AudioCodes

SIP Configuration Guide

for using Asterisk@Home with Mediant 1000, 2000 and MP-11x

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Notice

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1 Introduction

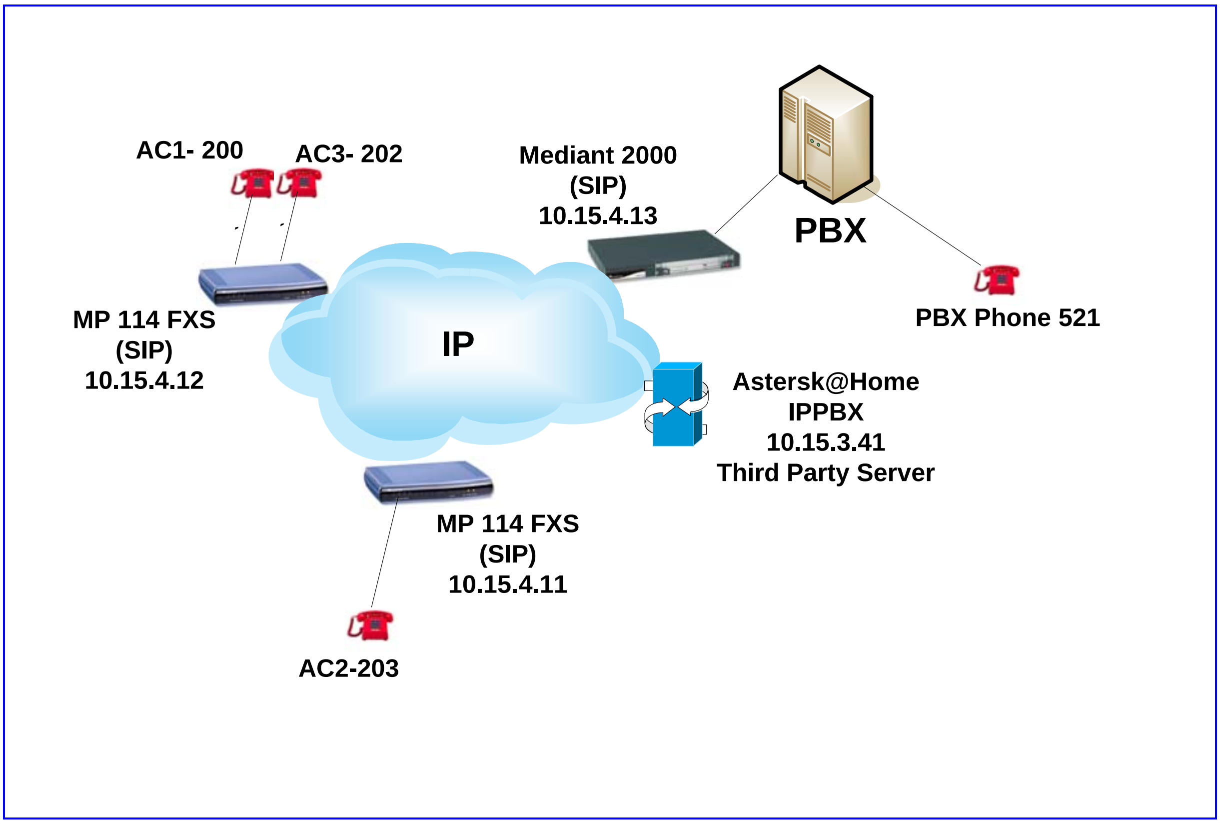
This SIP Configuration Guide is a quick guide to assist you to connect AudioCodes Media Gateways to the Asterisk@Home IPPBX. This guide is not a detailed Installation Manual and it does not cover all the variations and possibilities of Asterisk@Home in minute detail.

This SIP Configuration Guide describes:

- How to configure AudioCodes' SIP Gateway (Mediant 2000, Mediant 1000 and MP-11x) in the Asterisk@Home IPPBX Ver 2.1.
- How to prepare AudioCodes' SIP Gateways with the correct *ini* file.

Figure 1-1 illustrates an example layout of a network in which the Asterisk@Home IPPBX interoperates with AudioCodes' SIP Media Gateways.

Figure 1-1: Example of AudioCodes SIP Gateways with the Asterisk@Home IPPBX



In the example above, the AudioCodes MP-1xx users (10.15.4.11 and 10.15.4.12) were registered as SIP **extensions** to Asterisk@Home IPPBX server.

The Mediant 2000 (10.15.4.13) configured as a SIP **trunk** in Asterisk@Home IPPBX server (without registration process). All SIP signaling as well as the voice streams (RTPs) are managed and go through the Asterisk@Home IPPBX (10.15.3.41).

Reader's Notes

2 Configuring SIP Gateways in the Asterisk@Home IPPBX

2.1 Preparing Asterisk@Home Default Settings After Installation

Refer this section in case of a first time after installation.

Once Asterisk@Home has been installed, some default system changes need to be made to Asterisk.

Log in to your new Asterisk@Home box (***user:*** root, ***password:*** password)

Figure 2-1: Login Screen

```
login as: root
root@192.168.1.7's password:
Last login: Thu Aug 18 11:00:07 2005

Welcome to Asterisk@Home
-----------------------------------------------------

For access to the Asterisk@Home web GUI use this URL
http://192.168.1.7

For help on Asterisk@Home commands you can use from this
command shell type help-aah.

[root@asterisk1 root]#
```

2.1.1 Change Linux Password

**NOTE: CHANGE YOUR ROOT PASSWORD IMMEDIATELY by typing Passwd (note the spelling)

Figure 2-2: Change Linux Password

```
[root@asterisk1 root]# passwd
Changing password for user root.
New UNIX password:
BAD PASSWORD: it is based on a dictionary word
Retype new UNIX password:
passwd: all authentication tokens updated successfully.
[root@asterisk1 root]#
```

****Asterisk@Home boxes with default passwords have been hacked!!**

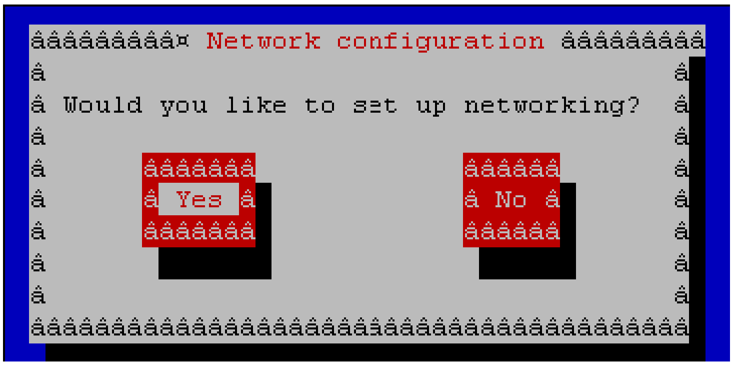
You may need to change some of the other default passwords as well and also set your date, and the IP address of your box to a static address.

