NEWSFLASH

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BUILD YOUR OWN PBX

User Rating: / 41

Poor Best Rate

Written by Kerry Garrison

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What would it mean to you to have your own full-featured PBX system at your home or small office? What would it mean to you if you could build an entire PBX system (minus the phones) on hardware you probably have laying around, AND that it can probably also save you money on your phone bill? Sounds too hard to believe doesn't it, but using old hardware and some open source software, you really can build a commercial quality phone system that would normally cost thousands of dollars.

The Hardware

As I mentioned in the opening, we are going to build our PBX system from equipment that we have laying around the house. After cannibalizing three spare systems, what was left was a PII 450, 386mb RAM, 12gb HD, 48x CDROM drive, and an Intel Pro 10/100 network card. This is all you "need" to get going as long as you are going to get VOIP dial tone service from a company like BroadVoice (more on this later). If you want to use regular analog phone lines you will need modem card. Not every card will work properly, however, the most recommended card is the Digium Wildcard X100P FXO card which can be purchased brand new on eBay for $6.95 each. So far, total out of pocket expense for the card plus shipping: $12.90.

The Software

The software for our PBX system is the open source package called Asterisk. When I said that this was a full-featured PBX system, I wasn't kidding. The following is NOT a complete list of features:

Asterisk@Home

- ADSI On-Screen Menu System
- Authentication
- Automated Attendant
- Blacklists
- Blind Transfer
- Call Forward on Busy
- Call Forward on No Answer
- Call Monitoring
- Call Parking
- Call Recording
- Call Retrieval
- Call Routing (DID & ANI)
- Call Transfer
- Call Waiting
- Caller ID
- Conference Bridging
- Distinctive Ring
- Do Not Disturb
- E911
- Interactive Directory Listing
- Interactive Voice Response (IVR)
- Music On Hold
- Music On Transfer
- Predictive Dialer
- Overhead Paging
- Remote Call Pickup
- Remote Office Support
- Roaming Extensions
- Route by Caller ID
- Spell / Say
- Supervised Transfer
- Talk Detection
- Text-to-Speech (via Festival)
- Three-way Calling
- VoIP Gateways
- Voicemail

While Asterisk can run on numerous systems from Linux to even flash ROM for some LinkSys routers, we will focus on installing Asterisk on our salvaged equipment as simply as possible. To aid in our install, Asterisk@Home is a pre-package ISO image that automates the installation of Asterisk and adds a usable web interface to monitor and configure your system.

With a VOIP PBX system you have three basic means of providing access to the users (the phones).

- SIP Compliant Handsets ($70 - $500)
- PC Based SoftPhones (Free)
- ATA (Analog Telecommunications Adaptor) ($50 - $500)

Without having to spend any money on our technogeek special PBX system, we will set it up for the time being with X-Lite softphones.

Installation

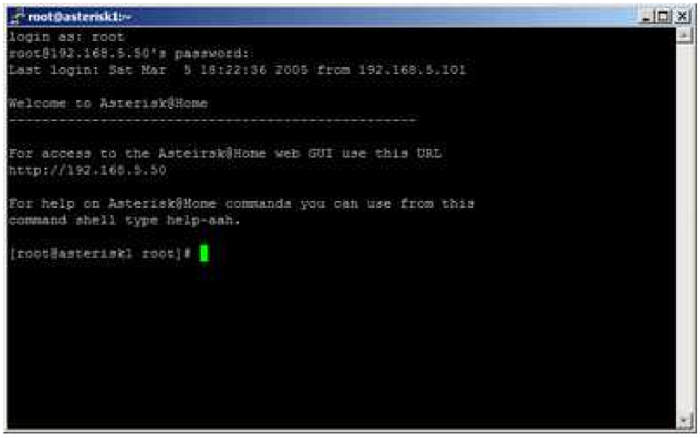
With Asterisk@Home, you simply need to download the disk image, burn it to a CD, and boot off of it.

- Burn Asterisk@Home iso to a blank CD
- Boot your Asterisk PC with the CD and press enter
- NOTE: This will erase all data on the hard drive of the PC!!!

