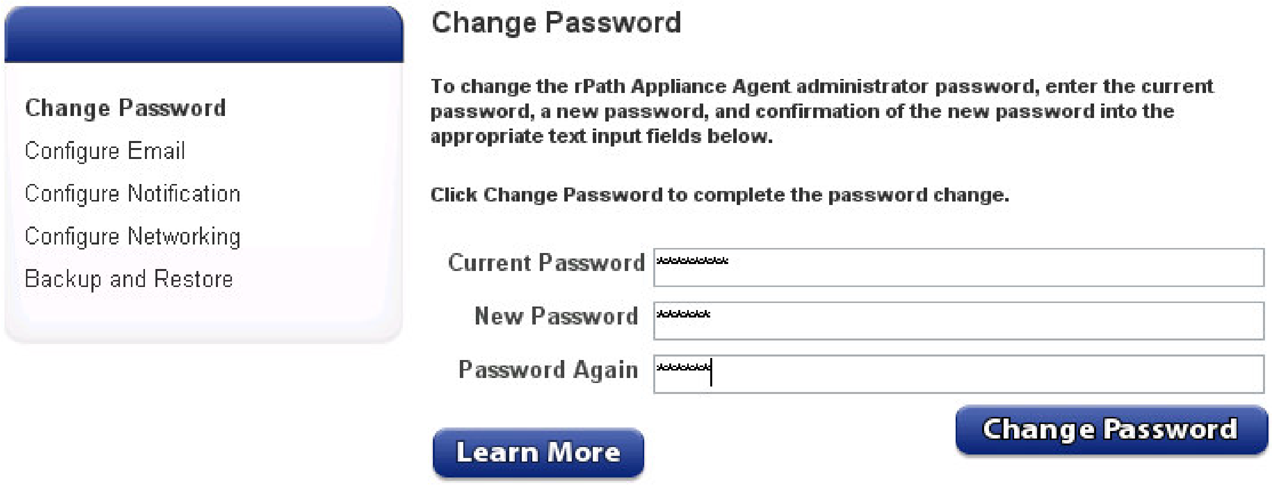
You will then get another popup screen as shown below.



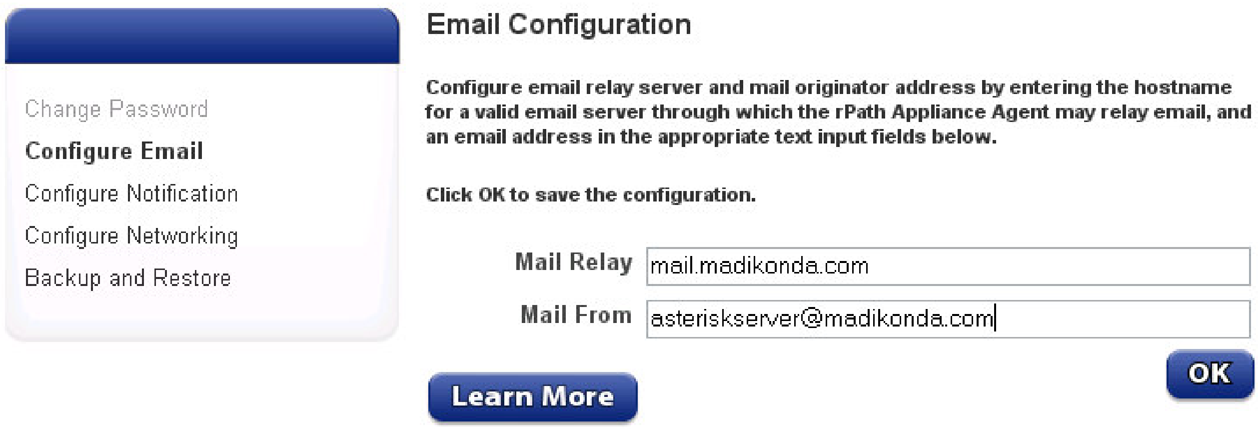
The login use the default username of **admin** and the password being **password**


rPath
About rPath
Welcome admin. Logout
Schedule
Updates
rPath Appliance Agent Configuration Wizard
Change Password
Configure Email
Configure Notification
Configure Networking
Backup and Restore
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.

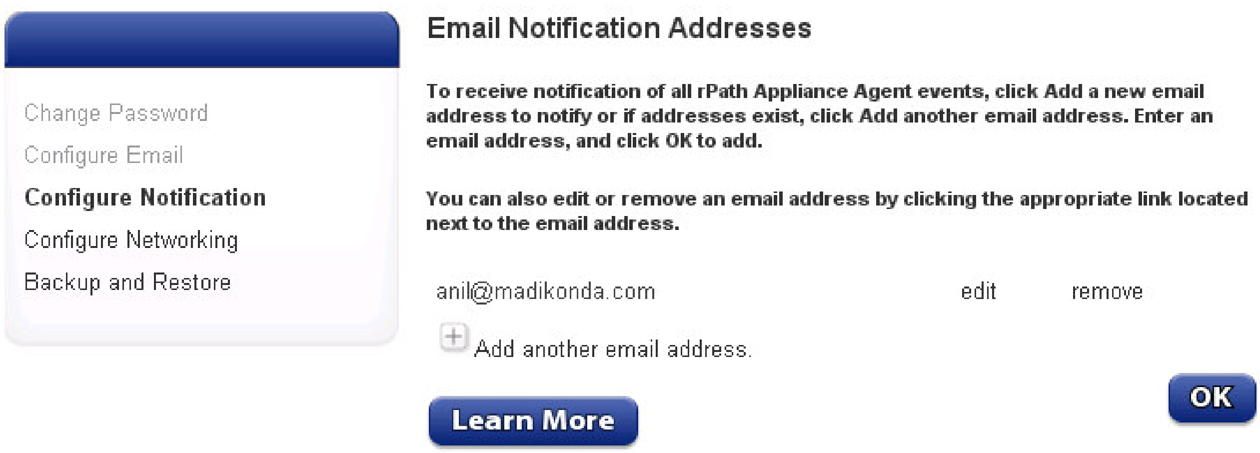
9.1. Change Password:



9.2. Email Configuration:



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

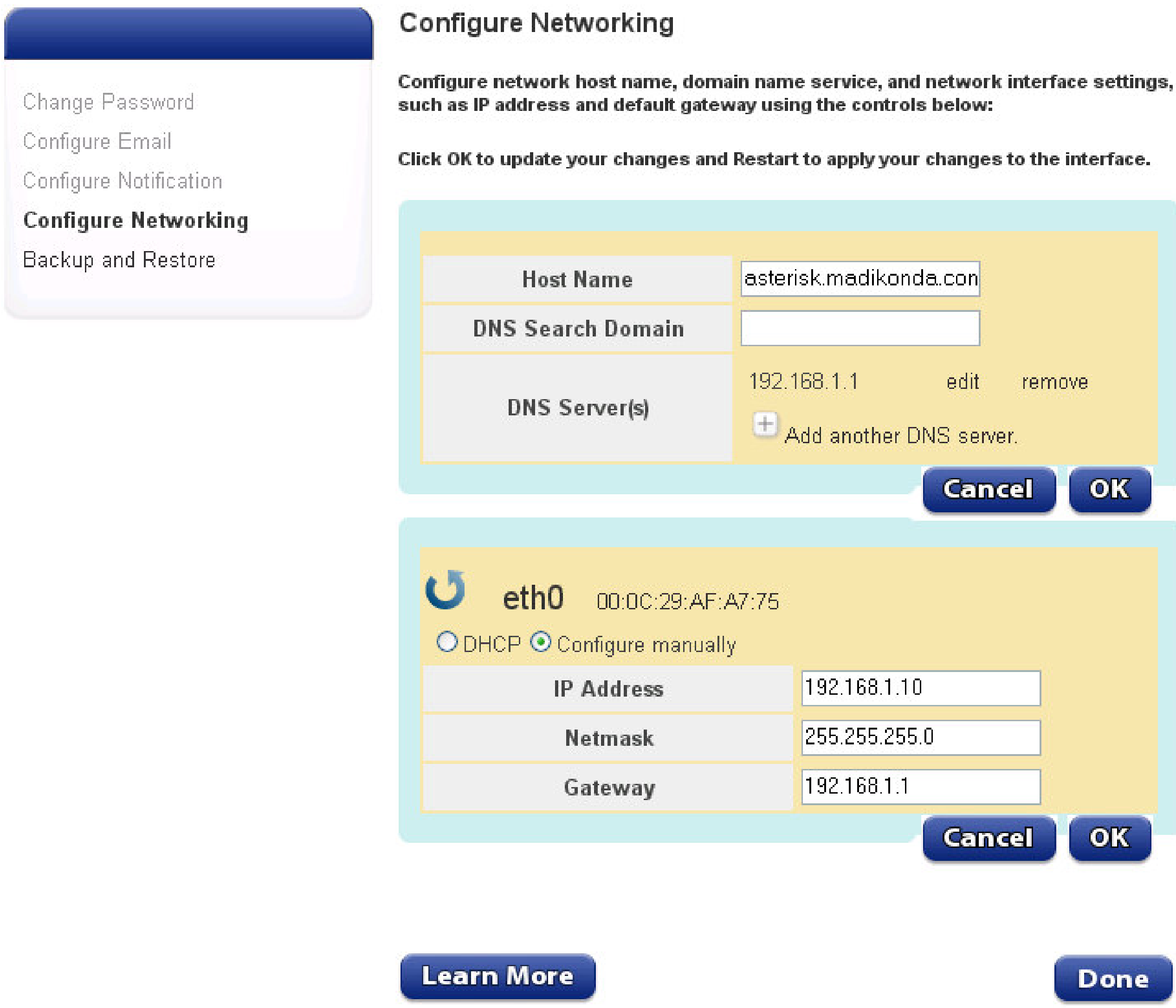


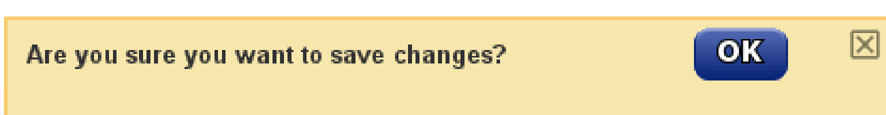
- Here you need to enter at least one email address to which all system notifications will be sent to. This should be the email of your system administrator.

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.