2.1.1 Change IP Address (set IP address to Static)

Change Asterisk IP address from DHCP to Static. At the command prompt enter: **netconfig**

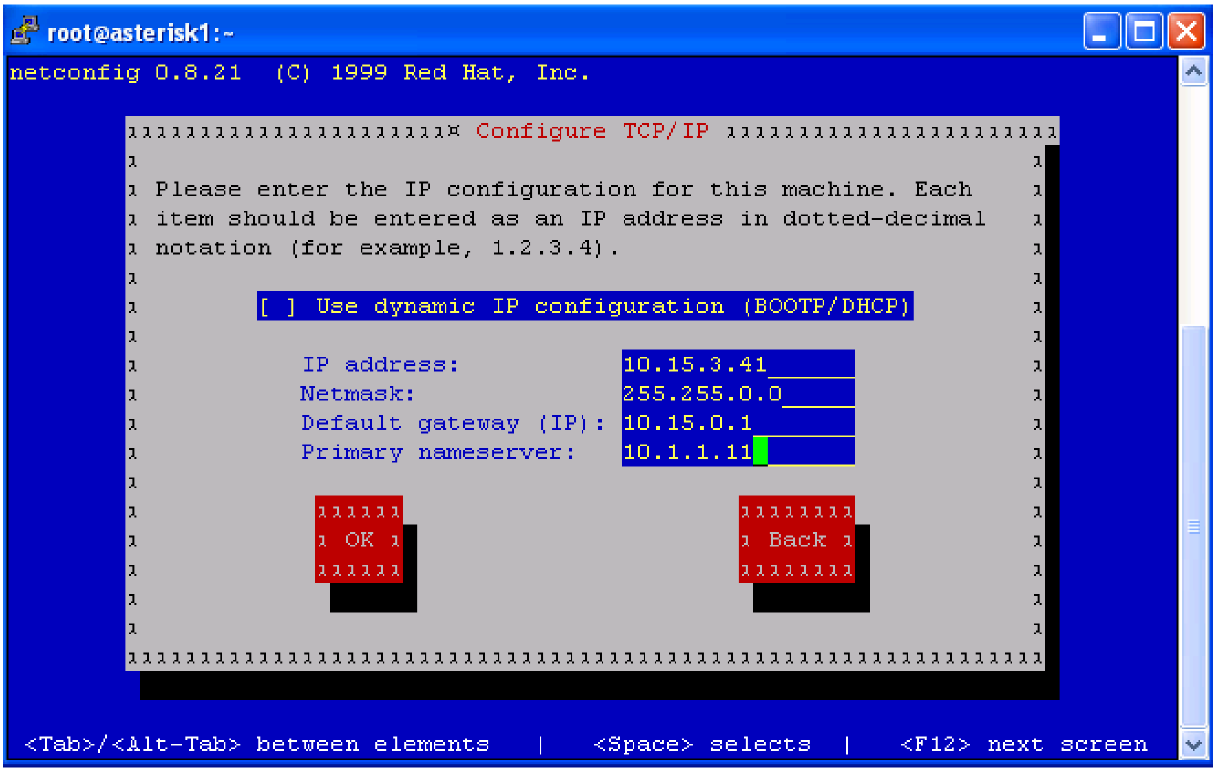
Select [**Yes**] to set up networking and hit enter.

Figure 2-3: Network Configuration Request



Select **[Yes]** to set up networking and hit enter.

Figure 2-4: Configuration TCP/IP



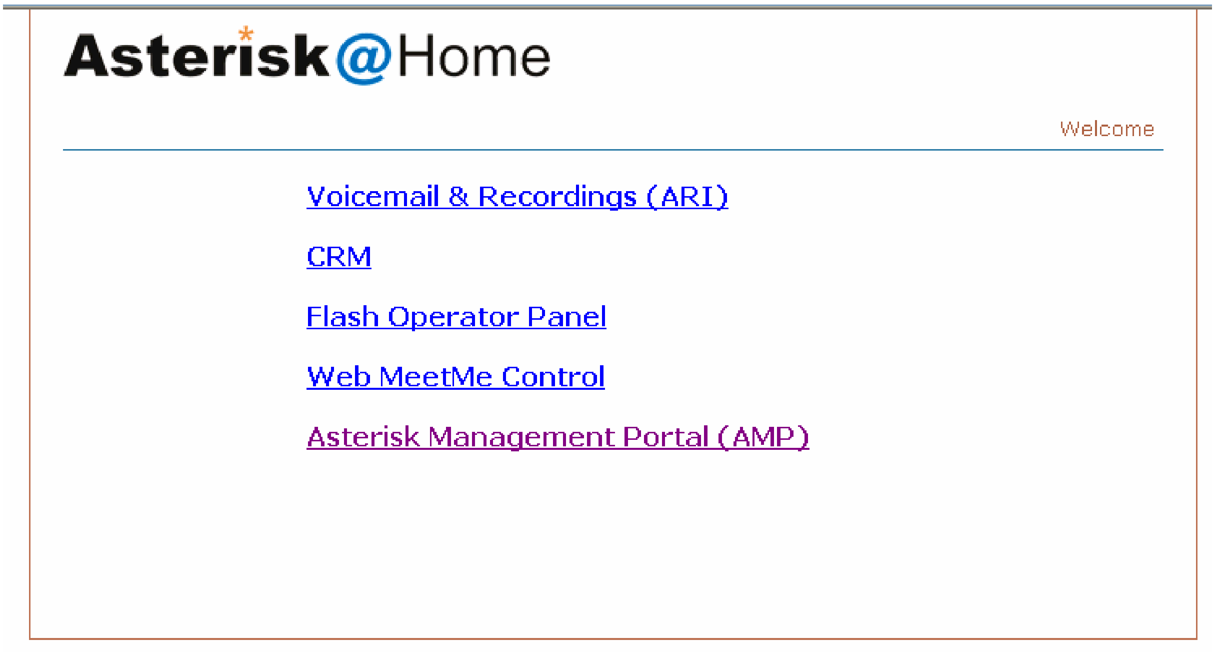
Use the Tab key to cycle through the fields. Enter the IP address that is to be allocated to the Asterisk box, the Netmask (subnet mask), Default Gateway and Primary nameserver as shown in the example above. In the example, we used our existing network regime.

Once done, select OK. You need to reboot for the changes to take effect.

2.2 Connect To AMP From Web Browser

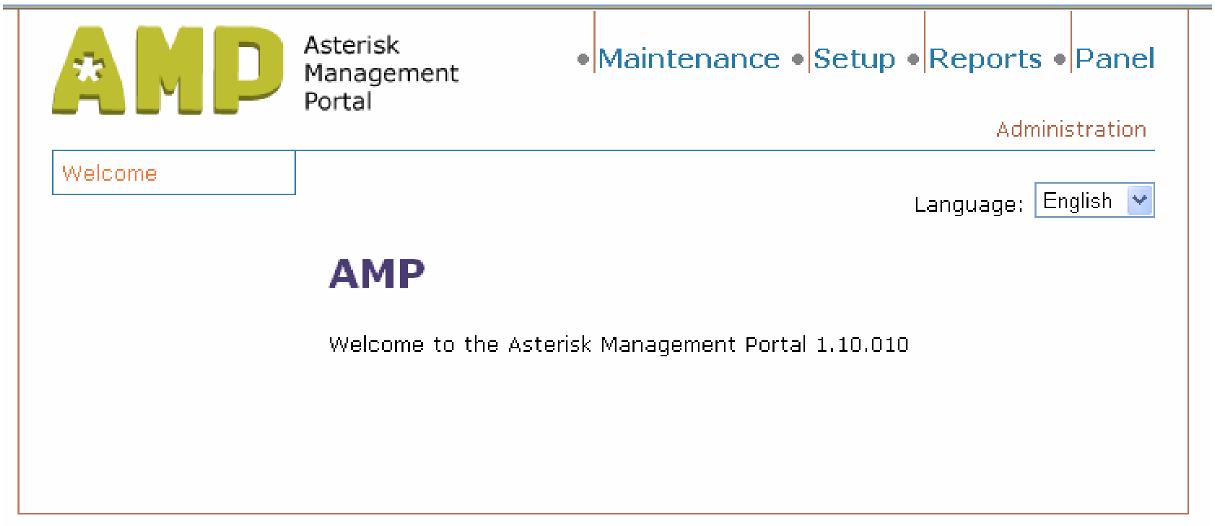
Now you can connect to ***http://ipaddress/*** (e.g. http:10.15.3.41) to configure Asterisk@Home. You are presented with AMP (Asterisk Management Portal) as illustrated below.

