

Asterisk^{*}@Home Handbook



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1 Introduction

Asterisk@Home was created to make installing Asterisk easy. Experimenting with Asterisk should be fun and not take hours or days to set up.

2 Installation

2.1 Hardware Requirements

The faster the system you use to run Asterisk the more simultaneous calls it will be able to handle a 500MHz PIII with 128 Megs of RAM should easily meet the needs of the average home use.

2.2 Install from ISO

Download the latest ISO from <http://asteriskathome.sourceforge.net> and burn it to a CD Insert the newly created Asterisk CD into the CD ROM drive of the system on which you want to install Asterisk@Home. Reboot the "soon to be" Asterisk system and press Enter when prompted.



Warning

This will erase all data on the hard drive of the PC!!!

After Linux has loaded, the CD will eject. Remove the CD from the system and wait for the

system to reboot. Booting the system might take a while, depending on the speed of your computer. It is necessary for the system to build Asterisk from source, so be patient. Once this process is complete, log in to your new Asterisk@Home system with the following username and password:

(user: root, password: password)



Caution

Change your root password immediately by typing **passwd** as this minimizes the chance of your Asterisk@Home system being hacked!!

2.3 Install to an existing CentOS 3.5 system

If you have problems with the ISO you can install onto an existing CentOS server.

Download CentOS 3.5 ISOs from the [CentOS Mirrors](#)

Make a directory to install from, put the install file there, and install.

```
mkdir /var/aah_load
cp asteriskathome-1.5.tar.gz /var/aah_load
cd /var/aah_load
tar xvfz asteriskathome-1.5.tar.gz
./install.sh
```

3 Configuration

Once you have installed your Asterisk@Home system you can start the configuration process. First register a phone to an extension. Next configure a trunk to make outbound calls and receive incoming calls.

3.1 Quick start

3.1.1 Set an IP address for your Asterisk@Home box

To configure Asterisk@Home you will need access to the Web GUI.

If you log into the console you will get a message which displays the IP address of your Asterisk@Home system. If this field is blank then you have a problem with the network card. Type **netconfig** at the command line. This will allow you to set a static IP address for your Asterisk@Home box.

After you have configured your network card reboot the system by typeing **reboot**. Your system will restart and you should be able to connect to your system from any web browser on your network.

3.1.2 Configure an extension

Go to a pc on your network with a web browser and connect to your Asterisk@Home box. Click on **Asterisk Management Portal (user: maint, pass: password)** and then **setup**. Click **Extensions** then **Add Extension**. Use the default extension **200** and type in a password for registration like "abc123". then enter the name of the person using this extension.

Go down to the voicemail section and enter a voicemail password. Use something you can type on a phone keypad like '1234'. Enter an e-mail address where you would like your voice messages sent and **click add extension**. Then click on the **red apply bar** at the top of the screen.

Set up a phone for this extension. Get a SIP phone an **X-Ten soft phone** is good for testing. See the [section on the X-Ten phone](#). remember to use your **extension** and **extension password**.

Make a call from your phone. (try *45 this is a local echo test)

3.1.3 Configure a trunk for outbound and inbound calls

Using **AMP (user: admin, pass: password)** select **setup** then **trunks**. Click on the type of trunk you want to create. See the VOIP Service Providers section for how to configure a trunk for your provider.

3.1.4 Configure Outbound Routing

Next you need a route to allow calls from your phones to go out on a trunk. If you have more than one trunk you can set up rules to determine how a trunk is chosen for each call. Here we will set all calls to go out one trunk.

Using **AMP (user: admin, pass: password)** select **setup** then **Outbound Routing**. Type in a **name** for your route. Then enter the following in the **dial pattern box**.

```
1NXXNXXXXXX  
NXXNXXXXXX  
NXXXXXX
```

This will set all calls to use this route.

Next go to the **Trunk Sequence** section. Drop down the box and **select the trunk you**

configured earlier. Click **add**.

That's it. Click **Submit Changes** and then click on the **red apply bar** at the top of the screen.

Try dialing a number on your phone like 19197543700 (RedHat customer support) and you should hear "Welcome to RedHat"

3.1.5 Configuring Incoming Calls

Next you need a route to allow calls from your provider to go someplace. Using **AMP (user: maint, pass: password)** select **setup** then **Incoming Calls**. Under the **Send Incoming Calls from the PSTN to: section** drop down the box next to **Extension:** and select the extension **200** you created earlier.

