

Asterisk/astguiclient install from scratch. v.2.0.4 2007-12-03
By the astGUIClient group astguiclient@eflo.net

**** IMPORTANT - In order for vicidial/astguiclient to function correctly please read the REQUIREMENTS.txt for a minimum requirements list. ***

End-user Manuals for Agents and Managers are available from <http://www.eflo.net>

This document is meant to be a very in-depth step-by-step explanation of installing the Asterisk open-source PBX on a Linux system and also installing the astGUIClient suite. The instructions will assume starting from nothing and will try to give several side step instructions to account for some differences in choices of hardware and software.

The actual installation that I am doing as I write these instructions will be on the following hardware:

- Pentium 3 500MHz
- Intel motherboard D815BN
- 256MB PC133 RAM
- 80GB IBM deskstar 7200RPM Hard Drive
- Digium Wildcard Single Span T1 Card T100P
- 2U rackmount case with 250W power supply
- Phone hardware will be a Grandstream BT102 and a Sipura SPA-2000 because they are so cheap and readily available

All of these parts, aside from the Digium card and the two SIP VOIP devices, were purchased from ebay and the entire package(with the two VOIP devices and all server hardware included) cost me about \$1100 to put together including the phones and Digium adapter.

We have many other Asterisk servers at our main office, but this one can be experimented with easily because it was so cheap to make and has a relatively small capacity when compared with a multi-processor server with a quad span T1 card.

This is our test Asterisk server and functions well for a dozen or so extensions in use if it were to be used in production. A size that is optimal for many small offices operating with a fractional data/voice T1 for instance.

For hardware you can use almost any Pentium-class processor(PII, PIII, Athlon, Xeon, etc), and you can use any digium telco interface card. Both of these choices will determine what the capacity of your Asterisk server will be. If you want to do simple IVR or conference calling and a few extensions, then a PIII with a single Digium T1 card will work just fine for you. If you want to use the VICIDIAL application, you will want to get as high-powered of a machine as you can afford and get a digium quad-span T1 card.

The following is assumed for these installation procedures:

- You have access to a CD burner and 3 blank CDs
- You have some sort of broadband internet connection
- You understand basic linux commands and can use a file editor like vi
- You have all of the necessary hardware:
 - a pentium-class computer
 - a digium telco interface card with appropriate telco lines
 - at least 1 SIP VOIP device
 - a Local Area Network(LAN) with extra ports enough for the new server and the number of phones you want

PHASE 1: INSTALLING AN OPERATING SYSTEM

This installation will be using Slackware 11 for the linux distribution, Slackware 10.X will also work with these instructions. There are several easier linux distributions and there are others that are more popular, but Slackware is a nice non-commercial distro that has been around for a long time and proven itself to be a very uncluttered and stable platform for development.

1. Go to <http://www.ultimatebootcd.com/> , download the latest bootcd and burn it to a CD. This will be needed to partition the hard drive prior to installation of Slackware linux. The latest version as of this writing is 4.1 (If you have problems with your hardware booting some of the utilities with 4.1 I suggest trying 1.7, that version has older utilities, but still gets the job done and works on every machine I've tried it on).
2. Insert the ultimatebootcd you just burned into your CDROM drive and boot to it. You will select "filesystem utilities" and then "XFDISK"
3. Select any old partitions and delete them and then create 2 new partitions:
 - 70000 MB, select yes to validate, change partition type to "Linux Native"
 - 3332 MB, select yes to validate, change partition type to "Linux Swap"
 - press F3 to exit and let it do it's thing, this will take an hour or so.
4. Go to <http://www.slackware.com/getslack/> to download Slackware linux. The most recent release as of this writing is 12.0. This release fits on 3 CDs or 1 DVD. Download both installation disks from any close server listed on the download page and burn them both to CDs.
5. Insert Disk 1 of the Slackware installation CD and boot your computer. If you have a simple computer with just an IDE drive just hit enter at the boot: prompt. If you have other hard drive adapters(SCSI/RAID/SATA/etc..) you will need to look at the Slackware installation help page to determine what boot image you will need to use to install Slackware correctly.
6. Login as root and type "setup" at the prompt to go to the setup menu.
7. Go to ADDSWAP and hit Enter
8. Select the swap partition you just created and hit Yes, The swap partition will then be formatted
9. Select the root partiton you just created as Linux Native and hit Select, then select "ext3" for the file system, then select 4096 for the inodes and the root partition will then be formatted
10. Select "Install Slackware from a CD" and hit OK
11. Select "auto" installation and hit OK
12. Select every package except for "KDEI" and hit OK
13. Select "full" installation and hit OK
14. Insert the next Slackware installation disk (disk 2) when it prompts you, and hit OK to continue
15. Now you will select the boot kernel that you will use from now on. If you have a simple system with IDE drives you can probably just select "skip" and go to the next step. If not then you should probably select "cdrom" and select the kernel from the list that you selected to boot into the installation.
16. You can make a bootdisk if you like, but you don't have to.
17. For Modem you can select "no modem" and hit OK to continue
18. Enable hotplug, hit Yes to continue
19. Install lilo "simple" and hit OK to continue
20. lilo frame buffer console 640x480 is safe choice if you're not sure
21. Optional Lilo append, leave blank and hit OK to continue
22. Lilo destination, I usually choose MBR but root works most of the time

