VoIP fun with Asterisk

What is Asterisk? - taken from the asterisk website
Asterisk is a complete PBX in software. It runs on Linux and provides all of the features you would expect from a PBX and more. Asterisk does voice over IP in three protocols, and can interoperate with almost all standards-based telephony equipment using relatively inexpensive hardware.

Asterisk provides Voicemail services with Directory, Call Conferencing, Interactive Voice Response, Call Queuing. It has support for three-way calling, caller ID services, ADSI, SIP and H.323 (as both client and gateway). Check the Features section for a more complete list.

Asterisk needs no additional hardware for Voice over IP. For interconnection with digital and analog telephony equipment, Asterisk supports a number of hardware devices, most notably all of the hardware manufactured by Asterisk's sponsors, Digium™. Digium has single and quad span T1 and E1 interfaces for interconnection to PRI lines and channel banks as well as a single port FXO card and a one to four-port modular FXS and FXO card.

Also supported are the Internet Line Jack and Internet Phone Jack products from Quicknet.

Asterisk supports a wide range of TDM protocols for the handling and transmission of voice over traditional telephony interfaces. Asterisk supports US and European standard signalling types used in standard business phone systems, allowing it to bridge between next generation voice-data integrated networks and existing infrastructure. Asterisk not only supports traditional phone equipment, it enhances them with additional capabilities.

Using the Inter-Asterisk eXchange (IAX™) Voice over IP protocol, Asterisk merges voice and data traffic seamlessly across disparate networks. While using Packet Voice, it is possible to send data such as URL information and images in-line with voice traffic, allowing advanced integration of information.

Asterisk provides a central switching core, with four APIs for modular loading of telephony applications, hardware interfaces, file format handling, and codecs. It allows for transparent switching between all supported interfaces, allowing it to tie together a diverse mixture of telephony systems into a single switching network.

Asterisk is primarily developed on GNU/Linux for x/86. It is known to compile and run on GNU/Linux for PPC along with OpenBSD, FreeBSD, and Mac OS X Jaguar. Other platforms and standards based UNIX-like operating systems should be reasonably easy to port for anyone with the time and requisite skill to do so. Asterisk is available in the testing and unstable Debian archives, maintained thanks to Mark Purcell.

Installing

Installation is easiest if done via cvs.

```
# cd /usr/src
# export CVSROOT=:pserver:anoncvs@cvs.digium.com:/usr/cvsroot
# cvs login                    -the password is anoncvs.
# cvs checkout zaptel libpri asterisk
```

After you recieve the latest code from CVSup or CVS, issue the following commands as root to install Asterisk™ on your system:

```
# cd zaptel
# make clean; make install
# cd ../libpri
# make clean; make install
# cd ../asterisk
# make clean; make install; make samples
```

The make samples command installs the sample config files in /etc/asterisk
We will replace a couple of these to keep things simple

Sample Configs

****extensions.conf****

```
[general]

static=yes       ; These two lines prevent the command-line interface
writeprotect=yes ; from overwriting the config file. Leave them here.

[bogon-calls]

exten => _.,1,Congestion

[from-sip]
;Here is where we tell the system what to do when a number is dialed
;you can see the extension, step #, and the actual command.
exten => 1001,1,Dial(SIP/1001,20)
exten => 1001,2,Voicemail(u1001)
exten => 1001,3,Hangup

exten => 1002,1,Dial(SIP/1002,20)
exten => 1002,2,Voicemail(u1002)
exten => 1002,3,Hangup
```

****sip.conf****

```
[general]

port = 5060
localnet=192.168.1.0/24
;bindaddr = 192.168.1.2
disallow=all
```

```
;allow=g729
allow=ulaw
;allow=alaw
;allow=ilbc
;allow=g723
allow=gsm
allow=wav
;allow=speex
maxexpirey=33600

[1001]

type = friend
username = 1001
secret = stratitec
host = dynamic
context = from-sip
dtmfmode=rfc2833

[1002]

type = friend
username = 1002
secret = stratitec
host = dynamic
context = from-sip
dtmfmode=rfc2833
```