Click **Submit Changes** and then click on the **red apply bar** at the top of the screen.

Call the phone number from your provider with a cell phone or other non-IP phone. Your SIP phone at extension 200 should ring.

3.2 PSTN interface cards

There are many types of PSTN interface cards. These can include a single FXO card that allow 1 home telephone line to be connected to asterisk, or a T1 card that allows a digital trunk to be connected to Asterisk. All of these cards allow you to make calls directly on the Public Switch Telephone Network without have to use a VOIP phone service provider.

PSTN cards can also be used to connect PSTN (Non VOIP) phone to Asterisk. These can include a single FXS card that allow 1 home telephone to be connected to asterisk, or a T1 card that allows a Channel bank with 24 phones or a tie line to a PBX to be connected to Asterisk.

3.2.1 FXO Cards

These cards allow you to connect a POTS (plain Old Telephone System) line to your Asterisk@Home box.

3.2.1.1 Digium Wildcard X100P OEM FXO PCI Card

These voice modems are available on e-bay for about \$10.00 They are not made by Digium and are of low quality. Having said this they can work very well in many situations. If your phone line is well balanced there will work well. If it's not you can get some bad echos. We recommend them only for testing not for a production system.

To configure these cards for use with Asterisk@Home first use the zaptel card auto-config utility to set up the zaptel driver.

Type **genzaptelconf -s -d** from the command line (if the card was in the system when you installed aah this already done)

Next go into the AMP web interface and create a trunk.

There is already a trunk called ZAP/g0 edit this

Enter the **phone number** for you pots line in the **Caller ID field**

Enter **1** for **Maximum channels**

Set a dial rule if you want for this trunk

Select an outbound dial prefix to select this trunk when dialing

Set the **Zap Identifier** to **1** (the default is g0)

This trunk is now configured

you must add a route for incoming calls or asterisk will not answer this line

click on **incoming calls** in amp and set up an incoming route.

to make outbound calls you will need an outbound route. set one up in AMP

3.2.2 FXS Cards

These cards allow you to connect an Analog phone to your Asterisk@Home box.

3.2.2.1 Digium TDM400P FXO/FXS Card

This card has 4 module ports that can be loaded with FXS or FXO mudules. You can auto config this card just like the X100P. **Channel 1** is the **top RJ-45** on the back of the [TDM400P](#) card.

To configure these cards for use with Asterisk@Home first use the zaptel card auto-config utility to set up the zaptel driver.

Type **genzaptelconf -s -d** from the command line (if the card was in the system when you installed aah this already done)

Next, using config edit, look in the **zapata-auto.conf** file you will see a list of all your channels. Set up the trunks as trunks and the extensions as extensions in AMP.

For example if your zapata-auto.conf file looks like this

```
; Span 1: WCTDM/0 "Wildcard TDM400P REV E/F Board 1"
```

```
signalling=fxo_ks
```

```
; Note: this is an extension. Create a ZAP extension in AMP for Channel 1  
channel => 1
```

```
signalling=fxs_ks
```

```
; Note: this is a trunk. Create a ZAP trunk in AMP for Channel 2  
context=from-pstn  
channel => 2
```

then add a zap extension for channel 1 and a zap trunk for channel 2

you may have to reboot your system to get everything going.

you must add a route for incoming calls or asterisk will not answer your trunk

click on **incoming calls** in amp and set up an incoming route.

to make outbound calls you will need an outbound route. Set one up in AMP.

3.2.3 T1/PRI Cards

Setup for Digium T100P and a PRI

login to the CLI as root

```
[root@asterisk1 /]# cd /etc  
[root@asterisk1 etc]# nano -w zaptel.conf
```

Add these lines to /etc/zaptel.conf

```
span=1,1,0,esf,b8zs  
bchan=1-23 # set this to 1-15,17-31 for E1  
dchan=24 # set this to 16 for E1
```

Add # to front of fxsks=1 so it looks like this:

```
span=1,1,0,esf,b8zs  
bchan=1-23 # set this to 1-15,17-31 for E1  
dchan=24 # set this to 16 for E1
```

```
#fxsks=1  
loadzone = us  
defaultzone=us
```