Figure 2-5: The AMP (Asterisk Management Portal) Welcome Page.



Using AMP (Asterisk Management Portal) we can easily configure our asterisk server.
Log in to AMP (Asterisk Management Portal)
To log in to AMP (Asterisk Management Portal) use* ***user*: maint,* ***password****: password unless you have changed the password during initial set up)*
Once you logged in to AMP, you are presented with the following screen,

Figure 2-6: The AMP (Asterisk Management Portal) Administration Page



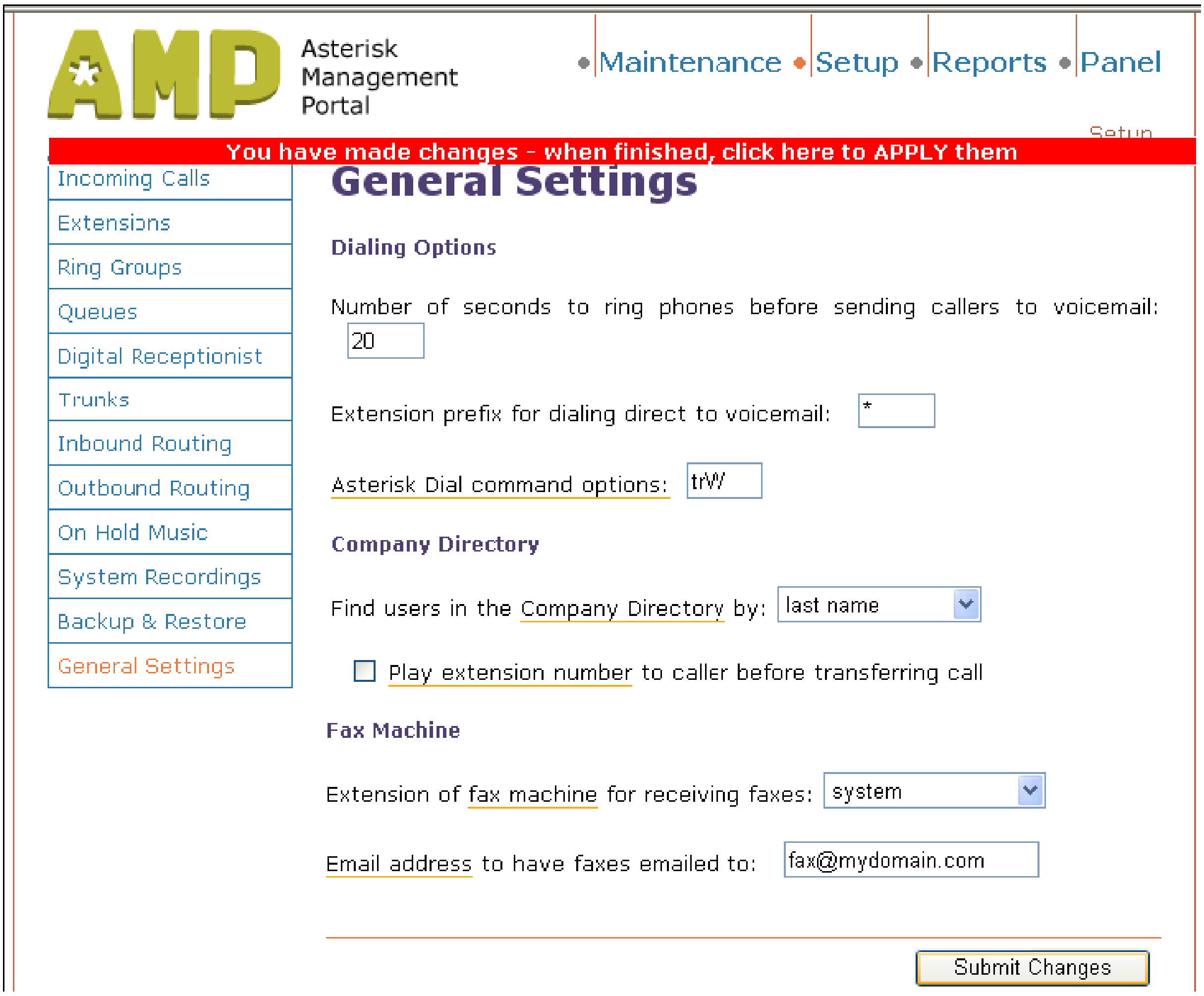
At this stage select the **Setup** Tab and you will be presented with the following screen.
This is where you will start configuring Asterisk@Home. Notice the selection options on the left. Selecting each option will display configuration screen for that particular function e.g., creating new extensions, creating new trunks etc.

Figure 2-7: The AMP (Asterisk Management Portal) Setup Page

2.3 General Settings

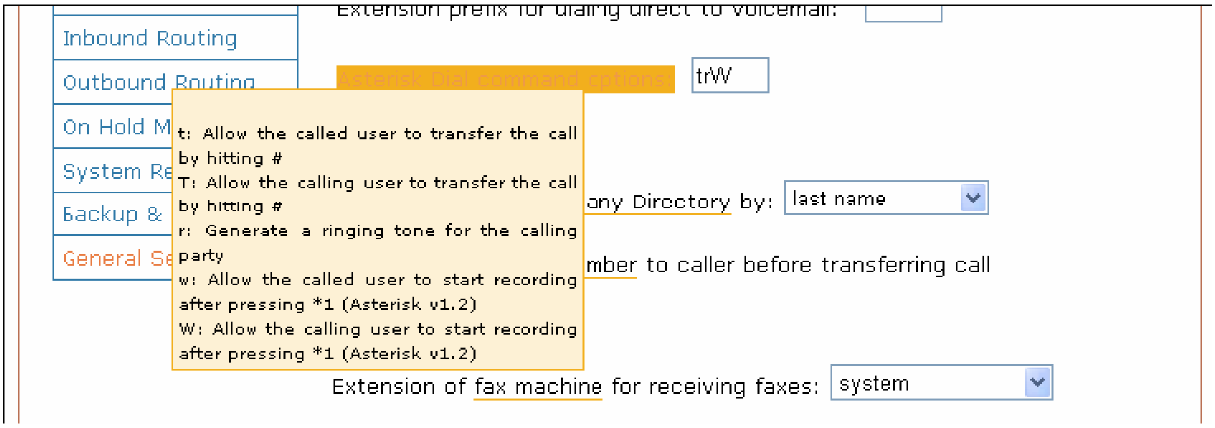
In the General Setting page you be able to configure the general behavior of the IPPBX see as example the illustration below.