23. Mouse, select the mouse type that you have hooked up, or select ps/2
24. Load GPM at boot time, Hit Yes to continue
25. select Yes to configure your network
26. Hostname, we are typing "phone"
27. network, we are typing our local domain name
28. IP address, we are selecting Static IP, here's what we enter for network, you should enter a network setup that will work with your local LAN:
 - IP address: 10.10.10.15
 - subnet: 255.255.0.0
 - gateway: 10.10.10.1
 - name server: 10.10.10.1
29. Accept your network settings
30. Startup services to run, change nothing and select OK to continue
31. select NO for custom screen fonts
32. Hardware clock to UTC, select NO
33. Select your time zone and hit OK
34. I usually select gnome as the window manager, even though you won't be using it on this machine
35. Select Yes to enter a root password. type something that you will remember.
36. Setup of Slackware Linux is complete, hit OK and EXIT then press CTRL-ALT-DELETE to reboot your computer

PHASE 2: COMPILING A CUSTOM LINUX KERNEL

From this step on you should be able to continue the installation remotely although it is wise to at least have quick access to the machine if something goes wrong.

To connect remotely through SSH on linux type "slogin serveripaddress" or to use Windows to connect you can use a piece of free software called putty available here: <http://www.chiark.greenend.org.uk/~sgtatham/putty/download.html>
Also, for windows you can use SSH file transferring(SFTP) with a program called filezilla: <http://filezilla.sourceforge.net/>

This is an optional step if your linux system is running, but compiling your own custom kernel is always a way to optimize your system for the hardware you have installed or a way to remove the unnecessary modules that are in the default kernel. You will definately want to build your own kernel if you have a multi processor machine. If you are new to Linux you probably do not want to do this.

If you are using a newer Digium Octasic-based echo-cancellation quad T1/E1 card then you need to use a 2.6 Linux kernel in order to use the echo-cancellation functionality of the card.

If you want to compile a 2.6 kernel then start with OPTION 1, otherwise to compile the 2.4 kernel that comes with Slackware(2.4.33) start with OPTION 2:

OPTION 1: compile Linux kernel 2.6.17 *RECOMMENDED*

1. cd /usr/src
2. wget <http://www.kernel.org/pub/linux/kernel/v2.6/linux-2.6.17.11.tar.gz>
3. gunzip linux-2.6.17.11.tar.gz
4. tar xvf linux-2.6.17.11.tar
5. mv -f /usr/src/linux /usr/src/linux-old
6. ln -s /usr/src/linux-2.6.17.11 /usr/src/linux
7. cd linux
8. make mrproper # prep for kernel assembly
9. make menuconfig # launch configuration menu app
(this part is very dependant upon your own hardware)
(what is mentioned below are only changes beyond what is selected by default)

Processor Type and Features --->

- >Symmetric multi-processing support
(if you have multiple processors or a Dual-core or HT enabled)
- >High Memory Support
(if you have more than 900MB of System RAM move upto 4GB)
- >Timer frequency (1000 HZ)
(change to 1000Hz if using ztdummy for timer)
- >Preemption Model (No Forced Preemption (Server))
(This one is very important!!!)

Power management options (ACPI, APM) --->

- >ACPI (Advanced Configuration and Power Interface) Support
(enable all down to Processor and thermal zone)

Bus options (PCI, PCMCIA, EISA, MCA, ISA) --->

- >[*] PCI Express support
(if using Sangoma PCI Express card)

Networking --->

Amateur Radio support --->

- <*> Amateur Radio AX.25 Level 2 protocol
- [*] AX.25 DAMA Slave support
- <*> Amateur Radio NET/ROM protocol

