

# Asterisk - The Basics

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# What is Asterisk

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- Asterisk, *The Open Source PBX*. [www.asterisk.org](http://www.asterisk.org)
- A complete PBX in software
- Runs on virtually any OS
- Support for most VoIP protocols
- Most full-featured PBX features already built in
  - MOH, conferencing, queues, voicemail, IVR...
- Supports many different hardware telephony cards

# Asterisk Documentation

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- There's lots of info all over the place, some of it contrary though
- [www.voip-info.org](http://www.voip-info.org)
  - Lots of really good information, lots of plain wrong information too!
  - Defacto documentation store at this stage
- [www.asterisk.org](http://www.asterisk.org)
- [www.digium.org](http://www.digium.org) - hardware cards
- Asterisk CLI !

# Useful Reading

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- Asterisk, The Future of Telephony. By Jared Smith, Jim Van Meggelen, Leif Madsen. ISBN: 0-596-00962-3
  - Published under Creative Commons license
  - Can download, or buy a real book from O'Reilly
  - <http://www.asteriskdocs.org/modules/tinycontent/index.php?id=11>

# Asterisk Versions

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- Three versions currently in popular use:
  - 1.0 - pretty much obsolete now, but it's good and stable
  - 1.2 - the current release of choice for most, stable
    - We'll be dealing with v1.2
  - 1.4 - all the new features in here, still a few bugs
  - 1.6 - very latest version, still in development

# Installing Asterisk

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- Asterisk uses three main packages:
  - asterisk
  - zaptel
  - libpri
- Compile Requirements:
  - GCC (version 3.x or later)
  - Kernel source
  - Kernel headers
  - bison
  - openssl, openssl-dev, libssl-dev
  - libnewt

# Download Source

---

```
# cd /usr/src/  
# wget --passive-ftp ftp.digium.com/pub/asterisk/asterisk-1.*.tar.gz  
# wget --passive-ftp ftp.digium.com/pub/asterisk/asterisk-sounds-*.tar.gz  
# wget --passive-ftp ftp.digium.com/pub/zaptel/zaptel-*.tar.gz  
# wget --passive-ftp ftp.digium.com/pub/libpri/libpri-*.tar.gz
```

```
# tar zxvf zaptel-*.tar.gz  
# tar zxvf libpri-*.tar.gz  
# tar zxvf asterisk-*.tar.gz  
# tar zxvf asterisk-sounds*.tar.gz
```

\* If using Linux kernel 2.4 a symbolic link named linux-2.4 is required pointing to your kernel source:

```
#ln -s /usr/src/`uname -r` /usr/src/linux-2.4
```

# Compile Zaptel

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- Several features in Asterisk require an accurate timing source, e.g. conferencing
- Digium PCI hardware provides this 1kHz timing clock
- If you aren't using PCI hardware the *ztdummy* driver can be used
  - Kernels 2.4.5 and greater use the UHCI USB controller for this (so you need the *usb-uhci* module loaded)
  - The 2.6 kernel provides a 1kHz so a USB controller is not needed
- Need to uncomment out 'ztdummy' in Makefile

```
MODULES=zaptel tor2 torisa wcusb wcfxo wctdm \  
ztdynamic ztd-eth wct1xxp wct4xxp wcte11xp # ztdummy
```



# Compile Zaptel

---

```
# cd /usr/src/zaptel-version
# make clean
# make
# make install
# make config
```

- Also installs some tools:
  - *ztcfg* - reads config in */etc/zaptel.conf* to configure hardware
  - *zttool* - for monitoring installed hardware
  - *ztmonitor* - for monitoring active channels
- *zconfig.h* contains many zaptel compile-time options - echo cancellation options, RAS options, etc.