After the Linux is loaded the CD will eject. Take out the CD and wait for the system to reboot

During the reboot Asterisk will be built from source for your hardware. This will take some time. Do not cancel the boot!

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



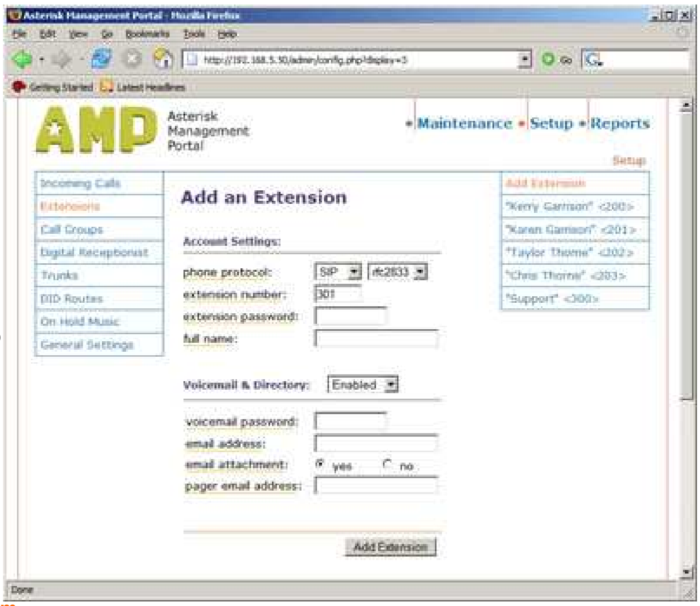
When you login, you will be given the URL to the web interface. You will also be told that you can use the help-aah command to get a list of quick commands. You can get into the Asterisk system for advanced settings by using asterisk -r command. We won't go into all of the advanced features of Asterisk in this article, that topic could consume an entire book.

If this machine is going to have any internet access, you should immediately change the root password with the passwd command. If you want to assign a static IP to the box, run the netconfig command. A simple interface will allow you to manually enter IP information. To configure the Wildcard X100P, simply run the setup script genzaptelconf script. This will set everything up for you.

The main menu presents you with the following choices:

- Web-access to Voicemail
- Web Address Book
- Flash Operator Panel
- Web MeetMe Control
- Asterisk Management Portal

To get things working, we only need to use the Asterisk Management Portal (AMP). To log in, use maint as the login and password as the password. At this point you will get the version number and links to Maintenance, Setup, and Reports.



Selecting Setup starts you off with the Incoming Calls setup screen. Before configuring this screen, you should start by adding at least one extension.

Click on the Extensions link, you should assign an extension number, extension password, and voicemail password.

Optionally you can enable email attachments and the email address to send them to. Once you have created an extension, there is a plethora of advanced options that are available by clicking on the name in the extension list. In most cases you will never need to touch any of the advanced extension options.

Before being able to record any messages, you will need to setup a handset or a PC-Based SoftPhone. One of the easiest to setup is X-Lite. X-Lite is available for free from **http://www.xten.com.**

Another good SoftPhone is SJPhone from SJ Labs (**http://www.sjlabs.com**) but for simplicity, we will focus on the setup of X-Lite.

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:

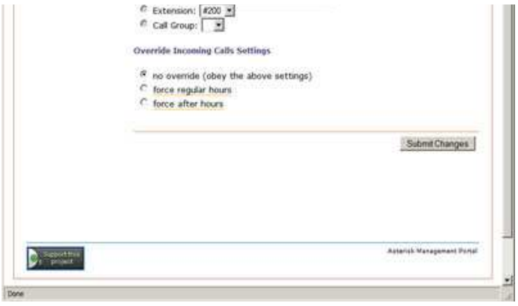
Username: 200 (my extension)
Authentication User: 200 (my extension again)
Password: 1111 (my extension password)
DomainRelm: 192.168.5.50 (PBX IP address)
SIP Proxy: 192.168.5.50 (PBX IP address)

Finally, you should record at least one message in the Digital Receptionist system.

Whenever you make a change, there will be a red bar on the screen that instructs you to click on it to apply the settings. Failure to apply the settings is an easy mistake to make and will keep your system from working properly.