Control-X then Y to save zaptel.conf

Log into Asterisk@home using a browser <http://-asterisk-ip-adress/admin>
Then click on **Maintenance**

Then click on **Config Edit**
Then click on **zapata.conf**

Comment out with ; to signalling=fxs_ks:

;signalling=fxs_ks

Then copy just below it:

```
signalling=pri_cpe ; pri_cpe = PRI slave ; pri_net = PRI master  
switchtype=national
```

Then add this callerid=asreceived under ;usedistinctiveringdetection=yes

Change echocancelwhenbridged=yes

Change echotraining=400 ; Asterisk trains to the beginning of the call, number is in milliseconds

at the end of the file copy and past:

```
channel => 1-23 ; Set this to 1-15,17-31 for E1
```

Click Update and Click Re-Read Config

Go back to SSH root

```
[root@asterisk1 etc]# modprobe wct1xxp  
[root@asterisk1 etc]# ztcfg -vv
```

Do a shutdown and restart the system

```
[root@asterisk1 etc]# shutdown -r now
```

You just need to login via web and make your setup.

3.3 VOIP Service Providers

There are many service providers. Some provide proxy server that make it possible to connect to other members of that provider. Other providers offer both incoming and outgoing PSTN to VOIP termination. Here are a few common providers and how to make the work with Asterisk@Home.

Most providers will give you phone number and a password for that provider some will also give you a user name. If you get a real PSTN number from the provider it will be a normal 10 digit number (US providers). some providers give out shorter number that can only be used by other members of that provider.

3.3.1 Free World Dialup (FWD)

Contact: <http://connect.voicepulse.com/>

Service: proxy to other FWD users, Gateway to other providers

Protocol: SIP or IAX

Cost: free

You should have a phone number (123456) and a password (wibble). You also need to have your FWD account setup for IAX. This is achieved by visiting <http://www.freeworlddialup.com>, logging in and turning on IAX. It does take a little bit of time to be set up (10 mins or so), so do that first. Once you've turned it on and clicked 'Submit' enough times (I noticed I had to click Submit two or three times before it came up with 'Changes Successful', that may have just been a temporary glitch) you're ready to proceed below.

Once again, you need to be in AMP, the Asterisk Management Portal. Click on **Setup** up the top, but this time click on **Trunks** on the left. Click on **Add IAX2 Trunk**

Dial Prefix is, usually, 393 — That's 'FWD' on your phones pad.

Leave **Default Trunk** switched off

Outbound Caller ID should (but doesn't have to be) set to your FWD Number. This is what is displayed when you call someone else through FWD. They'd normally just see your Extension (200).

Outgoing Settings

Trunk Name: fwd (This is just a descriptive name, and is what appears on the left of the screen)

PEER Details: (Change '123456' and 'wibble' to be your FWD Number and Password)

```
host=iax2.fwdnet.net
type=peer
username=123456
secret=wibble
```

Incoming Settings

USER Context: iaxfwd (Pay attention here. Don't change it. or it won't work)

USER Details (Nothing needs to be changed here, this can be pasted straight in)

```
allow=ulaw
auth=rsa
context=from-pstn
disallow=all
inkeys=freeworlddialup
type=user
```

Register String: should be set to yournumber:yourpassword@iax2.fwdnet.net, using our examples above, it would be **123456:wibble@iax2.fwdnet.net**

That's it. Click on **Submit Changes**, and then on the big red 'You have made changes' bar and you're done.

Assuming you've got your username and password correct, you should now be able to dial '393612', which will read out the time to you. If you're feeling exceptionally brave, call '393613', which is a useful little echo tester - it'll just bounce back to you everything you say to it. You can then try '393514' which is FWD's 'Coffee Lounge' - I've never actually successfully had a conversation with anyone there, however, or '39355555', which calls a random volunteer, so you can actually speak to a live person!

3.3.2 VoicePulse

Contact: <http://connect.voicepulse.com/>

Service: PSTN termination

Protocol: IAX

Cost: pay

Once again, you need to be in AMP, the Asterisk Management Portal. Click on **Setup** up the top, but this time click on **Trunks** on the left. Click on **Add IAX2 Trunk**

Dial Prefix 9 if you're not already

Leave **Default Trunk** switched off (or make this the default if you want all your calls to use it)

Outbound Caller ID should (but doesn't have to be) set to your VoicePulse Number.