# Compile Libpri

---

```
# cd /usr/src/libpri-version  
# make clean  
# make  
# make install
```

- Used by many manufacturers of PCI TDM cards
  - Safe to compile even if a card is not installed/used

# Compile Asterisk

---

```
# cd /usr/src/asterisk-version  
# make clean  
# make  
# make install  
# make samples
```

# Package Install

---

- Much easier to use pre-compiled binary packages!
  - RPM packages for redhat
  - DEB packages for Debian
  - Asterisk.pkg for MacOSX <http://www.astmasters.net>
- We'll be using Debian .deb packages
  - Debain testing
  - Asterisk version 1.2

# Debian Install

---

```
apt-get install asterisk
apt-get install asterisk-sounds-extra
apt-get install zaptel
apt-get install zaptel-source
apt-get build-dep asterisk
    * if you need ztdummy:
    m-a prepare
    m-a build zaptel
```

```
dpkg -i zaptel-modules-xxxxxx.deb
depmod
modprobe zaptel
modprobe wctellxp      # if using TE110P single span T1/E1 card
modprobe wcfxo         # if using single port FXO card
modprobe ztdummy       # if using ztdummy
```

```
ztcfg
zttool
```

```
nano /etc/default/asterisk
* To get ztdummy, modify Makefile to uncomment 'ztdummy'
* On Debian, add 'ztdummy' to /etc/module to get ztdummy to load at boot
* set RUNASTERISK=yes in /etc/default/asterisk
```

# Asterisk File Locations (debian)

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- `/etc/asterisk/` - Asterisk configuration files
- `/var/lib/asterisk/` - contains the astdb, firmware and keys
- `/usr/share/asterisk/sounds` - in built asterisk sound prompts
- `/var/spool/asterisk/` - temporary files and voicemail files
- `/var/log/asterisk/` - Asterisk log files
- `/var/log/asterisk/cdr-csv/` - Asterisk call detail records

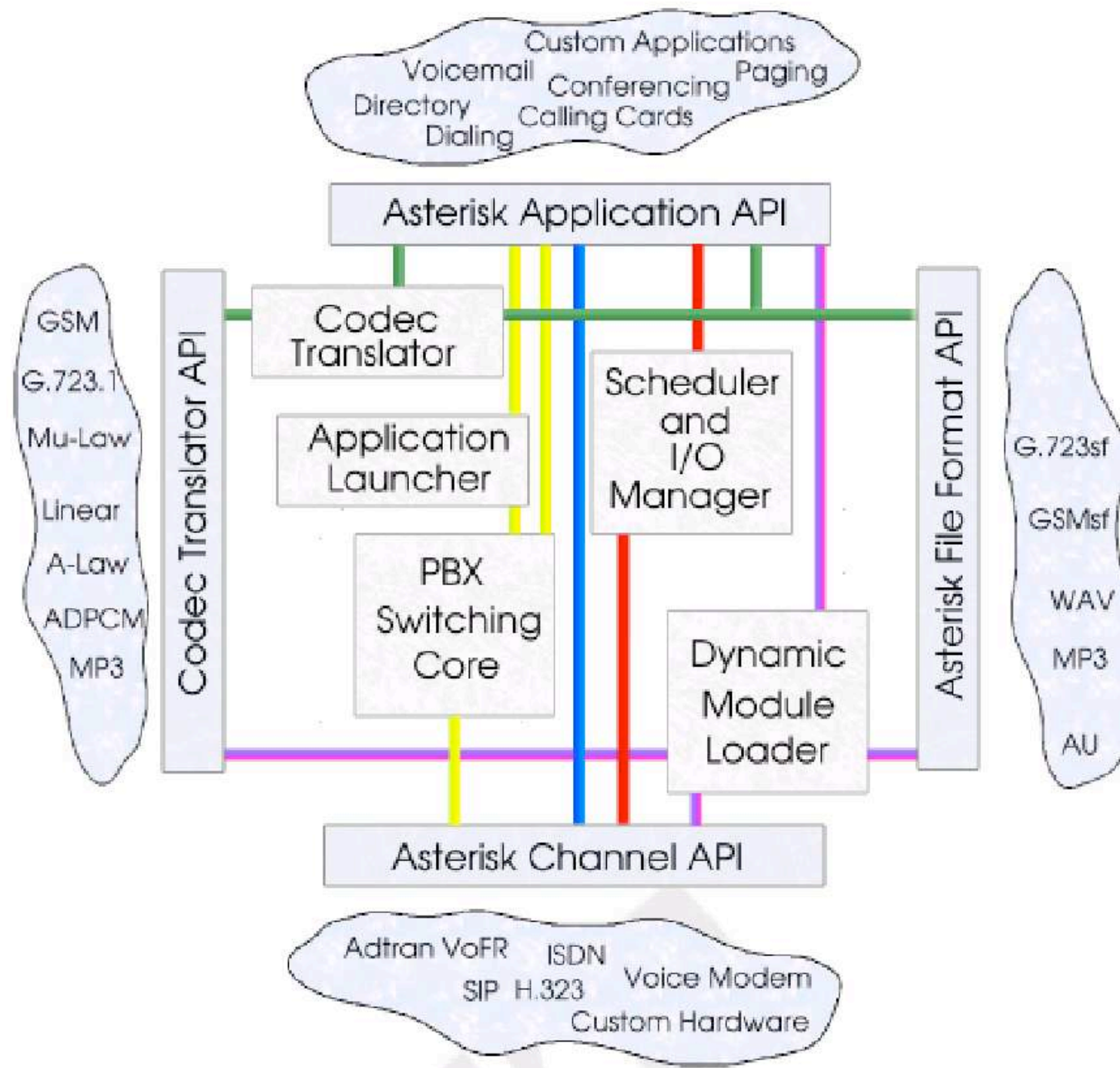
# How Asterisk Works, in one slide or less!