Figure 2-8: **The AMP (Asterisk Management Portal) General Setting Page**



Hovering your mouse on the corresponding field description with a yellow/amber underline displays the function of the fields.

Figure 2-9: **The Function of the Fields**



For example, if you want to allow the called user to transfer a call by hitting # before the number type 't' in the Dial Command Option field, hover your mouse on the label and you are informed of the other options.

After setting up the General Settings, click on Submit Changes button and the red bar on top of the screen for the change to take effect. Next, let us set up the extensions.

2.4 Extensions

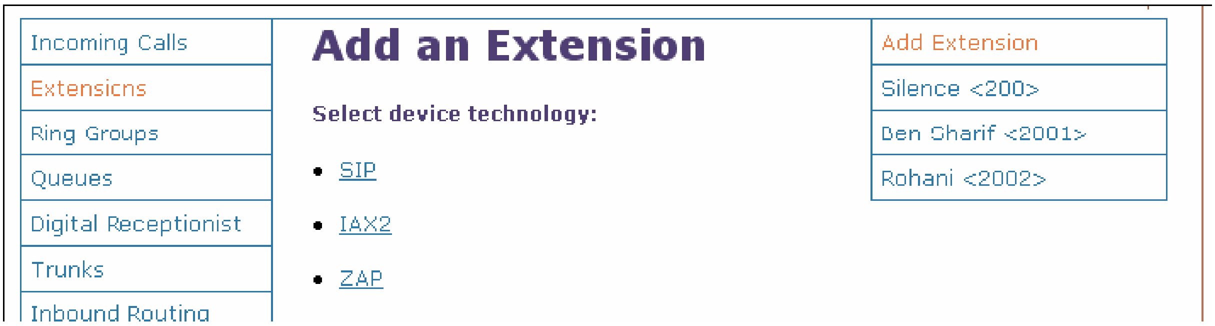
The following AudioCodes equipment can be configured as an extension in Asterisk@Home:

MP-11x FXS, MP-11x FXO, Mediant 1000 analog interface FXS and FXO

2.4.1 Configuring AudioCodes Devices as an Extension

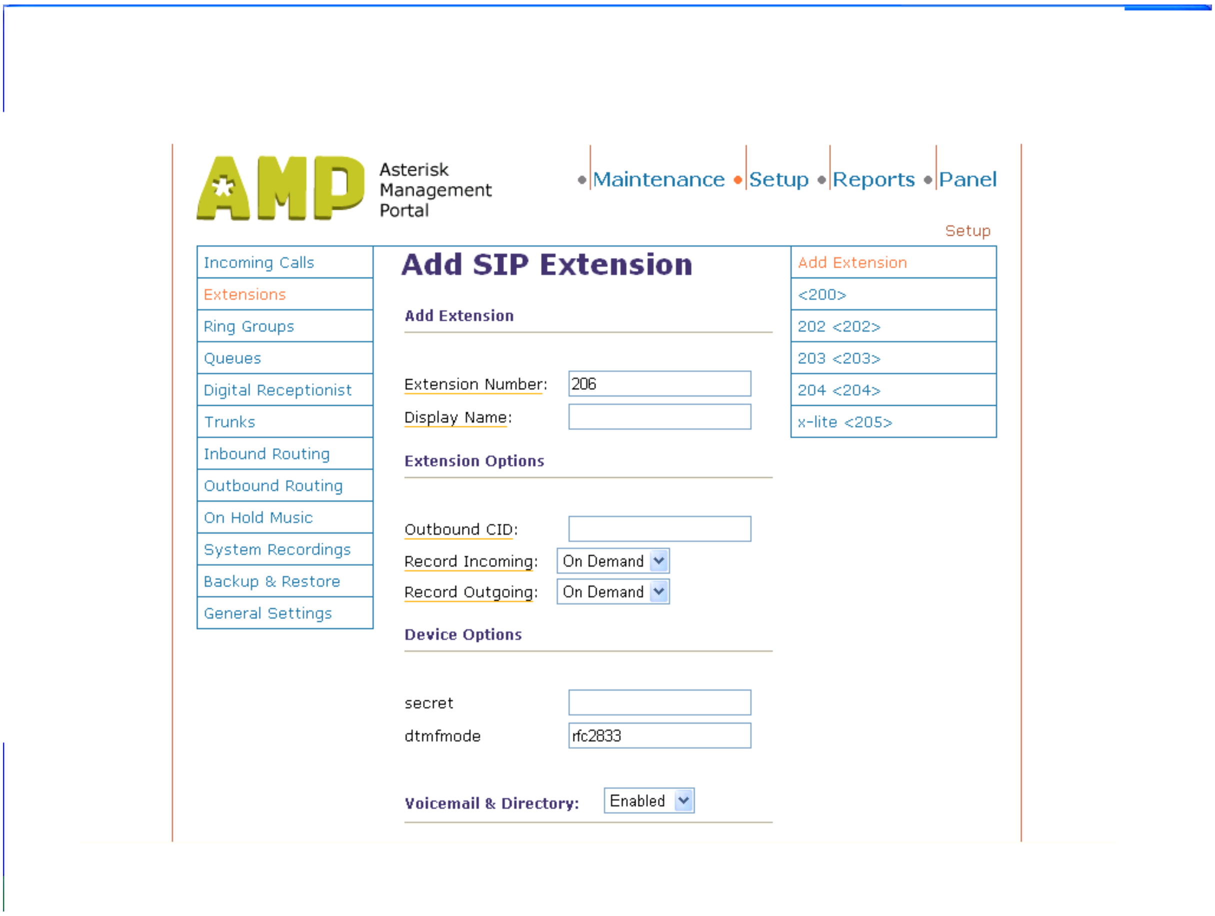
To add the AudioCodes FXS endpoint to Asterisk@Home press the Extension button

Figure 2-10: Add an Extension



For SIP extension select **SIP** in the ***Select device technology*** drop down box above.

Figure 2-11: Add SIP Extension page



Enter the **extension number**.

Enter in **secret** the sip password as configured in the MP-11x.

For ***DTMF Mode***, select "**rfc 2833**"

If you enabled Voicemail, you may allocate a password for voice mail. You may also nominate an email address for Voicemail Email.

After setting up the Extension parameters, click on Submit Changes button and the red bar on top of the screen for the change to take effect.

Click on the Add Extension button to add more extensions.

In our example above we added 3 Extensions (200,202,203).

2.5 Trunks and Routes

The following AudioCodes equipment can be configured as an extensions in Asterisk@Home:

Mediant 2000, MP-11x FXO, Mediant 1000 Digital interface.

2.5.1 Configuring AudioCodes devices as a Trunk

A trunk is the telephony service line that you will be using to make an external call. For example, AudioCodes Mediant 2000 gateway can be configured as a Trunk to enable you to make calls to and from PSTN.