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    <*> Amateur Radio X.25 PLP (Rose)
    (all needed for new Digium Octasic drivers)
Device Drivers --->
ATA/ATAPI/MFM/RLL support --->
    <*> SCSI emulation support
    (needed for SATA drives, also further down check chipset drivers)
Block devices --->
    <*> Loopback device support
    <*> RAM disk support
    (1) Default number of RAM disks
    (128000) Default RAM disk size (kbytes)
    [*] Initial RAM filesystem and RAM disk (initramfs/initrd) support
    (ONLY USE THE ABOVE 4 SETTINGS IF YOU WILL BE SETTING UP A RAM DRIVE FOR RECORDING)
SCSI device support --->
    <*> RAID Transport Class
    (needed if you are using a RAID)
SCSI low-level drivers --->
    <*> Serial ATA (SATA) support
    (required if using SATA drives)
    (if using a SCSI RAID card pick correct driver here)
Multi-device support (RAID and LVM) --->
    (select proper RAID types if using Linux RAID)
Network device support --->
    Ethernet (10 or 100Mbit) --->
    Ethernet (1000 Mbit) --->
    (select proper drivers for the type of network card you have)
Character devices --->
    <*> Enhanced Real Time Clock Support
    (double-check that this is enabled, very important)
Real Time Clock --->
    <*> RTC class
    (double-check that this is enabled, very important)
File systems --->
    <*> Ext3 journalling file system support
    (important if using ext3 filesystem)
Library routines --->
    <*> CRC-CCITT functions
    <*> CRC16 functions
    <*> CRC32c (Castagnoli, et al) Cyclic Redundancy-Check
    (important for new Digium Octasic drivers)
EXIT AND SAVE YOUR CONFIGURATION

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10. make clean                                # clean up the kernel build areas
11. make bzImage                             # create a kernel bzImage
12. make modules                             # build the modules into the image
13. make modules_install                     # install kernel modules
14. cp arch/i386/boot/bzImage /boot/bzImage-XXXX # copy image
    (put whatever you want in XXXX, that is your new kernel name)
15. cp System.map /boot/System.map-XXXX     # copy system map
16. mv -f /boot/System.map /boot/System.map-orig
17. ln -s /boot/System.map-XXXX /boot/System.map # symlink map
18. vi /etc/lilo.conf                        # edit the lilo boot config file
    image=/boot/bzImage-XXXX # add the new image in above-
    label=test-XXXX          # the previous one
    root=/dev/hda1           # device of root partition
    read-only
19. /sbin/lilo                             # run the lilo reload script
20. shutdown -r 0                         # reboot machine and hope it worked

```

OPTION 2: compile Linux kernel 2.4.33.3

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1. cd /usr/src/linux                # move to your linux source directory
2. cp .config config.save           # copy old config to a save file
3. make mrproper                    # prep for kernel assembly
4. make menuconfig                  # launch configuration menu app
   (this part is very dependant upon your own hardware)
   enable processor version          # select the processor that you have
   enable SMP                        # if more than 1 processor or Intel HT
   enable high memory ( )            # if more than 1GB of RAM
   enable SCSI Multiple              # if SCSI drives
   enable SCSI devices AMI Megaraid  # if SCSI Megaraid adapter
   enable 3com network devices      # if 3com network card
   enable ext3 file system           # for ext3 to work
   enable all ACPI options           # for SMP to work
   enable Enhanced Real Time Clock Support in Character devices section
                                     # for SMP to work
   enable any other hardware specific options
   exit and save configuration
5. make dep                         # build the kernel dependancies
6. make clean                       # clean up the kernel build areas
7. make bzImage                     # create a kernel bzImage
8. make modules                     # build the modules into the image
9. make modules_install             # install kernel modules
10. # nothing# mkinitrd /boot/initrd-XXXXXX.img XXXXXX *not needed on Slackware*
11. cp arch/i386/boot/bzImage /boot/bzImage-XXXXXX # copy image
    (put whatever you want in XXXXXX, that is your new kernel name)
12. cp System.map /boot/System.map-XXXXXXN # copy system map
13. mv -f /boot/System.map /boot/System.map-orig
14. ln -s /boot/System.map-XXXXXX /boot/System.map # symlink map
15. vi /etc/lilo.conf               # edit the lilo boot config file
    image=/boot/bzImage-XXXXXX      # add the new image in above-
    label=test-XXXXXX               # the previous one
    root=/dev/hda1                  # device of root partition
    read-only
16. /sbin/lilo                      # run the lilo reload script
17. shutdown -r 0                   # reboot machine and hope it worked

```

After compiling your kernel you can run a few commands to verify that you are running your new kernel and that devices are running as they are supposed to:

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ps --info (will show you your linux kernel version and other info)
cat /proc/cpuinfo (will show you processor type and more than one if SMP)
top (will show you system memory)

```

PHASE 3: INSTALLING SOFTWARE BEFORE ASTERISK

In this step we will be installing software that Asterisk and/or astGUIclient needs to be able to function to its fullest ability. Not all of these software packages are mandatory to successfully install Asterisk and some of them can be installed on other machines on your network like MySQL or Apache. But, in this installation we are assuming that there are no other machines on our network to help the Asterisk server, so it must have everything it needs installed locally.

SUBPHASE 3.0: install new Gnu Make

A new version of the "make" compilation application replaced the 4-year-old version that most Linux distros(Including Slackware) use. This is only needed if you will be building Asterisk from the 1.4 release tree.

- wget <http://mirrors.kernel.org/gnu/make/make-3.81.tar.gz>
- gunzip make-3.81.tar.gz
- tar xvf make-3.81.tar
- cd make-3.81
- ./configure
- make
- make install

SUBPHASE 3.1: MySQL requirements

You must at least have Mysql client installed on each VICIDIAL server, but you only need one database server.

MySQL is a fast database system that is very easy to integrate with any application. You can either install the server on the local Asterisk system or have one somewhere on your network. For our purposes, we are creating an Asterisk installation that is self contained and needs no other local servers to operate, so we will need to install mysql on this machine.

REQUIRED and OPTIONAL (only install MySQL server locally if you don't want to use an installation on another machine, Mysql client is required on all VICIDIAL servers)

NOTE: a minimum of MySQL server 4.0.X is required

Go to <http://www.mysql.com/> and download the mysql package

- to install this directly on the command line type:
 - cd /usr/local
 - wget http://mirror.trouble-free.net/mysql_mirror/Downloads/MySQL-4.0/mysql-4.0.27.tar.gz
http://mirror.trouble-free.net/mysql_mirror/Downloads/MySQL-5.0/mysql-5.0.41.tar.gz
 - gunzip mysql-4.0.27.tar.gz
 - tar xvf mysql-4.0.27.tar
 - cd mysql-4.0.27
 - groupadd mysql
 - useradd -g mysql mysql
 - ". /configure --prefix=/usr/local/mysql --enable-large-files --enable-shared=yes --with-readline"
 - **** If only MySQL client is needed for DBD::mysql then use this:
 - ". /configure --prefix=/usr/local/mysql --without-server --enable-shared=yes --with-readline"
 - **** If using version 5 tree MySQL client then use this:
 - ". /configure --prefix=/usr/local/mysql --without-server --enable-shared=yes --with-readline --enable-thread-safe-client"
 - make
 - make install
 - scripts/mysql_install_db

- chown -R root /usr/local/mysql
- chown -R mysql /usr/local/mysql/var
- chgrp -R mysql /usr/local/mysql
- cp support-files/my-medium.cnf /etc/my.cnf
- /usr/local/mysql/bin/mysqld_safe --user=mysql &
- ln -s /tmp/mysql.sock /var/run/mysql/mysql.sock
- **** For some systems you may need to add the mysql/bin directory to your PATH:
 - PATH=\$PATH:\$HOME/bin:/usr/local/mysql/bin/
 - export PATH
- **** you may also want to add those two lines to your /root/.bash_profile file
- **** For Mysql 5 tree only, you also may need to copy the libmysqlclient.so file to libs
 - cp /usr/local/mysql-5.0.27/libmysql/.libs/libmysqlclient.so /usr/lib/
 - cp /usr/local/mysql-5.0.27/libmysql/.libs/libmysqlclient.so.15 /usr/lib/

- you are done

** INSTALLATION NOTE **

If you are having Linuxthreads problems upon onfigure, just execute the following command:
 echo '/* Linuxthreads */' >> /usr/include/pthread.h

***** NOTE: if you will be using any of the DBI perl scripts: *****

Every machine that you will be using the newer BDI perl scripts on will need to have the perl modules DBI and DBD::mysql installed on them. To do this you will also need to at least have the MySQL client installed on the server (see above) then you will need to go to 'cpan' and "install DBI" and "install DBD::mysql". You may need to "force install DBD::mysql" if the DBD tests fail on your first try, but that is OK since the tests are not needed

SUBPHASE 3.2: Installing Perl Modules

NOTE - you can install ActiveState <http://www.activestate.com> perl which may improve performance, but it is not required. Here's the source for ActiveState Perl 5.8: (it's free)

http://downloads.activestate.com/ActivePerl/src/5.8/AP817_source.tgz

I hope to add the lengthy steps for installing it as your default perl on your server but I need some time and a free machine to do that.