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- Asterisk is a hybrid TDM and packet voice PBX
- Interfaces any piece of telephony hardware or software to any application
- Prime components: channels and extensions.conf - the Asterisk dial plan
- Channels can be many different technologies - SIP, IAX, H323, skinny, Zaptel, and others as they are created
- extensions.conf is basically a programming language controlling the flow of calls
- Applications do the work - answer a channel, ring a channel, voicemail, etc.

# Asterisk Architecture

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# TrixBox

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- [www.trixbox.org](http://www.trixbox.org)
- Asterisk PBX up and running in one hour
- The PBX formally known as Asterisk@Home
- Latest version = 2.0, based on Asterisk 1.2
- Full featured PBX system including all the regulars:
  - Voicemail, conferencing, call forwarding, extensions
- Provides web based interface, which in turn drives Asterisk configuration files
- We'll be looking at this later in the workshop

# Asterisk Configuration Details

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- Text based configuration files
  - sip.conf
  - extensions.conf
  - voicemail.conf
  - agents.conf
  - queues.conf

# sip.conf

---

**/etc/asterisk/sip.conf**

**[general]**

<b>context=default</b>	<b>; Default context for incoming calls</b>
<b>port=5060</b>	<b>; UDP Port to bind to (SIP standard port is 5060)</b>
<b>bindaddr=0.0.0.0</b>	<b>; IP address to bind to (0.0.0.0 binds to all)</b>
<b>srvlookup=yes</b>	<b>; Enable DNS SRV lookups on outbound calls</b>

**[2000]**

<b>type=friend</b>	<b>; both send and receive calls from this peer</b>
<b>host=dynamic</b>	<b>; this peer will register with us</b>
<b>username=2000</b>	
<b>secret=j3nny</b>	
<b>canreinvite=no</b>	<b>; don't send SIP re-invites (ie. terminate rtp</b>
<b>stream)</b>	
<b>nat=yes</b>	<b>; always assume peer is behind a NAT</b>
<b>context=phones</b>	<b>; send calls to 'phones' context</b>
<b>dtmfmode=rfc2833</b>	<b>; set dtmf relay mode</b>
<b>allow=all</b>	<b>; allow all codecs</b>

# sip.conf ...ctd

---

```
[pstn-gateway]
type=friend
disallow=all
allow=alaw
context=from-pstn-gateway
host=pstn-gateway.jonnynet.net
canreinvite=no
dtmfmode=rfc2833
allow=all
```

# extensions.conf

---

```
/etc/asterisk/extensions.conf
```

```
[general]
```

```
static=yes                ; default values for changes to this file  
writeprotect=no           ; by the Asterisk CLI
```

```
[globals]
```

```
; variables go here
```

```
[default]
```

```
; default context
```

```
[from-pstn-gateway]
```

```
; context for calls coming from wlg-gateway
```

```
exten => 4989560,1,GoTo(phones,2000,1)
```

```
exten => _.,1,Congestion()           ; everyone else gets congestion
```