****voicemail.conf****

```
[general]
format=wav
maxmessage=180 ; Anything over 3 minutes is too long for me
; How many miliseconds to skip forward/back when rew/ff in message playback
skipms=3000
maxlogins=3
maxsilence=10            ; Wait for 5 silent seconds and end the voicemail
silencethreshold=128     ; What do we consider to be silence

[local]
1001 => 1234,Chris Locke,defiance@stageofbattle.org
1002 => 1234,Silent Bob,stageofbattle@stageofbattle.org
```

The configs above were very basic and let us set up two phones with voicemail. Below is a copy of the dialplan I use for work. It is much more complex but will let you get an idea of what we can do.

```
[general]

static=yes       ; These two lines prevent the command-line interface
writeprotect=yes ; from overwriting the config file. Leave them here.

[bogon-calls]
```

```
exten => _.,1,Congestion

[incoming]



exten => s,1,Wait,2 ; Allow for PRI to grab info in facility
exten => s,2,BackGround(new-greeting)
exten => s,3,BackGround(new-mainmenu)
exten => s,4,Wait,5
exten => s,5,BackGround(new-mainmenu)
exten => s,6,Wait,5
exten => s,7,Hangup


exten => 0,1,Goto(s,2)

exten => 1,1,SetCallerID(Toll Free No Cpub)
exten => 1,2,AGI(openclose.agi)
exten => 1,3,GotoIf($[${STATUS} = closed]?6:4)
exten => 1,4,GotoIf($[${STATUS} = holiday]?8:10)
exten => 1,5,Goto(1,10)
exten => 1,6,BackGround(nighttime-greeting)
exten => 1,7,Hangup
exten => 1,8,BackGround(holiday-greeting)
exten => 1,9,Hangup
exten => 1,10,Goto(tech-context,s,1)

exten => 2,1,Wait,1
exten => 2,2,SetCallerID(TimeIPS)
exten => 2,3,AGI(salesopenclose.agi)
exten => 2,4,GotoIf($[${STATUS} = closed]?7:5)
exten => 2,5,GotoIf($[${STATUS} = holiday]?9:11)
exten => 2,6,Goto(2,11)
exten => 2,7,BackGround(nighttime-greeting)
exten => 2,8,Hangup
exten => 2,9,BackGround(holiday-greeting)
exten => 2,10,Hangup
exten => 2,11,Goto(time-ips-sales,s,1)

exten => 5,1,Wait,1
exten => 5,2,SetCallerID(Sales)
exten => 5,3,AGI(salesopenclose.agi)
exten => 5,4,GotoIf($[${STATUS} = closed]?7:5)
exten => 5,5,GotoIf($[${STATUS} = holiday]?9:11)
exten => 5,6,Goto(5,11)
exten => 5,7,BackGround(nighttime-greeting)
exten => 5,8,Hangup
exten => 5,9,BackGround(holiday-greeting)
exten => 5,10,Hangup
exten => 5,11,Goto(sales-context,s,1)

exten => 7,1,Wait,1
exten => 7,2,Goto(2,1)

include => from-sip
include => tech-context
```

```
include => time-ips-sales
include => sales context

[sales]

exten => s,1,Wait,1
exten => s,2,SetCallerID(Sales)
exten => s,3,AGI(salesopenclose.agi)
exten => s,4,GotoIf($[${STATUS} = closed]?7:5)
exten => s,5,GotoIf($[${STATUS} = holiday]?9:11)
exten => s,6,Goto(1,11)
exten => s,7,BackGround(nighttime-greeting)