cpan is the "Comprehensive Perl Archive Network". It's a mirrored archive of most of the perl modules out there complete with a installation and management command-line interface. Here's what you do to start it:

REQUIRED (needed for perl AGIs)

1. perl -MCPAN -e shell # type in the command line
2. You will then go through CPAN setup, just hit ENTER for most prompts except for the mirrors list, you will want to select at least 4 mirrors
 - yes for manual configuration
 - enter for the next 18 prompts
 - for the "make install options" it's a good idea to add UNINST=1
 - enter for the next 4 prompts
 - select your continent and country
 - select a few cpan mirrors
 - enter for the next 2 prompts
3. Once you see the cpan> prompt you can begin installing modules
4. If you've never installed cpan before you should probably install the following modules first: (say YES if asked to install prerequisites)
 - install MD5
 - install Digest::MD5
 - install Digest::SHA1
 - install readline (just hit Enter when it asks for operator)
 - install Bundle::CPAN
 - reload cpan

- then you can install other modules:
 - install DBI
 - force install DBD::mysql (must at least have mysqlclientlibs installed)
 - install Net::Telnet
 - install Time::HiRes
 - install Net::Server
 - install Unicode::Map (needed for super list loader Excel)
 - install Jcode (needed for super list loader Excel)
 - install Spreadsheet::WriteExcel (needed for super list loader Excel)
 - install OLE::Storage_Lite (needed for super list loader Excel)
 - install Proc::ProcessTable (needed for super list loader Excel)
 - install IO::Scalar (needed for super list loader Excel)
 - install Spreadsheet::ParseExcel (needed for super list loader Excel)
 - then quit cpan, you are done
5. Go to <http://asterisk.gnuinter.net/> and download the asterisk-perl module
 (backup link: <http://www.freedomphones.net/files/asterisk-perl-0.08.tar.gz>)
 NOTE: Do NOT use the 0.09 version, it does not work with VICIDIAL
- to install this directly on the command line type:
 - cd /usr/local
 - wget <http://asterisk.gnuinter.net/files/asterisk-perl-0.08.tar.gz>
 - gunzip asterisk-perl-0.08.tar.gz
 - tar xvf asterisk-perl-0.08.tar
 - cd asterisk-perl-0.08
 - perl Makefile.PL
 - make all
 - make install
 - you are done

SUBPHASE 3.3: Installing other utilities

Sox is an audio utility that allows you to mix audio files together at their start point into one file. it is necessary for Asterisk recordings that record in and out as separate files

REQUIRED (needed for recording mixing)

1. Go to <http://sourceforge.net/projects/sox/> and download the sox package
 - to install this directly on the command line type:
 - cd /usr/local
 - wget <http://easynews.dl.sourceforge.net/sourceforge/sox/sox-12.17.9.tar.gz>
 - gunzip sox-12.17.9.tar.gz
 - tar xvf sox-12.17.9.tar
 - cd sox-12.17.9
 - ./configure
 - make (if alsa.o errors add --disable-alsa-dsp to configure and redo)
 - make install
 - you are done

LAME is an MP3 encoder used to convert audio files from WAV to MP3. We prefer GSM usually, but some users have standardized on MP3 so they would need this utility to be loaded to use that option.

OPTIONAL (only needed if you will be converting recordings to MP3)

2. Go to <http://lame.sourceforge.net/> and download the lame package
 - to install this directly on the command line type:
 - cd /usr/local
 - wget <http://easynews.dl.sourceforge.net/sourceforge/lame/lame-3.96.1.tar.gz>
 - gunzip lame-3.96.1.tar.gz
 - tar xvf lame-3.96.1.tar
 - cd lame-3.96.1

- ./configure
- make
- make install
- you are done

Screen is a terminal emulator that allows you to run a process as command line and be able to detach from them('Ctrl+a' then 'd') and log all output of the terminal to a screenlog file if desired(add a '-L' to the launching command). In our installations this is how we launch Asterisk upon startup and still have the ability to log output and still attach to the screen that executed asterisk originally.

REQUIRED* *MANDATORY FOR VICIDIAL SERVERS

3. Go to <http://www.gnu.org/software/screen/> and download the screen package

- to install this directly on the command line type:
 - cd /usr/local
 - wget <http://ftp.gnu.org/gnu/screen/screen-4.0.2.tar.gz>
 - or for older version:
 - wget <http://mirrors.kernel.org/gnu/screen/screen-3.9.15.tar.gz>
 - gunzip screen-4.0.2.tar.gz
 - tar xvf screen-4.0.2.tar
 - cd screen-4.0.2
 - ./configure
 - make
 - make install
- you are done

ttload is a simple terminal application that shows the processor load in a graphical time-based scrolling graph. We use it to view how loaded the system is and it visualizes load spikes very well

OPTIONAL (only for obsessive admins like me)