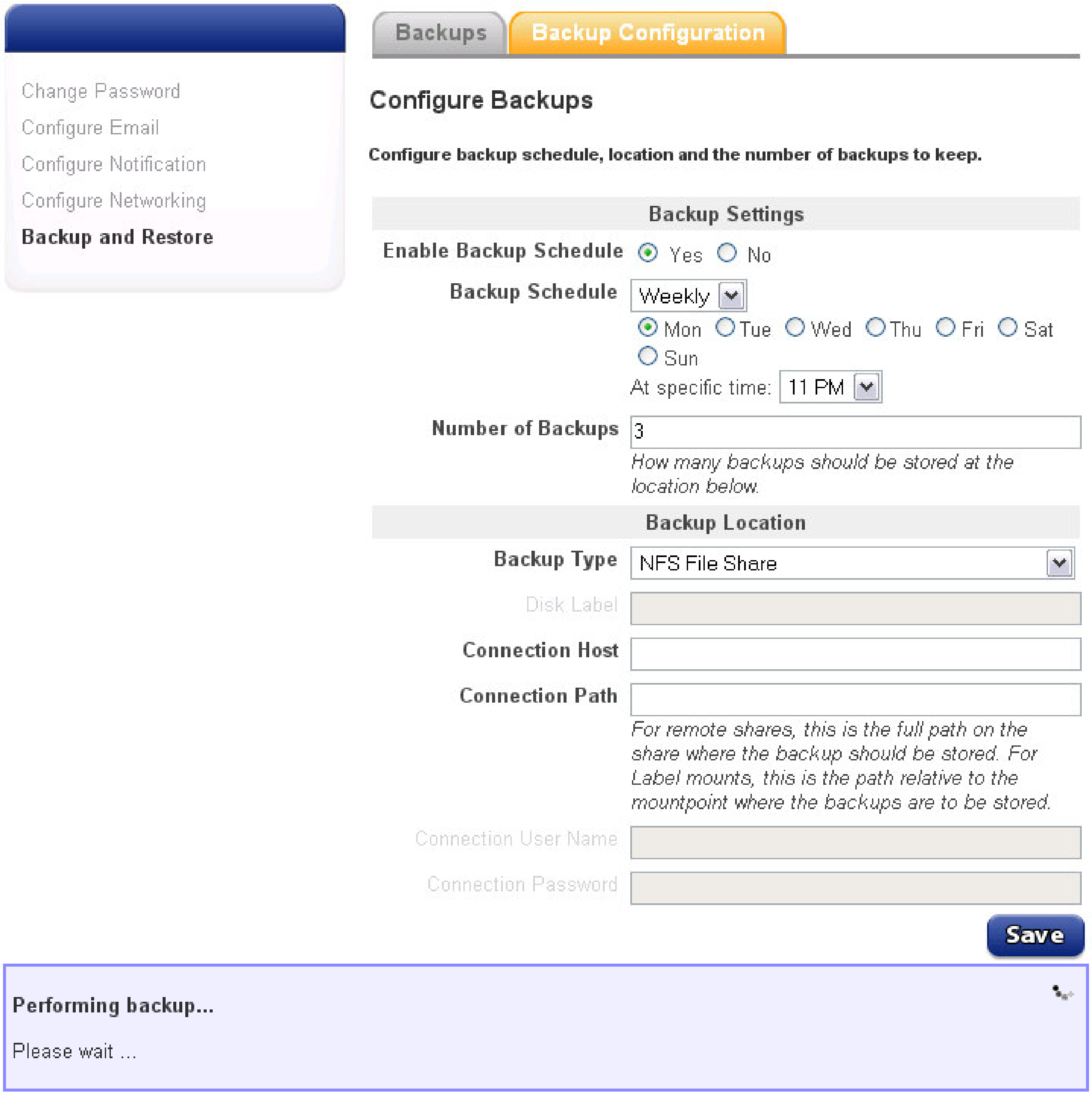
Generally if you intend to use it in a Home environment you can leave it to use the dhcp method. If you need to setup a static Ip address then be sure to enter all the required fields. Then don’t forget to reboot.





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

Backup:



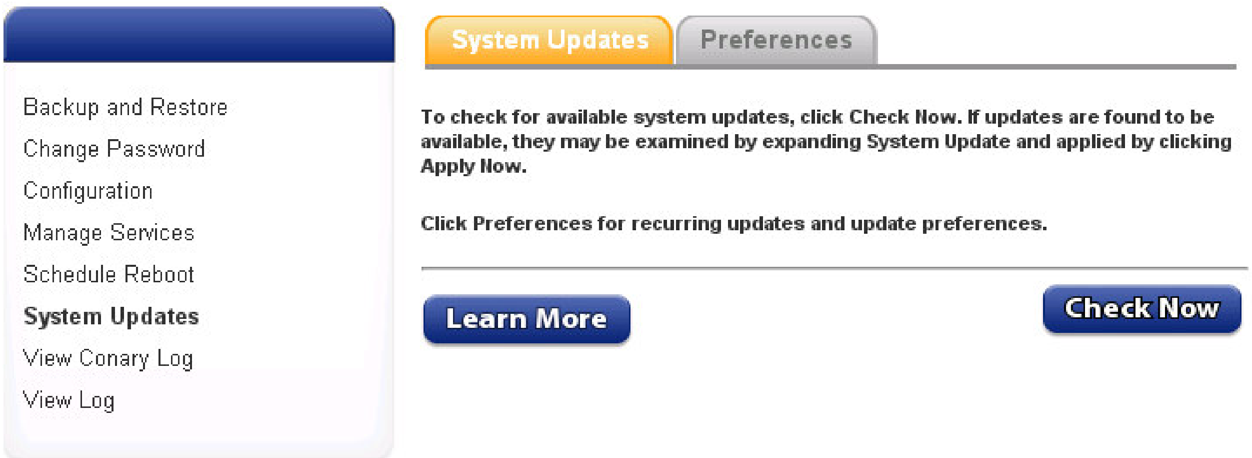

Backup and Restore
Change Password
Configuration
Manage Services
Schedule Reboot
System Updates
View Conary Log
View Log
rPath Appliance Agent
System information:
Total memory: 250 MB
Memory free: 15 MB
Total swap: 258 MB
Swap free: 258 MB
1 CPU(s):
CPU model: Intel(R) Pentium(R) M processor 1.70GHz
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:

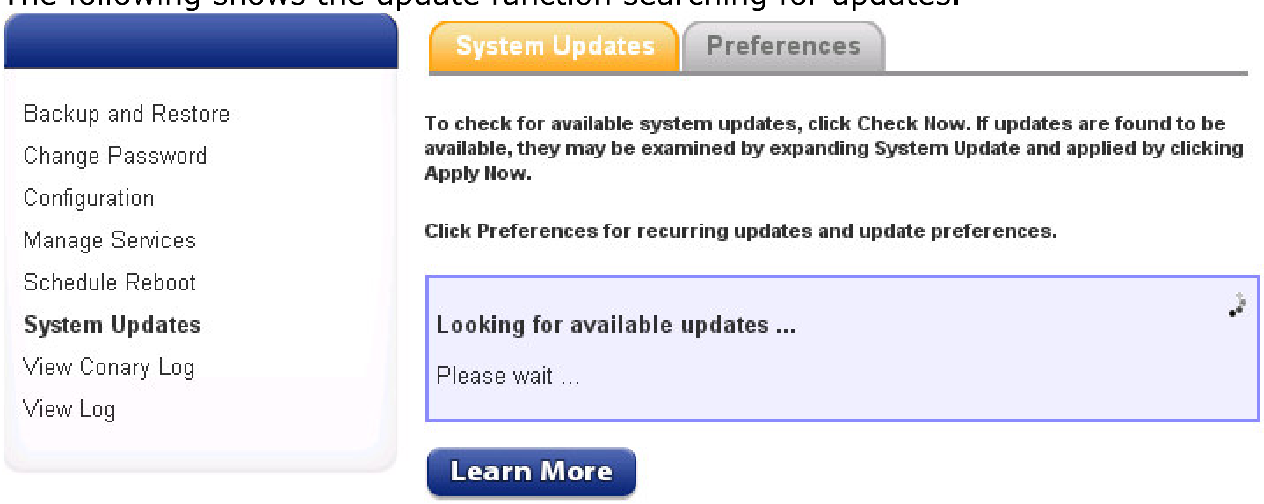
9.5. System Updates:

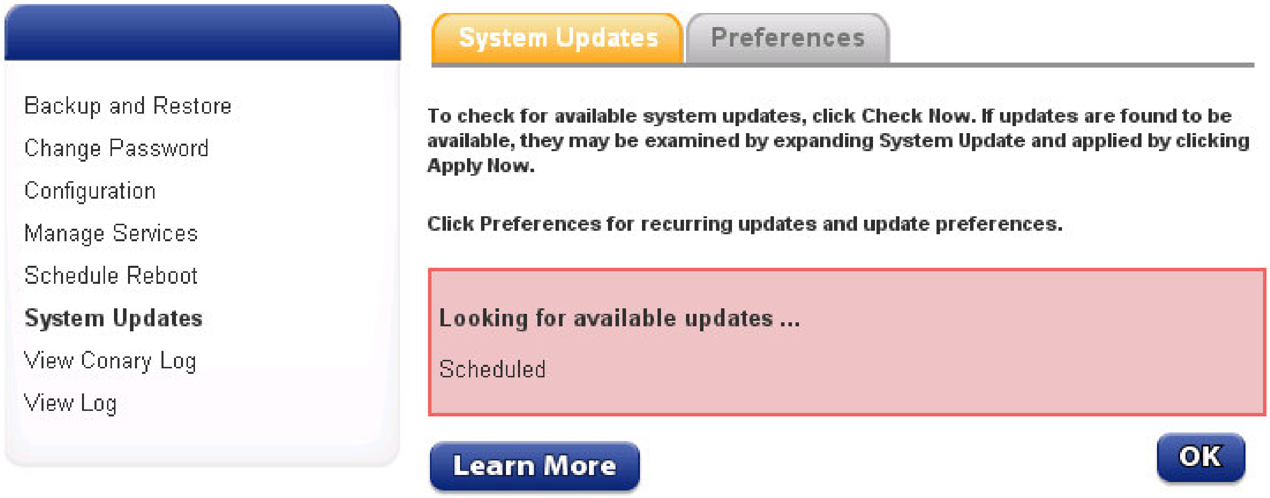
The System update panel gives you access to updating your entire system, including the latest releases of asterisk software. You can schedule a check.

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.



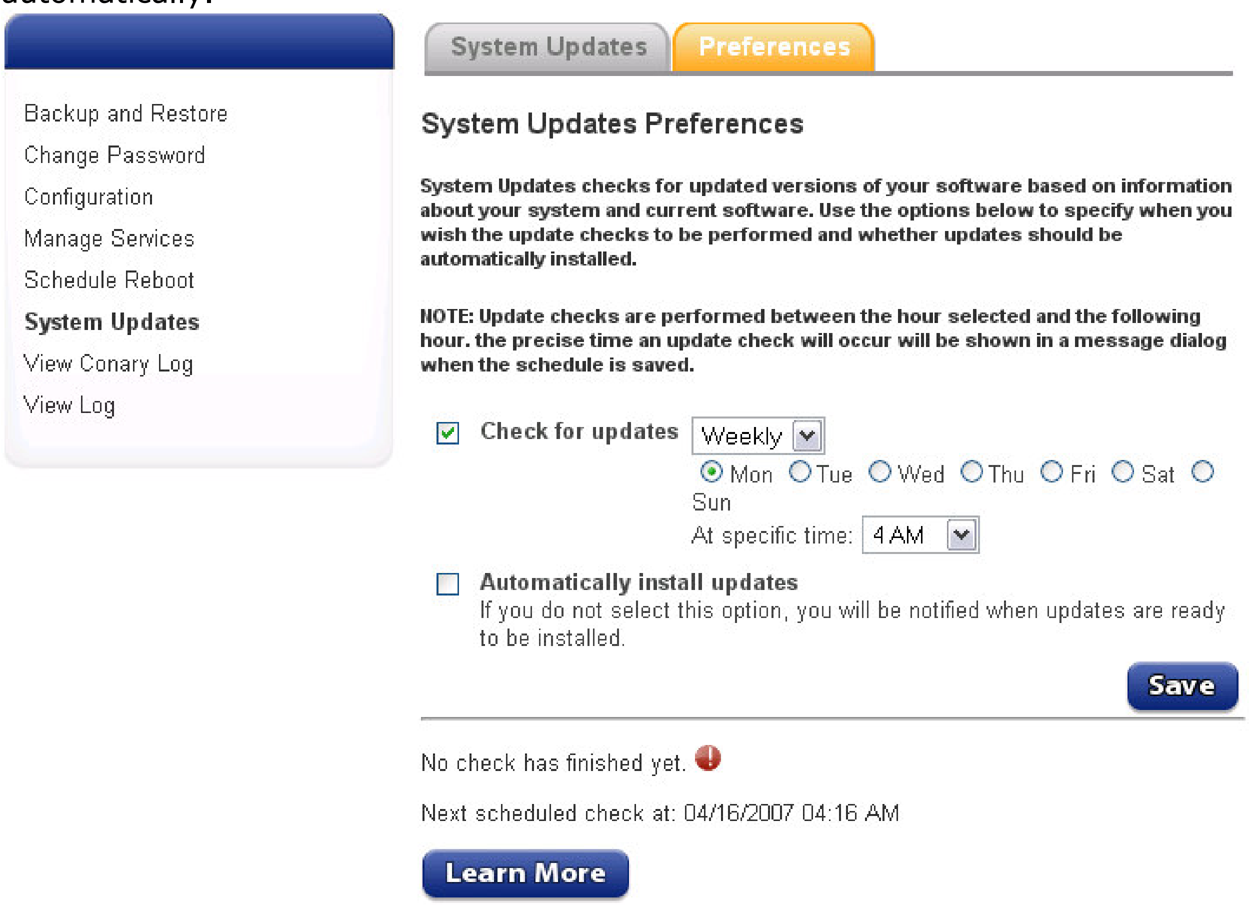
The following shows the update function searching for updates.