Outgoing Settings

Trunk Name: voicepulse-out-01 (This is just a descriptive name)

PEER Details: (Change <your username> and <your password> to be your VoicePulse Number and Password)

```
host=gwiaxt01.voicepulse.com
secret=<your password>
type=peer
username=<your username>
```

Incoming Settings

USER Context: voicepulse-in-01 (Pay attention here. Don't change it. or it won't work)

USER Details (Nothing needs to be changed here, this can be pasted straight in)

```
auth=rsa
context=from-pstn
```

```
inkeys=voicepulse01
type=user
```

Register String: should be set to <your username>:<your password>@gwiax-in-01.voicepulse.com example (bob:abc123 @gwiax-in-01.voicepulse.com)

That's it. Click on **Submit Changes**, and then on the big red 'You have made changes' bar and you're done.

For a test make a call (try 1-800-555-1212)

3.4 Additional feature not installed by default

Some of the features of Asterisk@Home are not installed by default. Here are the steps to install them.

3.4.1 FAX to E-mail

Type **install-pdf** from the Linux command line. This will install support for encoding incoming faxes as PDFs and e-mailing them.

4 Phones

VOIP phones vary widely in price, features, and sound quality. Phones support either SIP or IAX protocol. For simple testing a free soft phone like X-ten is good. For day to day use a good hard phone is the best. Grandstream makes a cheap easy to configure phone with good quality and feature. Polycom and Cisco make the best phones with the highest sound quality and feature. If you have an existing analog phone like a cordless that you want to use and ATA will convert an analog phone to SIP.

4.1 Soft phones

Soft Phones run on a host computer and use the computer speaker and mic or a headset for voice. Soft phones are available for most operating systems.

4.1.1 X-Ten

One of the easiest to setup is X-Lite. X-Lite is available for free from <http://www.xten.com>.

If X-Lite cannot connect, the setup screen should open, if not, click on the "drop down" icon just to the left of the green Off-Hook icon.

Under **System Settings**, select the **SIP Proxy settings**, then double-click on the first entry. You will see the SIP Proxy settings as shown here. The settings I changed are as follows:

Username: 200 (my extension)
Authentication User: 200 (my extension again)
Password: 1111 (my extension password)
Domain/Realm: 192.168.5.50 (PBX IP address)
SIP Proxy: 192.168.5.50 (PBX IP address)



4.1.2 sipXphone

The [sipXphone project](http://www.sipfoundry.org/pub/sipXphone/win32/), formerly known as [Pingtel](http://www.pingtel.com/)'s instant xpressa soft phone, is a fully functional SIP soft phone that runs on Microsoft Windows and Linux. The Pingtel's xpressa phone (no longer a Pingtel product) is based on this same source base. Windows Download from <http://www.sipfoundry.org/pub/sipXphone/win32/>

4.2 Hard phones

4.2.1 Cisco 7960/7940

The Cisco SIP IP Phone software allows businesses and service providers to use the Cisco 7940 and 7960 IP Phone platforms in any standard SIP network. The SIP software provides for both on-board traditional desktop services such as Caller-ID, Call Hold, Call Transfer, 3-Way Calling, and Call Waiting as well as an XML interface to allow for enhanced web based services. The XML interface allows the phone to transcend the traditional phone paradigm and become a true Internet appliance. By supporting web browsing type functionality as well as allowing for application developers to directly control the user interaction on the phone and integrate tightly with the Cisco SIP phone, the Cisco SIP phone is a key enabler of enhanced and rapid application deployment in any SIP customer's network.

Type **setup-cisco** from the command line to set up a basic SIPDefault.cnf in the /tftpboot directory.

Copy your Cisco 7.4 SIP firmware into the /tftpboot directory using [WinSCP](http://www.winSCP.com/).

using the **Cisco config** tool in the maint menu to set up your phone.

4.2.1.1 Setting up a Cisco phone step by step

Connecting a Cisco 7960/7940 to your Asterisk@Home. The easiest way to use these phones with Asterisk@Home is with the SIP firmware.

Power on your phone and connect it to your network. You will need to change the IP address of your phone to a free one on your network or enable DHCP if you have a DHCP server.