# extensions.conf ...ctd

---

**[phones]**

**; context for our phones**

**exten => 2000,1,Dial(SIP/2000)**

**exten => 2000,2,VoiceMail(u2000)**

**exten => 500,1,Answer()**

**exten => 500,2,Playback(demo-echotest)**

**; Let them know what's going on**

**exten => 500,3,Echo**

**; Do the echo test**

**exten => 500,4,Playback(demo-echodone)**

**; Let them know it's over**

**exten => 500,5,Hangup**

**exten => \_1.,1,Dial(SIP/\${EXTEN:1}@pstn-gateway) ; match all and send to wlg-gateway**

**exten => \_1.,2,Hangup**

# Dial Plan - Contexts

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- extensions.conf split into sections called contexts
- [context-name]
- contexts isolated from one another - can have the same extension in multiple contexts
- Calls from a channel land in the context specified by that channel,
- Calls land in default context if nothing is specified
- Be careful with what is in the default context - it is easy to give access to more than is intended

# Dial Plan - Extensions

---

- One or more extensions in each context
- An extension is followed by an incoming call or digits dialled on a channel
  - `exten => name,priority,application()`
  - `exten => 2000,1,Dial(SIP/2000)`
- Priorities are numbered and followed sequentially from '1'
  - Asterisk will stop processing an extension if you skip a priority
- Each priority executes one specific application



# Dial Plan - Applications

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- Applications 'do things' in the Asterisk dial plan
  - play a sound
  - answer a call
  - interact with a database
- Can take zero or more arguments
  - Answer()
  - Dial(SIP/2001)
  - AnApplicationWithThreeArguments(arg1,arg2,arg3)
- Arguments can be separated with a pipe ( | ) or a comma ( , ).

# Dial Plan - 'n' priority

---

- Asterisk 1.2 onwards understands the 'n' priority

exten => 2000,1,FirstApplication()

exten => 2000,n,NextApplication()

exten => 2000,n(priority\_label),AnotherApplication()

- Saves renumbering your extensions if you add or remove a priority
- Labels can make dial plan more readable, particularly when branching using gotos.

# Dial Plan - Variables

---

- Three types of variables available in the dial plan
- Global
  - Set in the [globals] section of extensions.conf
- Channel
  - Variables set automatically, and using the set command on a per channel basis
- A number of pre-defined channel variables - e.g. \${EXTEN}

# Dial Plan - Variables

---

- Some of the common pre-defined channel variables:

`${CALLERID}`

`${CALLERIDNAME}`

`${CALLERIDNUM}`

`${CHANNEL}`

`${CONTEXT}`

`${EXTEN}`

`${SIPUSERAGENT}`

# Dial Plan - Extension Matching

---

- `exten => _04NXXXXXX,1,SomeApplication()`
- `exten => _.,1,SomeApplication()`
  - `_` denotes a pattern matching extension
  - `N` matches any number from 2 through 9
  - `X` matches any single digit
  - `.` matches one or more of any digit
  - `[2-6]` matches any of 2,3,4,5,6

# Dial Plan - Extension Matching

---

- Examples
  - `_02[1579].`
    - matches NZ mobiles, i.e. numbers starting in 021, 025, 027, or 029
  - `_027NXXXXXX`
    - matches numbers starting in 027 and exactly 10 digits long, where the fourth digit is from 2 - 9

# Starting Asterisk

---

- On Debian systems:
  - `/etc/init.d/asterisk start`
- Or, `/usr/sbin/asterisk`
  - `asterisk -c` if you want asterisk to load straight into a console
- To connect to a running instance of Asterisk:
  - `asterisk -r`

# Running Asterisk

---

```
jonny@collins:~# asterisk -h
```

```
Asterisk 1.0.7-BRistuffed-0.2.0-RC7k, Copyright (C) 2000-2004, Digium.
```

```
Usage: asterisk [OPTIONS]
```

```
Valid Options:
```