Schedule your updates:

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

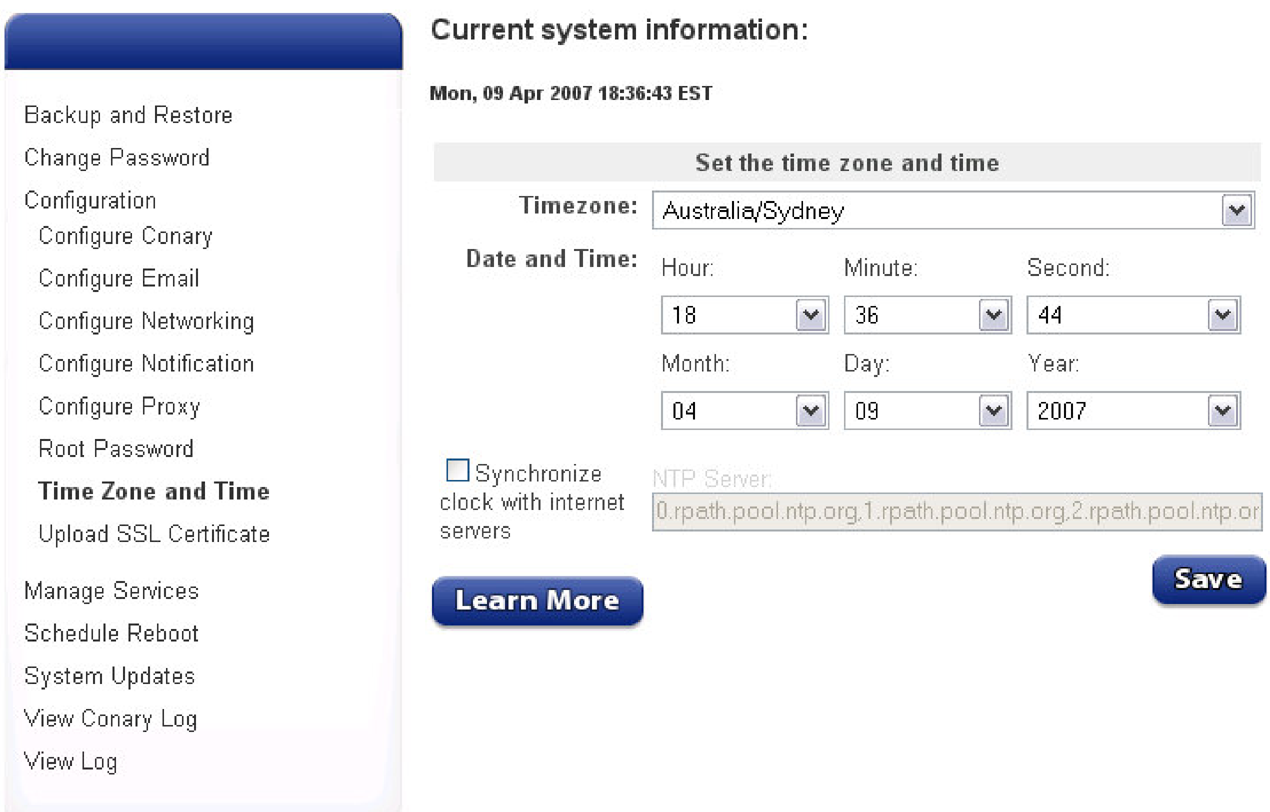


9.6. Conary Configuration:

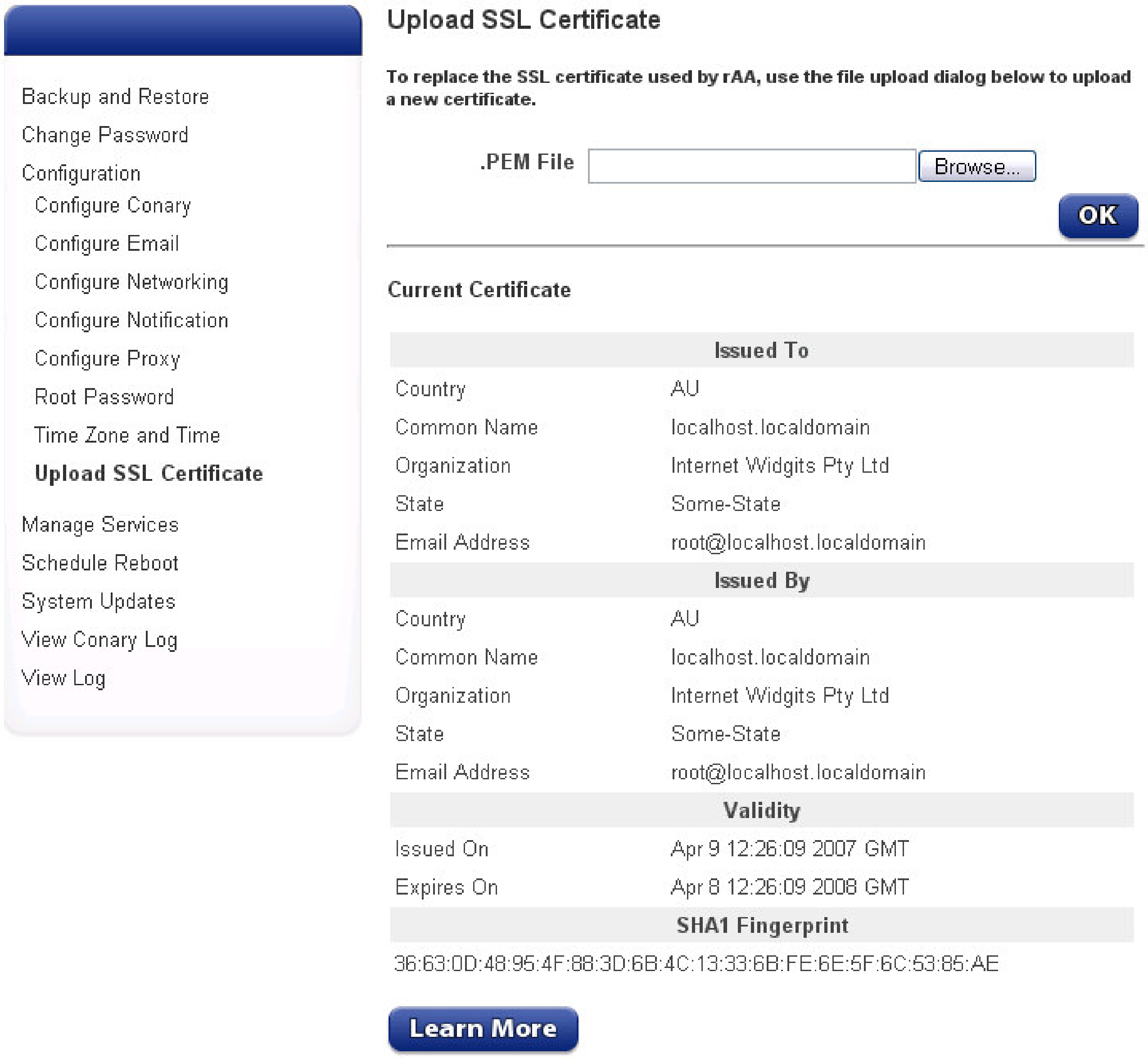
You do not need to update the configuration here. But if