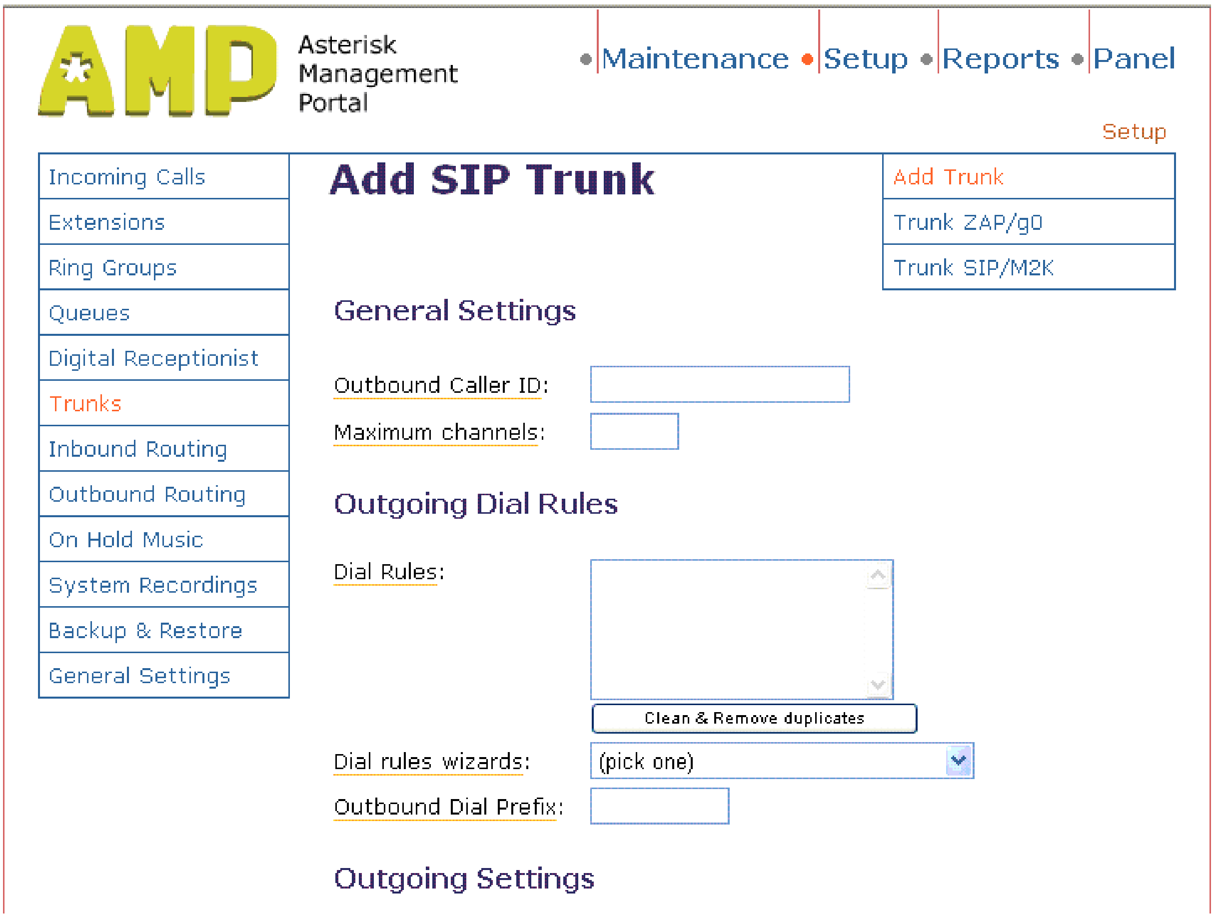
To make external, PSTN calls; you must have at least one trunk.

To create a new trunk using **AMP**, select **Setup** tab and then select the **Trunks** option from the vertical menu on the left.

Figure 2-12: Add a Trunk

To create a new **SIP Trunk**, click on the Add SIP Trunk option.

Figure 2-13: Add SIP Trunk Setting page



Next we need to create the Outgoing Setting, Incoming Settings

Outgoing Settings

In the **Trunk Name** field enter the name of this trunk: e.g. ***M2K***

In the **Peer Details** enter the following;

host="M2K IP address" e.g. 10.15.3.41

type=peer

Incoming Settings

In the **User Context**, enter the Number your provider expects e.g., **521**

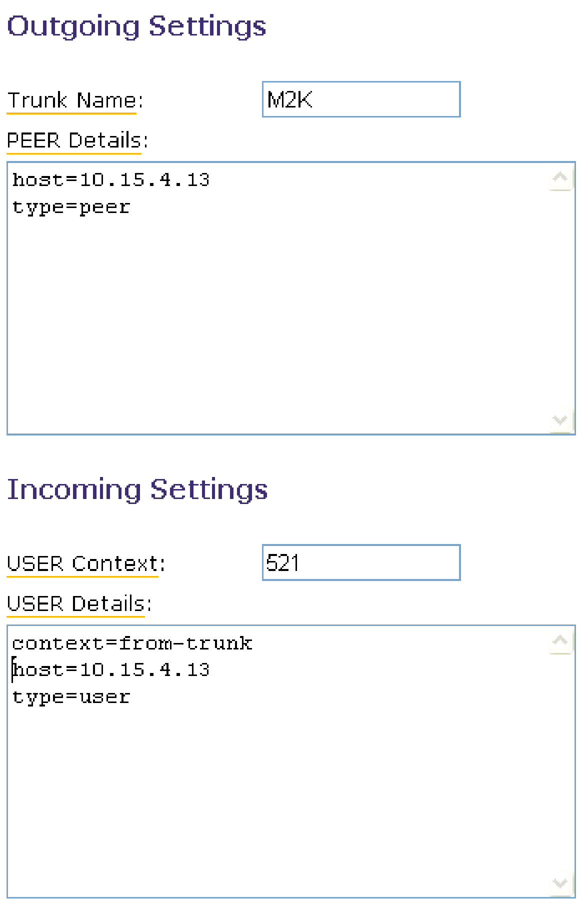
In the **User Details** we have the following:

context=from-trunk

host=" M2K IP address" e.g., 10.15.3.41

type=user

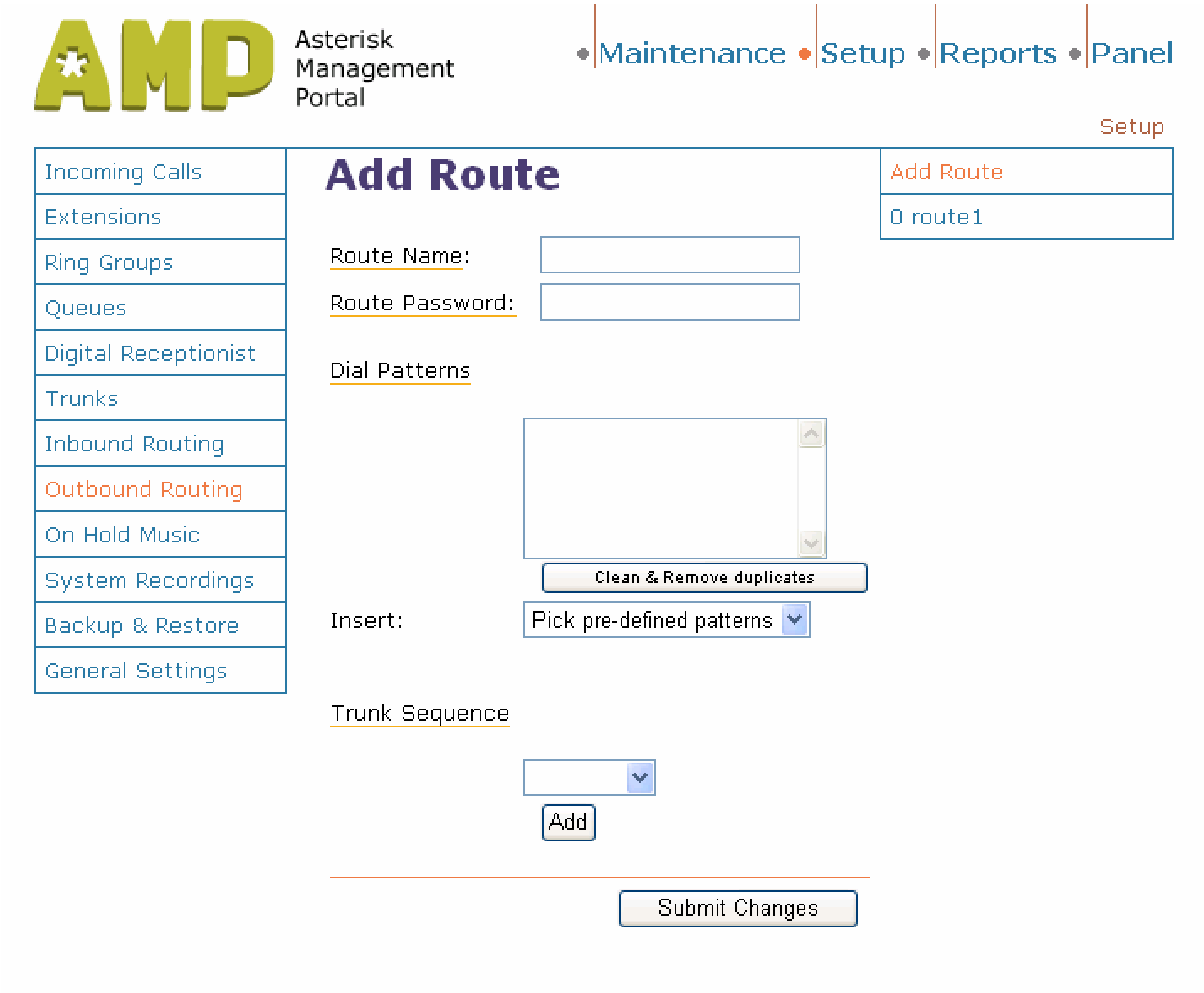
Figure 2-14: Outgoing Settings



2.5.2 Create Outbound Routing

An ***outbound route*** works like a traffic cop giving directions to road users to use a predefined route to reach a predefined destination. To create a new route using **AMP**, select **Setup** tab and then select the **Outbound Routing** option from the vertical menu on the left.

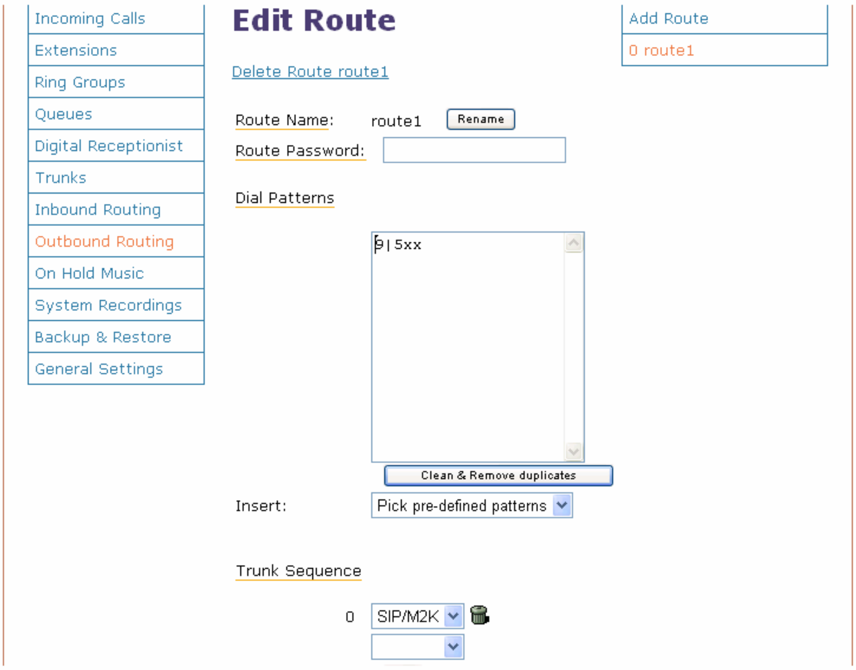
Figure 2-15: Add Route Setting Page



Every time you dial a number, Asterisk@Home will do the following in strict order,

- Examine the number you dialed.
- Check if the number is internal number. If matches it will initiate internal call.
- Using the outbound routing definitions compare the number with the pattern that you have defined in your first route and if matches, it initiates the call using that trunk. If it does not match, it compares the number with the pattern you have defined with the second route and so on.
- Pass the number to the appropriate trunk to make the call.
- To make a call out (except inter extension calls), you need at least one trunk and one route.
- To create a new Outbound Routing, Click on the Add Route in the menu on the right of the screen.
- In your **Dial Patterns** box, you need to type the pattern to match the Mediant 2000 dialing.
- For our example: **9|5xx** Meaning that when you dial 95xx, Asterisk routes the call to the Mediant 2000 trunk.
- In the **Trunk sequence** select the appropriate Trunk e.g. **M2k/SIP** Trunk.

Figure 2-16: Route Screen



2.5.3 Configuring Incoming Calls

This is where the behavior of incoming calls from PSTN is being handled.

When an incoming call from PSTN is received, Asterisk needs to know where to direct it. The call can be directed to a ring group (refer to section **2.5.3.3**), to a specific extension, Digital Receptionist or Queue.

There are two options to configure the incoming call:

Incoming call – general setting where to send the incoming calls comes from the PSTN.

Inbound route – specific route for individual incoming call.

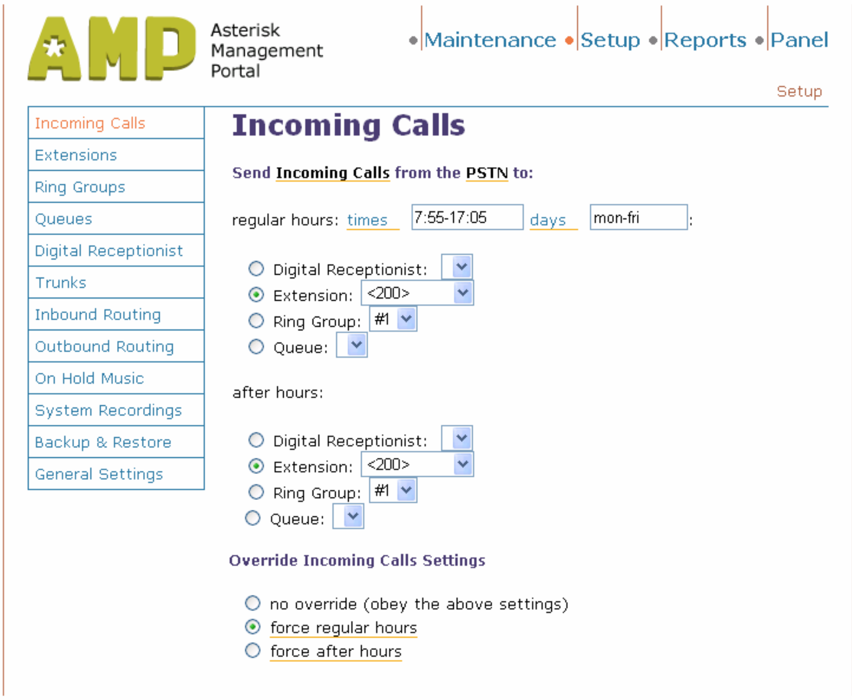
When an incoming call from PSTN is received, if it match to the inbound route definition Asterisk will initiate the call according to the Inbound route destination set, otherwise it will behave as defined in the Incoming call setting.