-V	Display version number and exit
-C <configfile>	Use an alternate configuration file
-G <group>	Run as a group other than the caller
-U <user>	Run as a user other than the caller
-c	Provide console CLI
-d	Enable extra debugging
-f	Do not fork
-g	Dump core in case of a crash
-h	This help screen
-i	Initialize crypto keys at startup
-n	Disable console colorization
-p	Run as pseudo-realtime thread
-q	Quiet mode (suppress output)
-r	Connect to Asterisk on this machine
-R	Connect to Asterisk, and attempt to reconnect if disconnected
-t	Record soundfiles in /var/tmp and move them where they belong

```
after they are done.
```

-v	Increase verbosity (multiple v's = more verbose)
-x <cmd>	Execute command <cmd> (only valid with -r)



# Running Asterisk

---

```
jonny@collins:~# asterisk -r
Asterisk 1.0.7-BRIstuffed-0.2.0-RC7k, Copyright (C) 1999-2004 Digium.
Written by Mark Spencer <markster@digium.com>
=====
Connected to Asterisk 1.0.7-BRIstuffed-0.2.0-RC7k currently running on collins (pid
= 10763)
collins*CLI>
```

# Asterisk CLI

---

- Similar to IOS:
  - sip show peers
  - reload
  - ? for help, tab for command autocomplete
  - sip show ?
- Restart commands
  - restart gracefully: Restart Asterisk gracefully
  - restart now: Restart Asterisk immediately
  - restart when convenient: Restart Asterisk at empty call volume
  - reload: Reload configuration
- stop gracefully: Gracefully shut down Asterisk
- stop now: Shut down Asterisk immediately
- stop when convenient: Shut down Asterisk at empty call volume

# Asterisk CLI

---

- sip debug: Enable SIP debugging
- sip no debug: Disable SIP debugging
- sip reload: Reload sip.conf
- SIP Show commands
  - sip show channels: Show active SIP channels
  - sip show channel: Show detailed SIP channel info
  - sip show inuse: List all inuse/limit
  - sip show peers: Show defined SIP peers (clients that register to your Asterisk server)
  - sip show registry: Show SIP registration status (when Asterisk registers as a client to a SIP Proxy)
  - sip show users: Show defined SIP users