4. Go to <http://www.davetld.com/src/util/ttyload/> and download the ttyload package

- to install this directly on the command line type:
 - cd /usr/local
 - wget <http://www.davetld.com/src/util/ttyload/ttyload-0.4.4.tar.gz>
 - gunzip ttyload-0.4.4.tar.gz
 - tar xvf ttyload-0.4.4.tar
 - cd ttyload-0.4.4
 - make
 - ln -s /usr/local/ttyload-0.4.4/ttyload /usr/bin/ttyload
- you are done

ntpd is the network time protocol daemon that matches the time on your machine with the time of a master server somewhere in the world. We use it to make sure the time is the same on our client computers and our servers.

MANDATORY FOR VICIDIAL SERVERS (install on server and all clients)

5. Go to <http://www.ntp.org/> and download the ntpd package

- to install this directly on the command line type:
 - cd /usr/local
 - wget http://www.eecis.udel.edu/~ntp/ntp_spool/ntp4/ntp-4.2/ntp-4.2.2p3.tar.gz
 - If you get compilation errors here try 4.1.2:
 - wget http://www.eecis.udel.edu/~ntp/ntp_spool/ntp4/ntp-4.1.2.tar.gz
 - gunzip ntp-4.2.2p3.tar.gz
 - tar xvf ntp-4.2.2p3.tar
 - cd ntp-4.2.2p3

- ./configure
- make
- make install
- vi /etc/ntp.conf (change to just 1 line: "server 18.145.0.30")
- cp /etc/ntp.conf /etc/ntpd.conf# just to be sure
- /usr/local/bin/ntpdate -u 18.145.0.30 # initial sync
- /usr/sbin/ntpd # run it
- you are done

iftop is a good console bandwidth visualization tool that shows you active connections, where they are going to/from and how much of your precious bandwidth they are using. *OPTIONAL*

6. Go to <http://www.ex-parrot.com/~pdw/iftop/> and download the package

- to install this directly on the command line type:
 - cd /usr/local
 - wget http://www.tcpdump.org/release/libpcap-0.9.4.tar.gz
 - gunzip libpcap-0.9.4.tar.gz
 - tar xvf libpcap-0.9.4.tar
 - cd libpcap-0.9.4
 - ./configure
 - make
 - make install
 - cd /usr/local
 - wget http://www.ex-parrot.com/~pdw/iftop/download/iftop-0.17.tar.gz
 - gunzip iftop-0.17.tar.gz
 - tar xvf iftop-0.17.tar
 - cd iftop-0.17
 - ./configure
 - make
 - make install
 - iftop
- you are done

ploticus is a free graph creation package that allows you to create line graphs within PNG files simply by creating a config file and a data file. We use this package along with the included PHP script to generate server performance graphs that can be displayed real-time on a web page.

OPTIONAL (only needed for server performance graphing web reports)

7. Go to <http://ploticus.sourceforge.net/> and download the package

- to install this directly on the command line type:
 - NOTE: you may have to edit the Makefile to remove X11 if you don't have it
 - cd /usr/local
 - wget http://ploticus.sourceforge.net/download/pl232src.tar.gz
 - gunzip pl232src.tar.gz
 - tar xvf pl232src.tar
 - cd pl232src/src/
 - make clean
 - make
 - make install
- you are done

NOTE: for the graphics to work on the AST_server_performance page you will need the 'pl' script to be linked or copied into your httdocs/vicidial/ploticus directory

NOTE: you may need to edit the Makefile for ploticus if you do not have X11

openssh is a remote login protocol server that is always a good idea to have updated on your system, so we're going to install a new version now.

OPTIONAL (only updated as a precaution, not mandatory)

[NOTE: newer zlib is needed before installing]

8. Go to <http://www.openssh.org/> and download the linux source for openssh

- to install this directly on the command line type:
 - cd /usr/local
 - wget <http://www.zlib.net/zlib-1.2.3.tar.gz>
 - gunzip zlib-1.2.3.tar.gz
 - tar xvf zlib-1.2.3.tar
 - cd zlib-1.2.3
 - ./configure
 - make
 - make install
 - cd /usr/local
- wget <http://gulus.usherbrooke.ca/pub/appl/ssh/OpenSSH/portable/openssh-4.7p1.tar.gz>
 - gunzip openssh-4.7p1.tar.gz
 - tar xvf openssh-4.7p1.tar
 - cd openssh-4.7p1
 - ./configure
 - make
 - make install
- you are done

openssl is the open-source SSL libraries package, and to install a fake SSL cert locally and have it work with apache, you need it installed on your machine

OPTIONAL (only install openssl if you want to use SSL secured web pages on your locally installed copy of Apache web server)

9. Go to <http://www.openssl.org/> and download the linux source for openssl

- to install this directly on the command line type:
 - cd /usr/local
 - wget <http://www.openssl.org/source/openssl-0.9.8g.tar.gz>
 - gunzip openssl-0.9.8g.tar.gz
 - tar xvf openssl-0.9.8g.tar
 - cd openssl-0.9.8g
 - ./config
 - make
 - make install
- you are done

apache is a web server that allows you to use many different modules with it to extend it's functionality. In order to use some of the astguiclient functionalities we need to have Apache and PHP installed on this machine.