Unlock your phone by pressing ****#** (for old firmware) or selecting **unlock config** from the **config menu** and type in the password (default is '**cisco**')

Change the IP address or enable DHCP. Enter the IP address of your Asterisk@Home box in the TFTP server field. If DHCP is enabled you will have to select **alternate TFTP server = yes** so you can edit the TFTP field.

Get the latest firmware for the Cisco phone (7.4) and copy the files firmware files to your Asterisk@Home box. Put them in the **/tftpboot** directory.

Create a Default.cnf file by using the setup-cisco command and change the file permissions on the firmware files that you uploaded in the last step. Type the following.

```
setup-cisco
chmod 777 /tftpboot/*
```

Add a Cisco config file for your phone. Use the web GUI and click **maintenance -> Cisco Config -> Create** a new phone configuration file.

Type in the **mac address** of your phone. It's located on the **bottom of your phone**. (**starts with 00**) Enter a name for your phone in the Phone Label box. Enter **200** in the **Line Name** and the **Display Name** fields for **Line 1**. Click **add**.

Click on **Edit phone configuration files** and click **edit** next to your phone. Go to **Line 1** and type 200 in the **Auth Name** box. Then enter a password like abc123 in the **password box**. Click **change**.

Create extension 200. **setup -> extensions** use **abc123** for the **extension password**.

Reboot your phone by disconnecting the power or pressing ***-6-settings**.

Your phone should upload the new firmware and extension. Make a test call like ***65**.

Trouble shooting: The Cisco phone can be difficult to upgrade. If your phone does not seem to upload the firmware try this. From command line.

```
cp /tftpboot/cisco_util/* /tftpboot
```

Then reboot the phone. After the phone has upgraded the firmware. Type the following so the phone will **boot normally**.

```
rm /tftpboot/xmlDefault.CNF.XML
rm /tftpboot/XMLDefault.cnf.xml
```

4.2.2 Polycom

<http://www.voip-info.org/wiki-Polycom+Phones>

5 Configuring and using Asterisk@Home features

5.1 Conferencing

Each extension you create will have its own conference. For example extension 200 has conference 8200. Just dial 8200 to get into your conference. If you dial from 8200 you are the moderator.

You need to set a password for the meetme user. Type passwd-meetme from the Linux command line.

Once you have started a conference you can manage it from the WebMeetme page. Just enter the extension for the conference.

External access for conferences is disabled for security reasons. It is not always desirable for people outside the system to have access to conferences

5.1.1 External access to conferences

If you want dial in access to conferences you need to be add an option on one of the auto attendants for this.

First go into the **Maintenance** page and select **Edit configs**. The file you want to edit is **extensions_custom.conf** add these lines to the bottom of the file. **Update** the file and **reload** configs.

```
[custom-meetme]
include => ext-meetme
exten => s,1,BackGround(enter-conf-call-number)
exten => h,1,Hangup()
```

Next go into **setup** and **Create a Digital receptionist** if you have not already done so. Record a main message for your receptionist. Something like "press # for a company

directory or press 1 to enter a conference" click continue and Select 1 for number of menu options.

Next you will have to select an **Action** for this menu item. Click **Custom App** and type in **custom-meetme,s,1** then click continue.

That should be it when you dial in you select 1 from the main menu and you will be prompted for your conference number. You can dial **7777** from and extension to simulate an incoming call.

5.2 AMP (Asterisk Management Portal)

AMP is a web GUI that allows the configuration of Asterisk without the need for editing config files. If you use AMP be careful not to make manual changes to the config files that will affect AMP's operation. You can learn more about AMP from their web site. <http://amp.coalescentsystems.ca/>

5.3 DHCP server

Asterisk@Home has a built in DHCP server. This can be used to configure your IP phones. The DHCP server is disabled by default. To make it active type **setup-dhcp** from the Linux command line. You must then **edit** the **/etc/dhcp.conf** file and set it to match your network. On line 4 set up your network address and netmask. On the next few lines enter the setting you want sent to your clients. Edit line 15 and add a range of IP address to give out.

Remember to disable any other DHCP servers you have on your network. These may include a Linksys router or Windows internet sharing PC. Reboot your [Asterisk@Home](#) system.

5.4 Flash Operator Panel

Flash Operator Panel is a real-time web interface to Asterisk. You can see what all of your extensions, trunks, and conferences are doing. We have extension 200 set up for testing. This is a live extension and will work as soon as a phone is registered to ext 200. The rest of the buttons are dummy objects for examples.