# Soft Phone Client

---

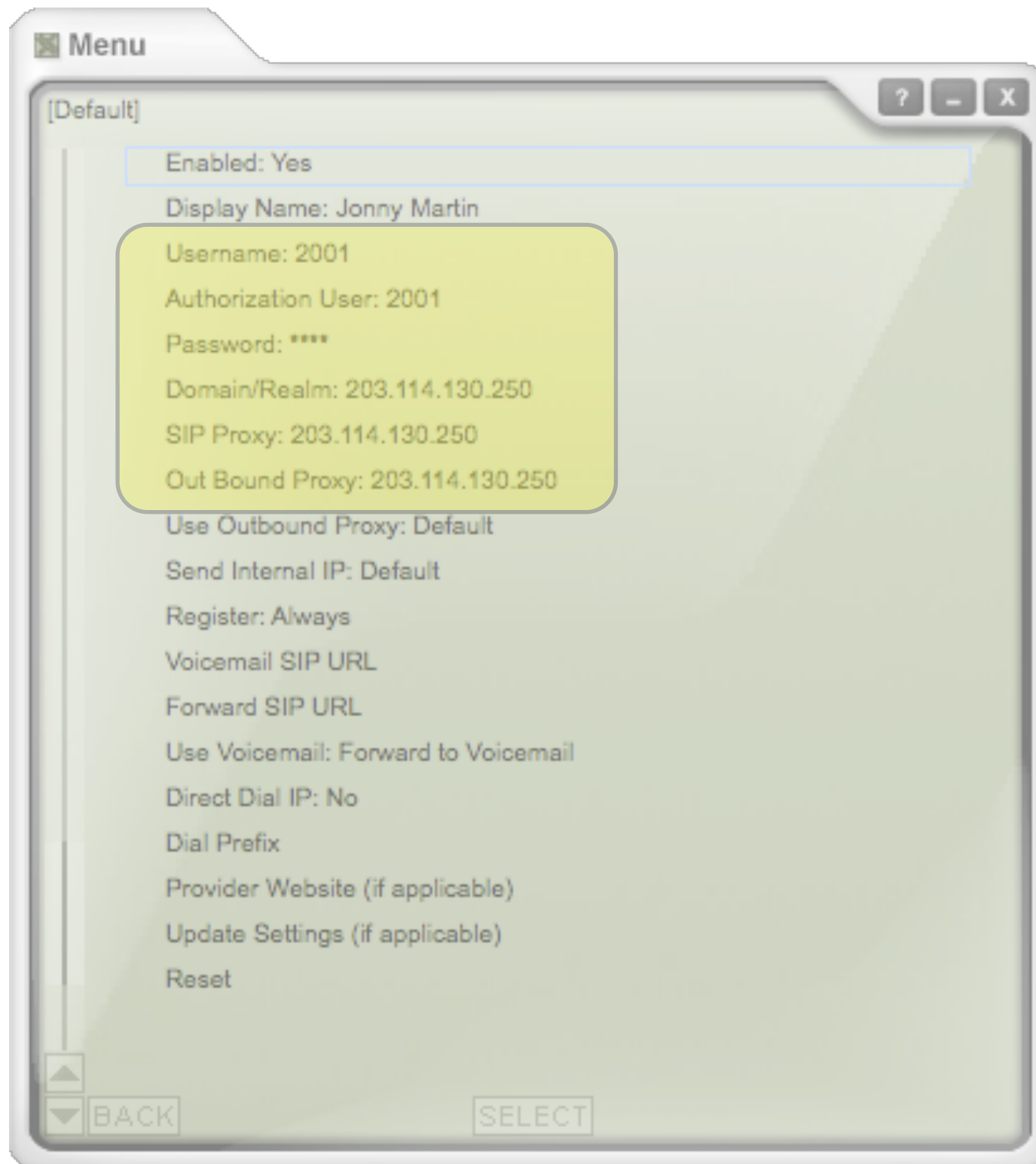
- Any SIP client can be used for the lab
- We'll use the Xten Xlite client
  - Works on Win, Mac, Linux
  - <http://www.xten.com/index.php?menu=download>
- You can use a Wifi phone or similar if you have one with you

# Xlite Softphone Setup

---

- Only need to set a few basic parameters
  - SIP username
  - SIP password
- This is done in
  - Main Menu > System Settings > SIP Proxy > Default

# Xlite Softphone Setup



# Lab 1: Initial Asterisk Install

# Asterisk Variables

---

- Why use variables?
  - Pattern match - how do we know what extensions was dialled?

exten => 2000,1,Dial(SIP/2000)

exten => 2001,1,Dial(SIP/2001)

exten => 2002,1,Dial(SIP/2002)

- OR

exten => \_200X,1,Dial(SIP/\${EXTEN})

- \${some\_variable} = the value of some\_variable.
- some\_variable = the variable itself



# Asterisk Variables

---

- Set default variables

[globals]

default\_ring\_time=10

[context]

exten => 2000,1,Dial(SIP/2000,\${default\_ring\_time})

- Now only one place in dial plan to update if it is changed
- Setting variables:
  - exten => s,1,Set(a\_variable=2000)

# Asterisk Variables

---

- Complete list of Asterisk variables

# The 's' Start Extension

---

- The standard extension a call starts in without needed to specifically match an extension
- Often used with FXS/FXO cards due to lack of end to end signalling with analogue channels

```
[incoming]
exten => s,1,Answer()
exten => s,2,Background(enter-ext-of-person)

exten => 1,1,Playback(digits/1)
exten => 1,2,Goto(incoming,s,1)

exten => 2,1,Playback(digits/2)
exten => 2,2,Goto(incoming,s,1)

exten => 3,1,Hangup
```

# The Standard Extensions

---

- i : Invalid
- s : Start
- h : Hangup
- t : Timeout
- T : AbsoluteTimeout
- o : Operator

# Dial Command

---

- `Dial(tech/username:password@hostname/extension,ring-timeout,flag)`
- Can include complete information in the dial string, or reference a peer in `sip.conf`
  - `exten => 2000,1,Dial(SIP/passwd:sipdevice@host.tld)`
    - or
  - `exten => 2000,1,Dial(SIP/sipdevice)`

where there is a channel `[sipdevice]` defined in `sip.conf` containing at least definitions for `username`, `password` and `host`.