exten => s,8,Goto(1,28)
exten => s,9,BackGround(holiday-greeting)
exten => s,10,Goto(1,28)
exten => s,11,Dial(SIP/3006,15|m)
exten => s,12,WaitMusicOnHold(45)
exten => s,13,Dial(SIP/3006,15|m)
exten => s,14,WaitMusicOnHold(45)
exten => s,15,Dial(SIP/3006,15|m)
exten => s,16,Voicemail(u3006)
exten => s,17,Hangup
include => from-sip

[tech]

exten => s,1,Wait,2 ; Allow for PRI to grab info in facility
exten => s,2,SetCallerID(Toll CPUB/Modem)
exten => s,3,AGI(openclose.agi)
exten => s,4,GotoIf($[${STATUS} = closed]?7:5)
exten => s,5,GotoIf($[${STATUS} = holiday]?9:11)
exten => s,6,Goto(s,10)
exten => s,7,BackGround(nighttime-greeting)
exten => s,8,Goto(s,14)
exten => s,9,BackGround(holiday-greeting)
exten => s,10,Goto(s,14)
exten => s,11,BackGround(greeting)
exten => s,12,Playback(tech-intro)
exten => s,13,Goto(tech-context,s,1)
exten => h,1,Hangup
exten => t,1,Hangup
include => from-sip

[from-sip]

exten => 3001,1,Dial(SIP/3001,20)
exten => 3001,2,Voicemail(u3001)
exten => 3001,3,Hangup

exten => 3003,1,Dial(SIP/3003,20)
exten => 3003,2,Voicemail(u3003)
exten => 3003,3,Hangup

exten => 3005,1,Dial(SIP/3005,20)
exten => 3005,2,Voicemail(u3006)
exten => 3005,3,Hangup
```

```
exten => 3006,1,Dial(SIP/3006,20)
exten => 3006,2,Dial(SIP/3005,20)
exten => 3006,3,Voicemail(u3006)
exten => 3006,4,Hangup

exten => 3007,1,Dial(IAX2/3007,20)
exten => 3007,2,Voicemail(u3007)
exten => 3007,3,Hangup

exten => 3008,1,Dial(IAX2/3008,10)
exten => 3008,2,Dial(IAX2/3008,10)
exten => 3008,3,Voicemail(u3008)
exten => 3008,4,Hangup

exten => 3009,1,Dial(SIP/3009,20)
exten => 3009,2,Voicemail(u3009)
exten => 3009,3,Hangup

exten => 3012,1,Dial(SIP/3012,20)
exten => 3012,2,Voicemail(u3012)
exten => 3012,3,Hangup

exten => 3013,1,Dial(SIP/3013,20)
exten => 3013,2,Voicemail(u3013)
exten => 3013,3,Hangup

exten => 3014,1,Dial(SIP/3014,20)
exten => 3014,2,Voicemail(u3014)
exten => 3014,3,Hangup

exten => 3015,1,Dial(SIP/3015,20)
exten => 3015,2,Voicemail(u3015)
exten => 3015,3,Hangup

exten => 3030,1,Dial(SIP/3030,20)
exten => 3030,2,Voicemail(u3030)
exten => 3030,3,Hangup

exten => 3033,1,Dial(IAX2/3033,20)
exten => 3033,2,Dial(Zap/16/3048733,20)
exten => 3033,3,Voicemail(u3033)
exten => 3033,4,Hangup

exten => 3034,1,Dial(IAX2/3034,20)
exten => 3034,2,Dial(Zap/15/2046084,20)
exten => 3034,3,Voicemail(u3034)
exten => 3034,4,Hangup

exten => 3035,1,Dial(SIP/3035,20)
exten => 3035,2,Voicemail(u3035)
exten => 3035,3,Hangup

exten => 3037,1,Dial(SIP/3037,20)
exten => 3037,2,Voicemail(u3037)
exten => 3037,3,Hangup

exten => 3042,1,Dial(SIP/3042,20)
```

```
exten => 3042,2,Voicemail(u3042)
exten => 3042,3,Hangup