OPTIONAL (only install Apache and PHP locally if you don't want to use an installation on another machine)

10. Go to <http://www.apache.org/> and download the apache unix source

Go to <http://www.php.net/> and download the php unix source code

- to install this directly on the command line type:
 - cd /usr/local
 - wget <http://apache.secsup.org/dist/httpd/httpd-2.0.61.tar.gz>
 - gunzip httpd-2.0.61.tar.gz
 - tar xvf httpd-2.0.61.tar
 - wget <http://us2.php.net/distributions/php-4.4.7.tar.gz>
 - NOTE: PHP 5.1.1 has also been tested with this release
 - gunzip php-4.4.7.tar.gz
 - tar xvf php-4.4.7.tar
 - cd httpd-2.0.61
 - ./configure --enable-so --with-apxs2
 - make
 - make install

- cd ../php-4.4.7
- ./configure --with-apxs2=/usr/local/apache2/bin/apxs --with-mysql
- make
- make install
- cp php.ini-dist /usr/local/lib/php.ini
- NOTE: you will want to make sure notice logging is turned off:
error_reporting = E_ALL & ~E_NOTICE ; (this is default)
- !!! REQUIRED !!! raise the memory limit for scripts in php.ini:
memory_limit = 48M
- vi /usr/local/apache2/conf/httpd.conf
- add the following lines:
"AddType application/x-httpd-php .php .phtml"
"LoadModule php4_module libexec/libphp4.so"
or
"LoadModule php4_module modules/libphp4.so"
- modify the index.html line and add index.php to the list
- /usr/local/apache2/bin/apachectl start
- go to http://your-new-asterisk-server-ipaddress/ to see if it worked
- you are done

OPTIONAL- Load eaccelerator PHP-caching application:

- Go to <http://eaccelerator.net> and download the most recent package
- cd /usr/local
- wget <http://bart.eaccelerator.net/source/0.9.5/eaccelerator-0.9.5.zip>
- unzip eaccelerator-0.9.5.zip
- cd eaccelerator-0.9.5
- export PHP_PREFIX="/usr/local"
- \$PHP_PREFIX/bin/phpize
- ./configure --enable-eaccelerator=shared --with-php-config=\$PHP_PREFIX/bin/php-config
- make
- make install
- vi /usr/local/lib/php.ini
- Add the following lines to the dynamic extensions section of php.ini:
(you may need to change the extension location depending on your install of php)
extension="../../usr/local/eaccelerator-0.9.5/modules/eaccelerator.so"
eaccelerator.shm_size="48"
eaccelerator.cache_dir="/tmp/eaccelerator"
eaccelerator.enable="1"
eaccelerator.optimizer="1"
eaccelerator.check_mtime="1"
eaccelerator.debug="0"
eaccelerator.filter=""
eaccelerator.shm_max="0"
eaccelerator.shm_ttl="0"
eaccelerator.shm_prune_period="0"
eaccelerator.shm_only="0"
eaccelerator.compress="1"
eaccelerator.compress_level="9"
- mkdir /tmp/eaccelerator
- chmod 0777 /tmp/eaccelerator
- # to verify installation:
- php -v

balance is a load-balancing application for Linux that will allow you to spread the load of your web traffic across many servers. If you are running more than 70 agents on a single server you may want to install this application and build another cheap web server to handle the extra load.

OPTIONAL

11. Go to <http://balance.sourceforge.net> to download the most recent source version
- to install this directly on the command line type:
 - cd /usr/local
 - wget <http://voxel.dl.sourceforge.net/sourceforge/balance/balance-3.34.tgz>
 - gunzip balance-3.34.tgz
 - tar xvf balance-3.34.tar
 - cd balance-3.34
 - make
 - make install
 - /usr/sbin/balance -f 81 localhost:80 10.10.10.16:80
- That command will take port 81 traffic and send it evenly to the local server and the 10.10.10.16 server reducing the load and speeding up the applications. More info on balance: <http://www.inlab.de/balance.pdf>

subversion is the new code control framework use by the Asterisk community. If you want to use the latest development code of Asterisk you will need to have this loaded on your system.

OPTIONAL

12. Go to <http://subversion.tigris.org> to download the most recent source version
- to install this directly on the command line type:
 - cd /usr/local
 - wget <http://subversion.tigris.org/downloads/subversion-1.3.2.tar.gz>
 - gunzip subversion-1.3.2.tar.gz
 - tar xvf subversion-1.3.2.tar
 - cd subversion-1.3.2
 - ./configure
 - make
 - make install

mtop is a great utility for real-time monitoring of mysql and the queries that are running in it.

OPTIONAL

13. Go to <http://mtop.sourceforge.net> to download the most recent version
- to install this directly on the command line type:
 - cd /usr/local
 - wget <http://superb-east.dl.sourceforge.net/sourceforge/mtop/mtop-0.6.6.tar.gz>
 - gunzip mtop-0.6.6.tar.gz
 - tar xvf mtop-0.6.6.tar
 - cd mtop-0.6.6
 - cpan
 - install Curses
 - install Getopt::Long
 - install Net::Domain
 - quit
 - perl Makefile.PL
 - make
 - make install
 - /usr/local/bin/mtop --dbuser=root --seconds=3