2.5.3.1 Incoming Call

Incoming Calls options need to be set up to allow calls from your PSTN to go someplace. In our example above we configured to send the incoming call from the PBX (that connected to Mediant 2000) to extension 200.

Select the Incoming Calls selection in the left bar of the screen.

Figure 2-17: Incoming Call Setting Page

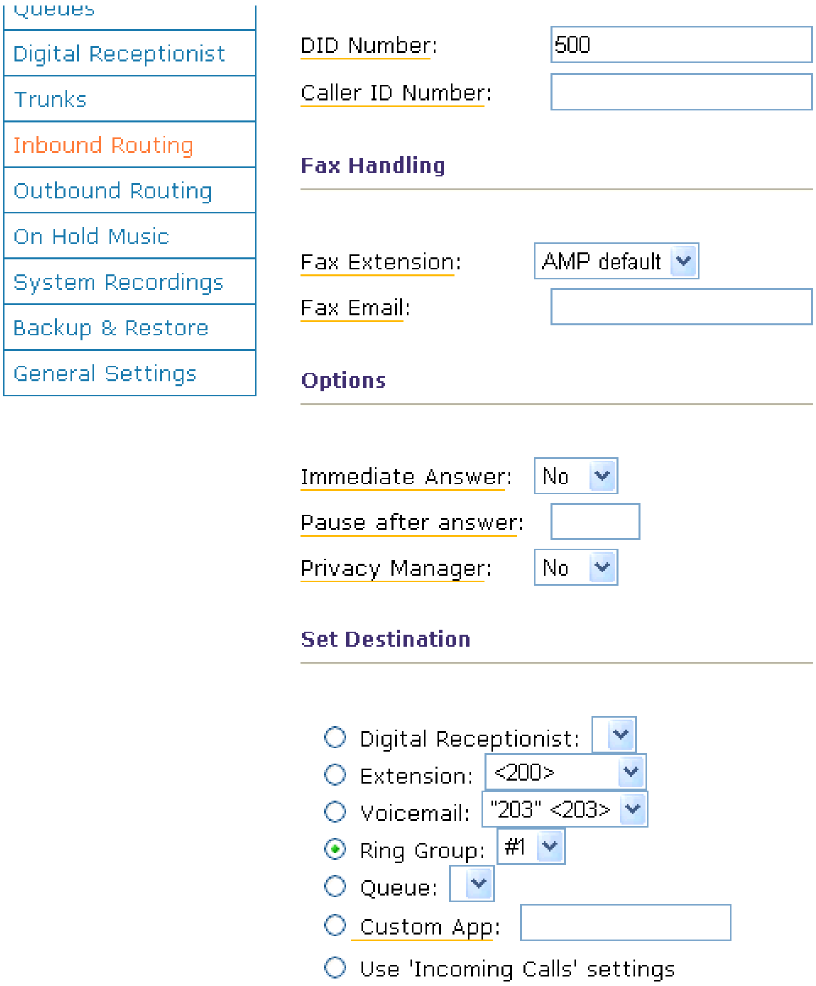


2.5.3.2 Inbound Routing

Incoming calls from any of trunks can be routed to specific extension/s or Ring Group, which is definable when setting up the individual route.

For our example we configured that all calls that come in through the Mediant 2000 trunk from DID 500 are routed to Ring group 1 (refer to section **2.5.3.3**) as shown in the illustration below.

Figure 2-18: Inbound Routing



2.5.3.3 Ring Groups

You may not want a Ring Group – it's entirely up to you. If you don't require a ring group, you may ignore this section

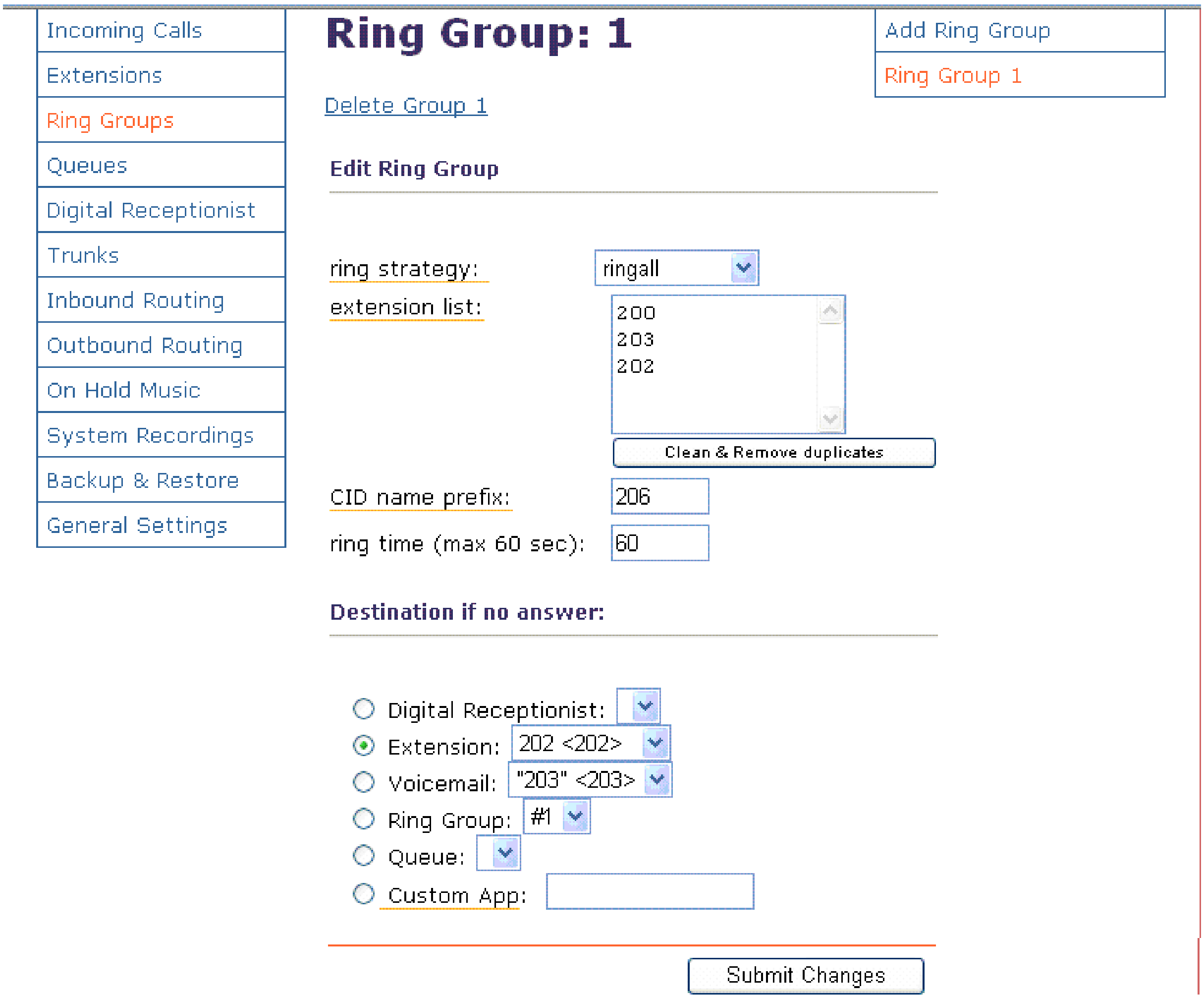
A Ring Group is a group of extensions that will ring when there is an external incoming call.