exten => 3043,1,Dial(SIP/3043,20)
exten => 3043,2,Voicemail(u3043)
exten => 3043,3,Hangup

exten => 3044,1,Dial(SIP/3044,20)
exten => 3044,2,Voicemail(u3044)
exten => 3044,3,Hangup

exten => 3045,1,Dial(SIP/3045,20)
exten => 3045,2,Voicemail(u3045)
exten => 3045,3,Hangup

exten => 3048,1,Dial(SIP/3048,20)
exten => 3048,2,Voicemail(u3048)
exten => 3048,3,Hangup

exten => 3050,1,Dial(SIP/3050,20)
exten => 3050,2,Voicemail(u3050)
exten => 3050,3,Hangup

exten => 3060,1,Dial(SIP/3060,20)
exten => 3060,2,Voicemail(u3060)
exten => 3060,3,Hangup

exten => 3061,1,Dial(SIP/3061,20)
exten => 3061,2,Voicemail(u3061)
exten => 3061,3,Hangup

exten => 3062,1,Dial(SIP/3062,20)
exten => 3062,2,Voicemail(u3062)
exten => 3062,3,Hangup

exten => 3063,1,Dial(SIP/3063,20)
exten => 3063,2,Voicemail(u3063)
exten => 3063,3,Hangup

exten => 3064,1,Dial(SIP/3064,20)
exten => 3064,2,Voicemail(u3064)
exten => 3064,3,Hangup

exten => 3081,1,Dial(SIP/3081,20)
exten => 3081,2,Voicemail(u3081)
exten => 3081,3,Hangup

exten => 3085,1,Dial(SIP/3085,20)
exten => 3085,2,Voicemail(u3085)
exten => 3085,3,Hangup


;this is to transfer to tech queue
exten => 3004,1,SetVar(QUEUE_PRIO=10)
exten => 3004,2,Goto(incoming,1,1)

;this is to transfer to sales
```

```
exten => 3016,1,Goto(sales-context,s,1)

;Here are the conference rooms
exten => 8600,1,Meetme(8600|M)
exten => 8601,1,Meetme(8601|M)

;this is for parking calls
include => parkedcalls
include => tech-context
include => incoming

;here is our intercom
exten => 6000,1,Dial,console/dsp
exten => 6000,2,Hangup

;voicemail
exten => 2999,1,VoicemailMain(${CALLERIDNUM})
exten => 2888,1,VoicemailMain(3043)


;tech login
exten => 91,1,AddQueueMember(tech)
exten => 91,2,Playback(agent-loginok)
exten => 91,3,Hangup

;tech logoff

exten => 92,1,RemoveQueueMember(tech)
exten => 92,2,Playback(agent-loggedoff)
exten => 92,3,Hangup

include => local-trunks
include => toll-access


[sales-context]

exten => s,1,Background(sales-menu)
exten => s,2,Dial(SIP/3006,15|m)
exten => s,3,WaitMusicOnHold(30)
exten => s,4,Dial(SIP/3006,15|m)
exten => s,5,WaitMusicOnHold(30)
exten => s,6,Dial(SIP/3007,15|m)
exten => s,7,Voicemail(u3006)
exten => s,8,Hangup

exten => 2,1,Goto(incoming,1,1)

include => from-sip
include => incoming

[tech-context]
exten => s,1,Queue(tech)
;exten => s,2,Dial(SIP/3048,15|m)
;exten => s,3,WaitMusicOnHold(30)
;exten => s,4,Dial(SIP/3044,15|m)
;exten => s,5,WaitMusicOnHold(30)
```

```
;exten => s,6,Dial(SIP/3043,20|m)
;exten => s,7,WaitMusicOnHold(30)
;exten => s,8,Dial(SIP/3045,20|m)
;exten => s,9,Background(sales-hold)
;exten => s,10,WaitMusicOnHold(60)
;exten => s,11,Dial(SIP/3048,15|m)