# Voicemail

---

- Comedian Mail - a fully functional voicemail system included with Asterisk
  - Supports busy and unavailable messages
  - `exten => 2001,1,Voicemail(b2001)`
  - `exten => 2001,1,Voicemail(u2001)`
- Voicemail can be emailed out a .wav attachment to users
- Standard IVR voicemail access
  - `exten => 510,1,VoicemailMain`

# voicemail.conf

---

```
[general]
format=wav49|gsm|wav
serveremail=voicemail@jonnynet.net
mailcmd=/usr/sbin/sendmail -t
attach=yes
maxmsg=100
maxmessage=180
skipms=3000
maxsilence=10
silencethreshold=128
maxlogins=3
```

```
emailbody=Dear ${VM_NAME}:\n\n\tjust wanted to let you know you were just left a
    ${VM_DUR} long message (number ${VM_MSGNUM})\nin mailbox ${VM_MAILBOX} from ${V
M_CALLERID}, on ${VM_DATE}, so you might\nwant to check it when you get a chance
.  Thanks!\n\n\t\t\t\t\t--Asterisk\n
emaildateformat=%A, %B %d, %Y at %r
```

```
[default]
; all our mailboxes here
; mailbox number => pin,name,email
2000 => 1234,Jonny,jonny@jonnynet.net
```

# Music on Hold

---

- Music on hold (MOH) played automatically when a channel is placed on hold

- Multiple classes of MOH defined

exten => 100,1,Answer()

exten => 100,2,MusicOnHold(default) ; class = default, could be any other

- Default file directory, Debian:
  - /usr/share/asterisk/mohmp3
- RedHat, or if compiling from source
  - /var/lib/asterisk/mohmp3



# musiconhold.conf

---

```
[default]
;mode=quietmp3
mode=files
directory=/var/lib/asterisk/mohmp3

; valid mode options:
; quietmp3      -- default
; mp3           -- loud
; mp3nb         -- unbuffered
; quietmp3nb    -- quiet unbuffered
; custom        -- run a custom application
; files         -- read files from a directory in any Asterisk supported format
```

# MeetMe Conferencing

---

- Powerful application built in to Asterisk
- Some use Asterisk purely for it's conferencing abilities
- Ad Hoc MeetMe conferencing, or individual conference rooms with PIN

```
/etc/asterisk/meetme.conf
; Configuration file for MeetMe simple conference rooms
;
[rooms]
; Usage is conf => confno[,pin]
;
conf => 101,1234
conf => 102,2345
```

```
/etc/asterisk/extensions.conf
exten => 5101,1,Meetme(101|M)
exten => 5102,2,Meetme(102|M)
```

# Interactive Voice Response

---

- Interactive Voice Response (IVR) is inherent to the Asterisk dialplan
- Simply a matter of playing prompts, waiting, accepting input in a channel, and moving around the dial plan
  - Useful applications:
    - Background(prompt-to-play-whilst-waiting-for-input)
    - Playback(prompt-to-play-whilst-NOT-accepting-input)
    - Goto(context,extension,priority)
    - Dial(SIP/2000)
    - Wait(seconds)

# Sample IVR

---

```
[test-ivr]
exten => s,1,Answer()
exten => s,2,Background(enter-ext-of-person)
exten => s,3,WaitExten(5)
```

```
exten => 1,1,Playback(digits/1)
exten => 1,2,Goto(incoming,s,1)
```

```
exten => 2,1,Playback(digits/2)
exten => 2,2,Goto(incoming,s,1)
```

```
exten => i,1,Playback(pbx-invalid)
exten => i,2,Goto(incoming,s,1)
```

```
exten => t,1,Playback(vm-goodbye)
exten => t,2,Hangup()
```

```
[phones]
; allow our phones to dial into the IVR
exten => 2010,1,Goto(test-ivr,s,1)
```

## Lab 2: Basic Asterisk Configuration