When there is an incoming call, the phones nominated in the selected group will ring. You may select different Ring Groups for each of the incoming trunks or you may nominate the same group for all the trunks, in which case you will only need to define only one Ring Group.

In our example above we configured one Ring Group with the nominate 200,202,203. when there is an incoming call from the PBX through the Mediant 2000 to one of the MP-11x extension all of them will ring.

The Ring Group screen is illustrated below:

Figure 2-19: Ring Group Settings Page



2.6 Edit Configuration Files.

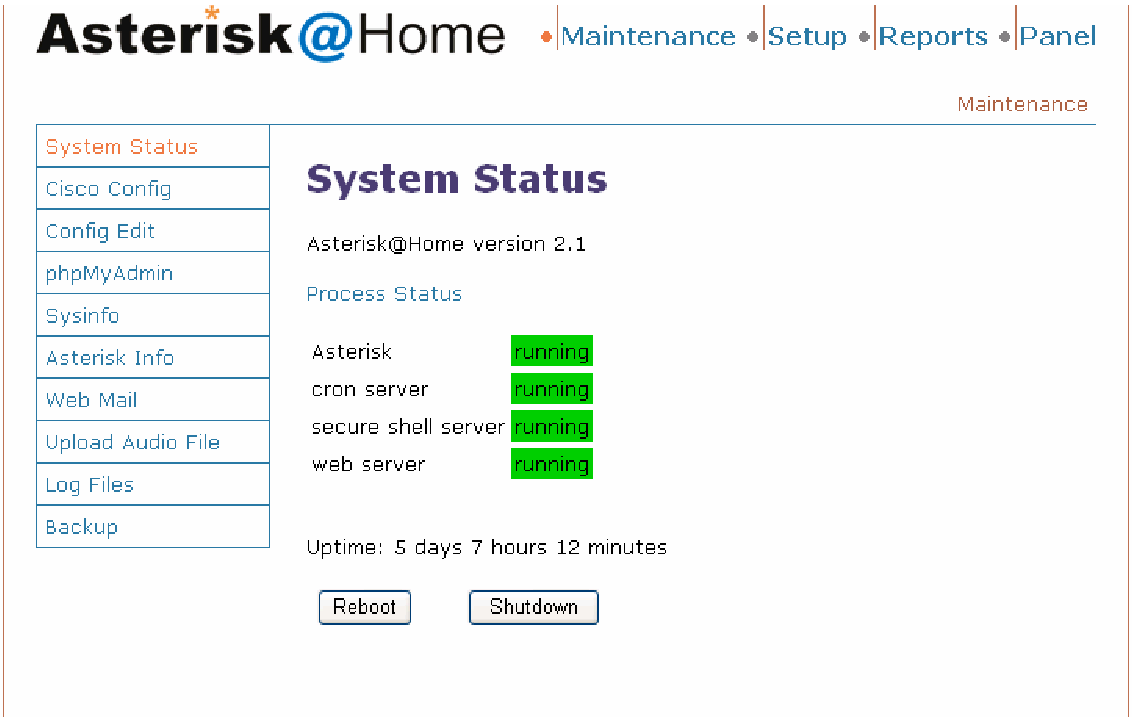
Just as you think that all is OK, you realize something else requires attention.

This is true with Asterisk@Home as well.

To do this you need to edit some configuration files **(.conf)** that reside both in the /etc/asterisk directory and /etc directories. Configuration files in the **/etc/asterisk** are generally editable through AMP Maintenance Tab.

To start editing the **.conf** files you need to log in to AMP and select the Maintenance Tab. AMP presents you with the following screen.

Figure 2-20: The AMP (Asterisk Management Portal) Maintenance Page



Select **Config Edit** and you can see a new screen with a list of all the **.conf** files, which can be edited from AMP.

You may scroll down the page to find the file that you wish to edit.

Figure 2-21: Config Screen

A number of ***.conf*** files may require editing for Asterisk to function, depending on the individual requirements. For example if there is need to support with more Codecs than µ-law or A-law you need to configure the SIP.conf configuration file as shown below:

After made changes in the configuration files you will need to reboot for the changes to take effect or made other change in the setup configuration and press the red bar on top of the screen.

SIP.Conf

Note: If your SIP devices are behind a NAT and your Asterisk server isn't, try adding "nat=1" to each peer definition to solve translation problems.

[general]

port = 5060 ; Port to bind to (SIP is 5060)
bindaddr = 0.0.0.0 ; Address to bind to (all addresses on machine)
disallow=all
allow=alaw
allow=ulaw
allow=g729
allow=g723
allow=g726
context = from-sip-external ; Send unknown SIP callers to this context
callerid = 200

#include sip_nat.conf
#include sip_custom.conf
#include sip_additional.conf

3 Preparing SIP Gateways to Interoperate with Asterisk@Home

Define the following parameters in the *ini* file that relate to configuring the Mediant 2000, and MP-11x to interoperate with the Asterisk@Home

1. In the case of an AudioCodes **FXS** device (AudioCodes' MP-11x FXS and the Mediant 1000 FXS module), define the endpoint phone number as defined in the Asterisk Extensions.
2. Set the Authentication User name and password parameters as defined in the Asterisk Extinctions numbers and secret.
3. **IsProxyUsed=1**
4. **ProxyIP="Asterisk@Home ip address"**
5. **PRACKMode = 0 or 1 but not 2, if the PRACK required the calls will fail.**
6. **IsFaxUsed = 2,** Initiates fax using the coder G.711 A-law/µ-law with adaptations
7. **TxDTMFOption=4, RFC2833**
8. **DISCONNECTONBROKENCONNECTION=0**
9. Define the coder list and priority as defined in the Asterisk SIP.Conf file.
10. In the case of an AudioCodes **FXS** device (AudioCodes' MP-11x FXS and the Mediant 1000 FXS module), define the **IsRegisterNeeded=1.** for Mediant 2000 that configured in Asterisk as a Trunk define the **IsRegisterNeeded=0**
11. In the case of an AudioCodes **FXO** device it can be configured in Asterisk@home as an Extension or as a trunk depend in the implementation and the network needs.

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