you want to check for updates from a different group and install, then you can change and update the configuration.

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.



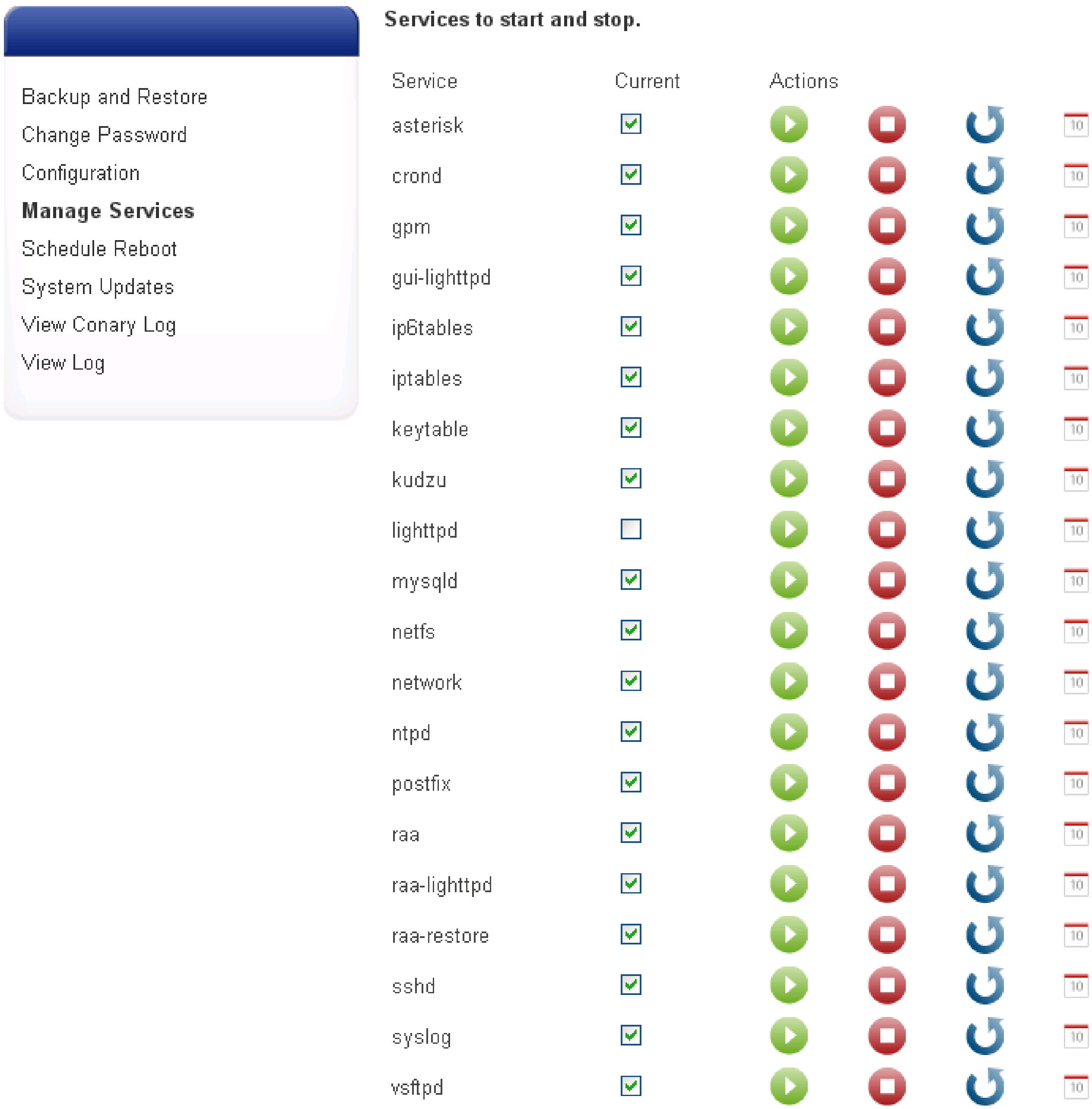
9.8. Upload SSL Certificate:



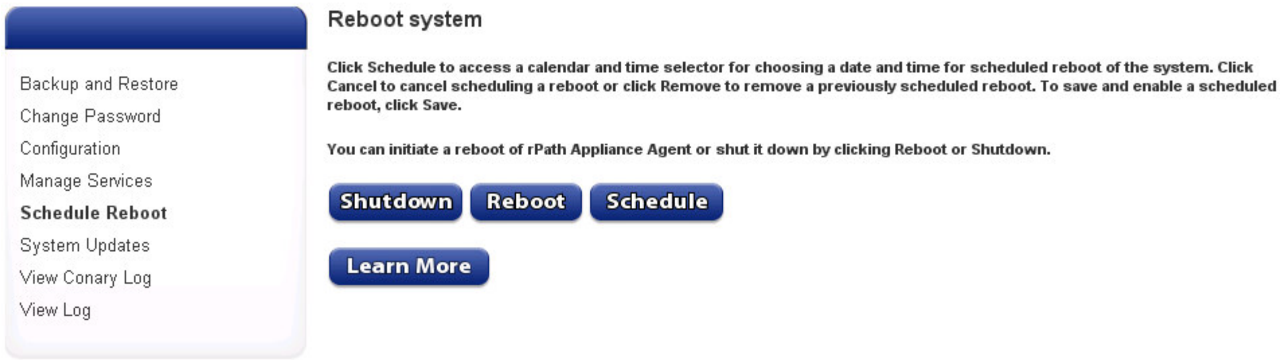
If you want to use your own certificate then upload the certificate here and then reboot the server. From next login you will see the new certificate being used.

9.9. Services:

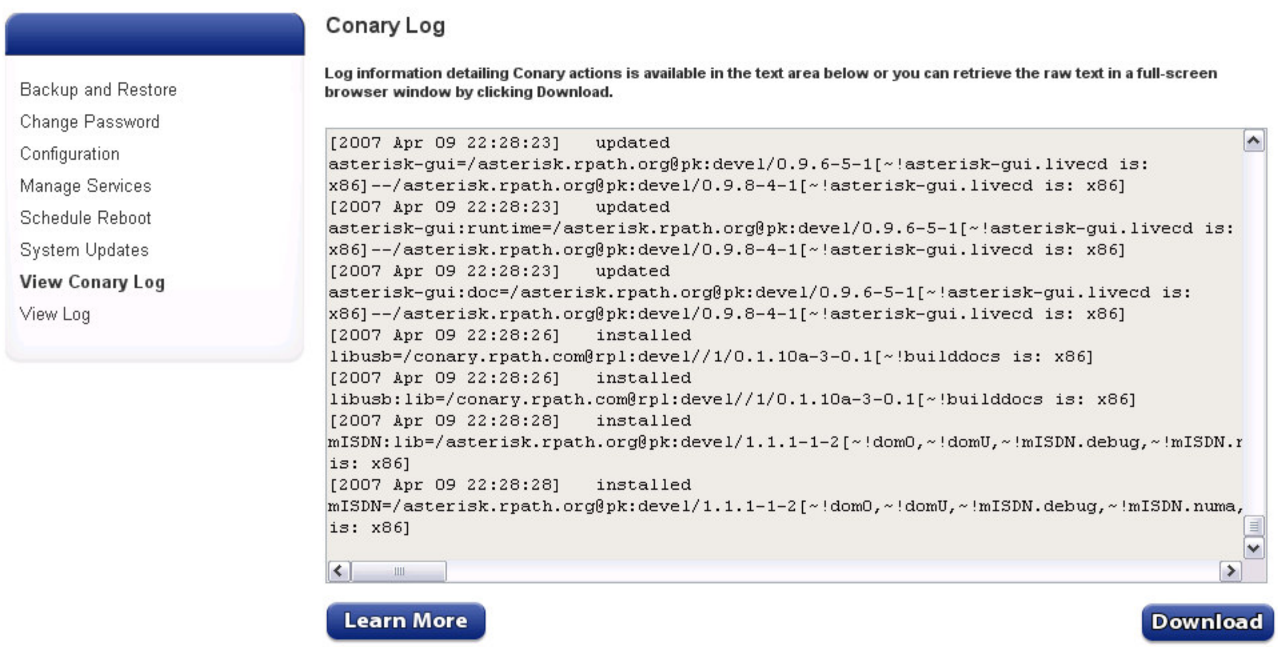
Manage any services that you want to use form this control panel.