;exten => s,12,WaitMusicOnHold(30)
;exten => s,13,Dial(SIP/3045,15|m)
;exten => s,14,WaitMusicOnHold(30)
;exten => s,15,Dial(SIP/3044,15|m)
;exten => s,16,WaitMusicOnHold(30)
;exten => s,17,Dial(SIP/3043,15|m)
;exten => s,18,Background(sales-hold)
;exten => s,19,Goto(s,1)
exten => h,1,Hangup
exten => t,1,Hangup
include => from-sip
include => incoming

[time-ips-sales]

exten => s,1,Background(timeips-sales)

exten => 1,1,Goto(s,1)

exten => 5,1,Goto(tech-context,s,1)

exten => 2,1,Wait(1)
exten => 2,2,Dial(SIP/3006,15|m)
exten => 2,3,WaitMusicOnHold(45)
exten => 2,4,Dial(SIP/3006,15|m)
exten => 2,5,WaitMusicOnHold(45)
exten => 2,6,Dial(SIP/3006,15|m)
exten => 2,7,Voicemail(u3006)
exten => 2,8,Hangup

exten => 3,1,Wait(1)
exten => 3,2,Dial(SIP/3006,15|m)
exten => 3,3,WaitMusicOnHold(45)
exten => 3,4,Dial(SIP/3006,15|m)
exten => 3,5,WaitMusicOnHold(45)
exten => 3,6,Dial(SIP/3006,25|m)
exten => 3,7,Voicemail(u3006)
exten => 3,8,Hangup

include => from-sip
include => incoming

[local-trunks]


exten => _9NXXXXXX,1,Dial(Zap/3/${EXTEN:1})
exten => _9NXXXXXX,2,Dial(Zap/4/${EXTEN:1})
exten => _9NXXXXXX,3,Dial(Zap/5/${EXTEN:1})
exten => _9NXXXXXX,4,Dial(Zap/6/${EXTEN:1})
exten => _9NXXXXXX,5,Dial(Zap/8/${EXTEN:1})
exten => _9NXXXXXX,6,Dial(Zap/19/${EXTEN:1})
```

```
exten => _9NXXXXXX,7,Dial(Zap/23/${EXTEN:1})
exten => _9NXXXXXX,8,Dial(Zap/24/${EXTEN:1})
exten => _9NXXXXXX,9,Congestion
include => from-sip

[toll-free]

;exten => _91800NXXXXXX,1,Dial(Zap/13/${EXTEN:1})
;exten => _91800NXXXXXX,2,Congestion
;exten => _91888NXXXXXX,1,Dial(Zap/14/${EXTEN:1})
;exten => _91888NXXXXXX,2,Congestion
;exten => _91877NXXXXXX,1,Dial(Zap/15/${EXTEN:1})
;exten => _91877NXXXXXX,2,Congestion
;exten => _91866NXXXXXX,1,Dial(Zap/16/${EXTEN:1})
;exten => _91866NXXXXXX,2,Congestion


[toll-access]

exten => _91NXXNXXXXXX,1,Dial(Zap/3/${EXTEN:1})
exten => _91NXXNXXXXXX,2,Dial(Zap/4/${EXTEN:1})
exten => _91NXXNXXXXXX,3,Dial(Zap/5/${EXTEN:1})
exten => _91NXXNXXXXXX,4,Dial(Zap/6/${EXTEN:1})
exten => _91NXXNXXXXXX,5,Dial(Zap/8/${EXTEN:1})
exten => _91NXXNXXXXXX,6,Dial(Zap/19/${EXTEN:1})
exten => _91NXXNXXXXXX,7,Dial(Zap/23/${EXTEN:1})
exten => _91NXXNXXXXXX,8,Dial(Zap/24/${EXTEN:1})
exten => _91NXXNXXXXXX,9,Congestion
ignorepat => 9
include => from-sip
include => local-trunks
```

Links

The Asterisk Homepage - http://asterisk.org
Voip info wiki - http://voip-info.org
Digium (Asterisk developers and hardware makers) http://www.digium.com
Asterisk Users Mailing List - http://lists.digium.com/mailman/listinfo/asterisk-users
My Site (more sample configs coming soon) http://stageofbattle.org

You can email me questions
defiance at stageofbattle dot org