The `op_buttons.conf` file has all of the configs for each button in Flash Operator Panel you can edit this using the Config Edit button on the maint menu. You must reboot the Asterisk or restart the panel service after you make changes. The normal config files for Flash Operator Panel end in `.cfg` we changed them to `.conf` so they would match the other Asterisk files.

See the Flash Operator Panel we site for more info on Flash Operator Panel. <http://www.>

5.5 FTP server (vsftpd)

Asterisk@Home has an FTP server that is used primarily for configuring phones that use FTP such as Polycom. For security reasons only one user is configured by default to user ftp. This user is ftpuser (password: asteriskftp). It can access file in the /var/ftp directory. If you would like other users to have ftp access add them to the /etc/vsftpd.user_list file.

5.6 Cisco XML Services

Asterisk@Home has a Cisco XML Services that can be used to look up contacts in a database. Click on CRM from the Asterisk@Home main menu. (user: admin password: password) and enter all of your contacts and their phone numbers.

Use the **setup-cisco** command to set up SIPDefault.cnf for your Cisco phones. The setup-cisco script will set your Services URL to <http://ipaddressofyourasteriskbox/cisco/directory/services.php> you can then use the Services button of your Cisco phone to call your contacts.

5.7 Music On Hold (mpg123)

Asterisk@Home has now uses mpg123 for music on hold. Put a call on hold and they hear music not dead air! See voip-info.org for more info.

[www.voip-info.org music on hold info](http://www.voip-info.org/music-on-hold-info)

5.8 Fax support (SpanDSP)

SpanDSP allows Asterisk to receive faxes. This service must be installed. Type **install-pdf** from the command line. You can set this up in AMP.

5.9 xPL

xPL is a very simple and powerful home automation protocol. It can be used it interface Asterisk into an existing home automation system. For example if you get a call your [TiVo](http://www.tivo.com) can display the caller ID info on your TV screen and turn down the stereo.

If you have a Microsoft based home automation system see the xPL project page (<http://www.xplproject.org.uk/>).

If you have a fully Linux based system MisterHouse (<http://www.misterhouse.net>) has a

good xPL interface.

One of the best features of xPL is the ability to mix and match applications. A few Windows apps, MisterHouse on Linux, Asterisk@Home on CentOS, and a TiVo can all be on the same network and share xPL information.

xPL is not running by default. To enable it type `cd /usr/src/xplast` then type `./install.sh`

Once xPL is installed and running it will send out xPL heartbeat messages. xPL will also send out notification of new voicemail messages. xPL can also send out callerid info for any incoming calls. you must add a line to `extensions.conf` to make it do this. Add this line to `extension.conf` at a location where you want the callerid sent out.

```
exten => s,1,AGI,xplring.agi
```

For example a good place to add this line is to the `[from-pstn-timecheck]` section. Be sure to renumber the following lines.

```
[from-pstn-timecheck]
exten => .,1,Goto(s,1)
exten => s,1,AGI,xplring.agi
exten => s,2,GotoIf($[${IN_OVERRIDE} = forcereghours]?from-pstn-reghours,s,1:)
exten => s,3,GotoIf($[${IN_OVERRIDE} = forceafthours]?from-pstn-afthours,s,1:)
exten => s,4,GotoIfTime(${REGTIME}|${REGDAYS}|*|*?from-pstn-reghours,s,1:)
exten => s,5,Goto(from-pstn-afthours,s,1)
```

5.10 Sugar CRM

[SugarCRM](#) is a complete CRM (Customer Relation Management) system. Although most of the feature of this software package are beyond the needs of home users the contact management module in this package is the best we have ever seen in a web application. SugarCRM also has sophisticated user management.

Asterisk@Home uses SugarCRM mainly to manage contacts. SugarCRM is integrated into the Cisco XML phone directory making it possible to look up CRM contacts from a Cisco phone.

We have also added a **click-to-dial** function to CRM so that you can dial contacts directly from the web interface.

The default account for CRM is **admin** with a password of **password**. Once you have logged in you can create CRM users. You should make a CRM user for each extension. Enter the extension of a phone when you create a user so Asterisk will know where to send click-to-

dial calls for that user. **Use standard Asterisk notation for extensions.** For example for sip extension 200 enter **SIP/200** in the extension field.