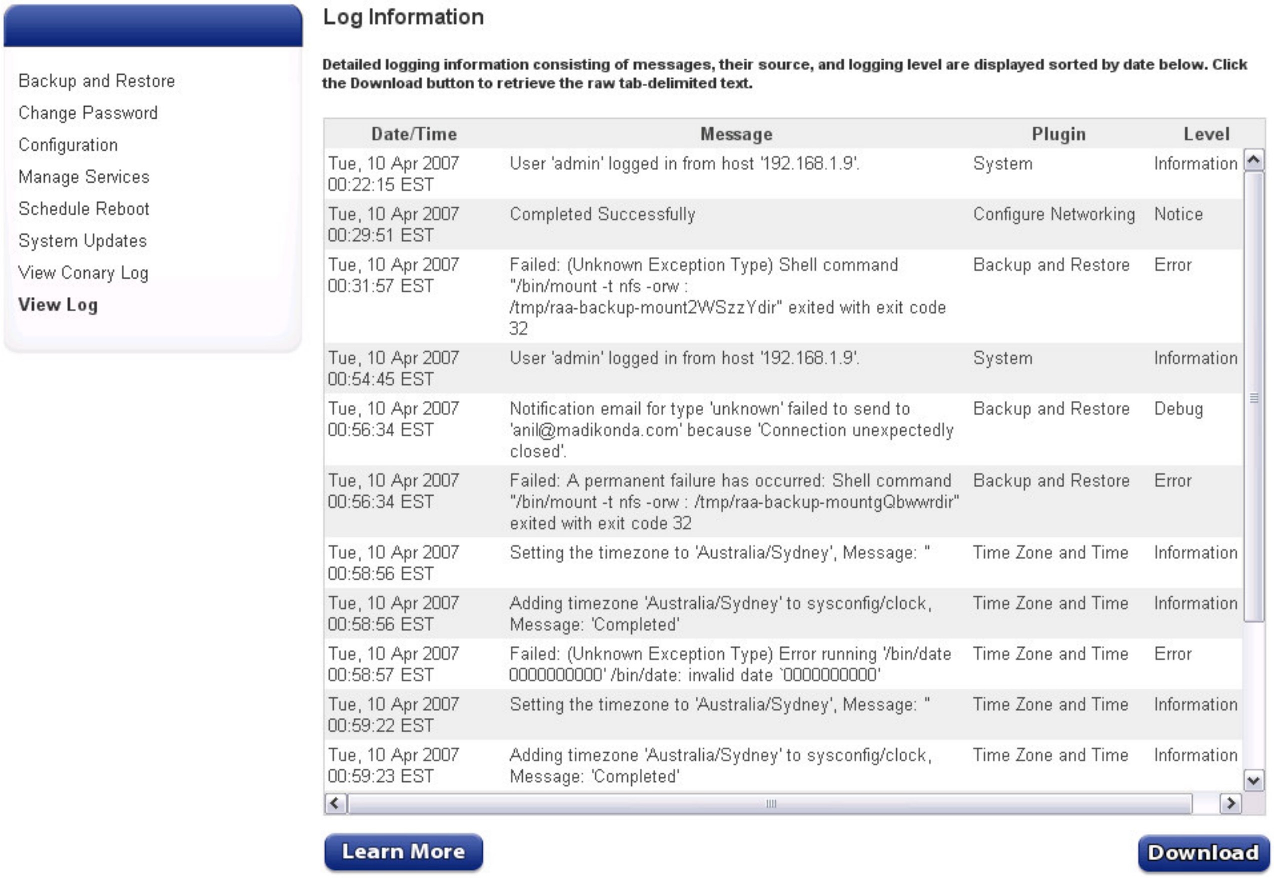
9.10. Scheduled Reboot:



9.11. Conary Log:



9.12. View Log:



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-mountgQbwwrdir" 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 0000000000' /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

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 promt 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!

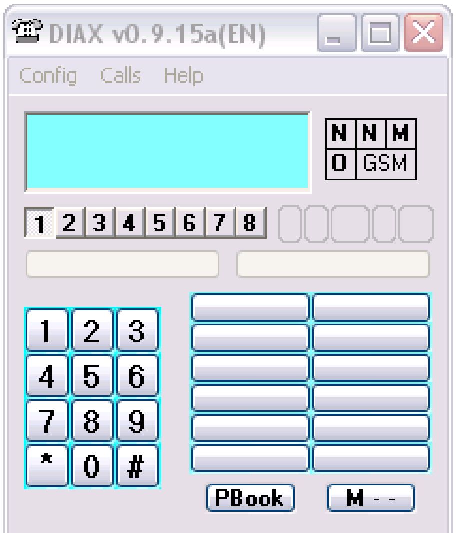
Success,

Willem

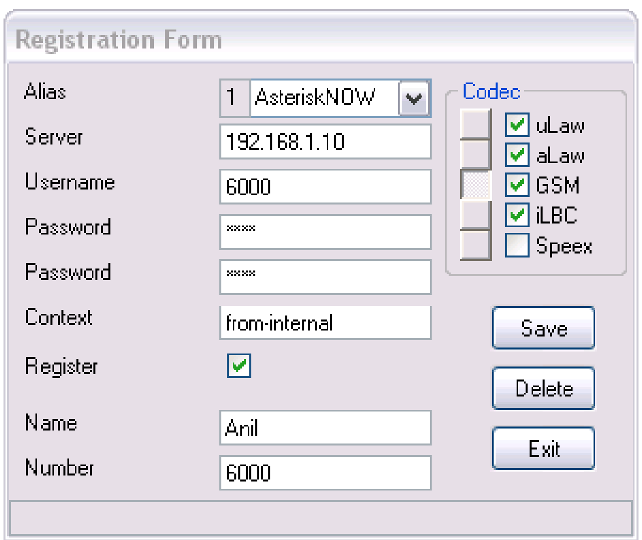
14. Client Connections:

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]*