With a extension created, an opening message in the

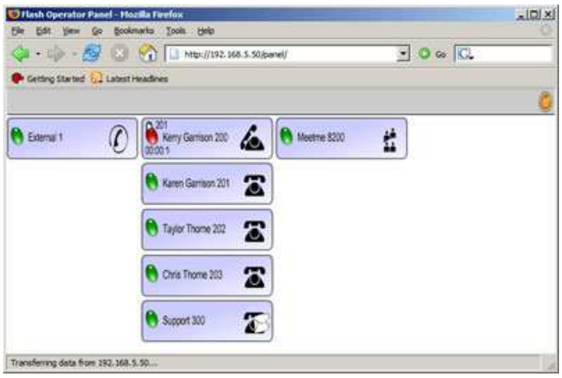
Digital Assistant, you are now ready to configure your incoming calls. For testing, I suggest not using different day/night modes (see image for example), make sure the radio button is selected for your opening message. You should now be setup and ready to go for you first calls.



Monitoring

To make sure things are setup, go back to the main menu and launch the Flash Operator Panel. This will display your trunk line status, extension status, and conference room status.

Our screen shot here shows that extension 200 is off the hook but nothing else is active so you can deduce that the person on that extension is probably checking voicemail. You can also see that extension 300 currently has an existing voicemail that is waiting to be read. In a business environment, I would setup the receptionist with a second monitor that had only this screen running on it so she could easily see the status of every line.



Common Commands

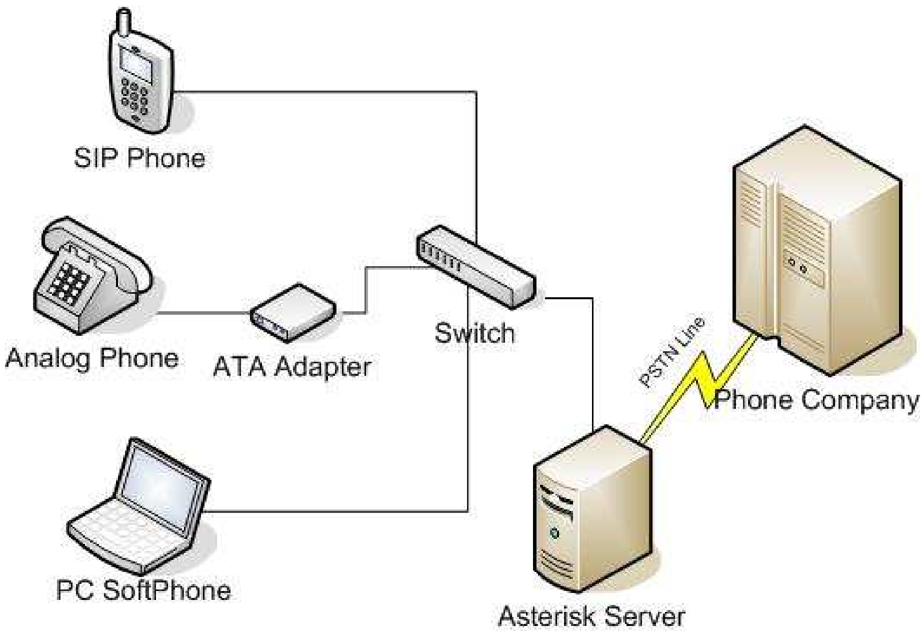
Below is a list of command Asterisk commands that will need to share with all of your users:

*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
*98	Enter Message Center
*99	Playback IVR Recording
7777	Simulate incoming call
1234	System will tell you your extension

With this information, you should be able to get your own PBX system up and running in less than an hour. While there are numerous functions and features built into Asterisk, covering more of them is not possible in the scope of an article like this.

Addendum

Some people didn't notice how this became a fully working system. Take note that I used a Digium modem card purchased for $6.95. This allowed me to plug in my existing analog phone line. If you called my phone number, you were greeted by the auto attendant. Dialing extension 200 rings the X-Lite SoftPhone on my laptop. The soft phone could easily have been replaced by a SIP phone or an analog phone with a SIP ATA Adapter. Below is a diagram of what this would look like:



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