6 Asterisk@Home add-ons

These add-on packages are not included with the Asterisk@Home iso. They can be installed to add functionality to your Asterisk@Home system.

6.1 Click-to-Dial using Microsoft Outlook and AstTapi

[AstTapi](#) is a [Microsoft TAPI](#) to Asterisk bridge that makes it possible to do click-to-dial from Microsoft Outlook and other TAPI compliant applications.

Download it from <http://sourceforge.net/projects/asttapi/>

First make a login for AstTapi. Use the **config editor** to edit **manager_custom.conf**. There is a default AstTapi account you can use. Just remove the **#** from the **permit** line and change the **192.168.1.0** to the **network address** of your network. Then reload Asterisk.

Install AstTapi

In outlook click on a contact and click on the phone icon. Select a phone number. In the "**new call**" dialog box select dialing options. In the "**connect using line**" box drop down and select Asterisk. Click on line properties.

Host: <ip of Asterisk server >

Port: 5038

User: AstTapi

Password: AstTapi

User Channel: sip/200 (your extension)

Select **dial by context**

Enter **outbound-allroutes** in the context box

Click Apply

You can now make calls from Outlook!

6.2 H.323 add-on

This package adds H.323 support to Asterisk it also install the GnuGK H.323 gatekeeper.

Installation

Copy the [asteriskathome-h323.zip](#) file to you Asterisk@Home server using [WinSCP](#). Unzip the file by typing

```
unzip asteriskathome_h323.zip
```

from the command line. Next type

```
./install.sh
```

When the install is done reboot your Asterisk@Home system.

Testing

register a SIP phone with Asterisk@Home

open Microsoft Netmeeting click on tools -> options -> Advanced calling
Under Gateway Settings check off Use a gateway and enter the ipaddress of your Asterisk@Home system.
Click OK

The calls from the gatekeeper will be put in the **from-pstn context**. Make sure you have an incoming call route set up so the calls have some place to go.

go back to the main NetMeeting screen and type in the extension of your sip phone. You should be connect to your incoming call route.

For support try

The asterisk-oh323 project home page
<http://www.inaccessnetworks.com/projects/asterisk-oh323>

GnuGK home page
<http://www.gnugk.org/>

6.3 Webmin - web based Linux management

Webmin in a great package for managing a Linux box from the web. Webmin make it easy to configure application like SMTP mail. To install Webmin download the latest RPM from their web site and install it.

<http://www.webmin.com/>

Example

```
wget http://unc.dl.sourceforge.net/sourceforge/webadmin/webmin-1.220-1.noarch.rpm  
rpm -Uvh webmin-1.220-1.noarch.rpm
```

7 Non-standard Hardware

Asterisk@Home is designed to work "out of the box" with standard PC hardware such as an Intel 1GIG PIII system. Here is some info on how to get it to work on other hardware.

7.1 VIA EPIA 800 mhz board (586 board)

Asterisk@Home can run on 586 hardware. you must change the Asterisk Make file and recompile.

Load Asterisk@Home on your system. Asterisk will not start.

```
CD to /usr/src/asterisk and edit the Makefile  
Remove the # in front of the line "PROC=i586"  
type  
cd ..  
then  
./rebuildastsrc.sh
```

8 Useful Info for Asterisk@Home

8.1 Feature Codes - List

- *411 Directory
- *43 Echo Test
- *60 Time
- *61 Weather
- *62 Schedule wakeup call
- *65 festival test (your extension is XXX)
- *70 Activate Call Waiting (deactivated by default)
- *71 Deactivate Call Waiting
- *72 Call Forwarding System
- *73 Disable Call Forwarding
- *77 IVR Recording
- *78 Enable Do-Not-Disturb
- *79 Disable Do-Not-Disturb
- *90 Call Forward on Busy
- *91 Disable Call Forward on Busy
- *97 Message Center (does no ask for extension)
- *98 Enter Message Center
- *99 Playback IVR Recording
- 666 Test Fax
- 7777 Simulate incoming call

If you found this documentation helpful please consider making a donation to the Asterisk@Home project:

http://sourceforge.net/donate/index.php?group_id=123387

Asterisk^{*}@Home Handbook