sipsak is an optional utility that VICIDIAL can use to send messages to an agent's SIP-based phone(like the Snom 320) to display text on their LCD screen. If you want to use this, make sure it is installed on the same server that your web server is installed on(Apache).

OPTIONAL

14. Go to <http://sipsak.org> to download the most recent version
- to install this directly on the command line, type:
 - cd /usr/local
 - wget <http://download.berlios.de/sipsak/sipsak-0.9.6-1.tar.gz>

- gunzip sipsak-0.9.6-1.tar.gz
- tar xvf sipsak-0.9.6-1.tar
- cd sipsak-0.9.6-1
- ./configure
- make
- make install
- /usr/local/bin/sipsak --version

PHASE 4: INSTALLING ASTERISK

OK, all the prep work is done, now it's time to start having fun with Asterisk. There are two basic ways to install Asterisk, an official release(at the time of this writing the official release is 1.2.24) and the SVN_DEV version(development branch). We recommend using Asterisk 1.2.24. These instructions are how we get our Asterisk system with it's T1 line installed with our 2 SIP VOIP devices and one IAX2 softphone.

NOTE: If you want to use Asterisk 1.4, you will need to make sure that you set the servers table "asterisk_version" field to the proper version number and you can use the docs/conf_examples/extensions.conf.sample-1.4 file for your default dialplan

NOTE: If you want to use release 1.0.8 or 9 we would recommend either using the CVS_v1-0 branch where the issues are fixed, or patching your 1.0.8/1.0.9 code with the following patch:

(http://astguiclient.sourceforge.net/experimental_code/localmasq.patch)

- If you do patch your system make sure you put the asterisk version field for the server on the admin pages as '1.0.11.1'

1. follow these command line steps:

- mkdir /usr/src/asterisk
- cd /usr/src/asterisk

A. if you want 1.2 release (reliable with new features):

- wget <http://ftp.digium.com/pub/asterisk/releases/asterisk-1.2.24.tar.gz>
- wget <http://ftp.digium.com/pub/zaptel/releases/zaptel-1.2.21.tar.gz>
- wget <http://ftp.digium.com/pub/libpri/releases/libpri-1.2.5.tar.gz>
- gunzip asterisk-1.2.24.tar.gz
- tar xvf asterisk-1.2.24.tar
- gunzip zaptel-1.2.21.tar.gz
- tar xvf zaptel-1.2.21.tar
- gunzip libpri-1.2.5.tar.gz
- tar xvf libpri-1.2.5.tar

B. if you want latest SVN_1.2 version (release tree with new patches)

- svn checkout <http://svn.digium.com/svn/asterisk/branches/1.2> asterisk-1.2
- svn checkout <http://svn.digium.com/svn/zaptel/branches/1.2> zaptel-1.2
- svn checkout <http://svn.digium.com/svn/libpri/branches/1.2> libpri-1.2

C. if you want latest SVN_DEV version (not recommended) [1.6 tree]

- svn checkout <http://svn.digium.com/svn/asterisk/trunk> asterisk
- svn checkout <http://svn.digium.com/svn/zaptel/trunk> zaptel
- svn checkout <http://svn.digium.com/svn/libpri/trunk> libpri

- ALL ->

- (1.0 tree)if you want to allow for more than 100 voicemail messages in a mailbox(warning this will slightly increase memory usage when a call is in voicemail) edit the voicemail source code file:

- vi /usr/src/asterisk/asterisk/apps/app_voicemail.c
edit this line and change 100 to 999:
#define MAXMSG 100

- (1.0 tree)if you have no X server installed on your Asterisk machine, then you will need to comment out the gtk console lib(only affects 1.0 releases)
edit the voicemail source code file:

- vi /usr/src/asterisk/asterisk/pbx/Makefile
edit this line at the top and just add a hash # in front of it as shown:

#PBX_LIBS+=\$(shell \$(CROSS_COMPILE_BIN)gtk-config --cflags >/dev/null 2>/dev/null && echo

"pbx_gtkconsole.so")

- cd ./zaptel-1.2.21
- make clean
- make

- make install
- cd ../libpri-1.2.5
- make clean
- make
- make install
- cd ../asterisk-1.2.24
 - (1.2 tree) If you want to include Answering Machine Detection ability you will need to download app_amd.c and amd.conf and alter the apps/Makefile to compile it properly
 - cd apps
 - wget http://www.eflo.net/files/app_amd2.c
 - mv app_amd2.c app_amd.c
 - vi Makefile
 - replace this line(line 32):
 - app_mixmonitor.so app_stack.so
 - with this line:
 - app_mixmonitor.so app_stack.so app_amd.so
 - wget http://www.eflo.net/files/amd2.conf
 - mkdir /etc/asterisk
 - mv amd2.conf /etc/asterisk/amd.conf
 - *OPTIONAL*(1.2.23 thru 1.2.24) apply the meetme DTMF passthru patch
 - wget http://www.eflo.net/files/meetme_DTMF_passthru-1.2.23.patch
 - patch -p1 < ./meetme_DTMF_passthru-1.2.23.patch
 - File to patch: app_meetme.c
 - *OPTIONAL*(1.2.12.1 thru 1.2.24) apply the meetme volume control patch
 - *Different patches available for 1.2.7.1 through 1.2.14
 - wget http://www.eflo.net/files/meetme_volume_control_1.2.16.patch
 - patch -p1 < ./meetme_volume_control_1.2.16.patch
 - File to patch: app_meetme.c
- cd ../
 - (1.2 tree) apply the cli delimiter patch
 - wget http://www.eflo.net/files/cli_chan_concise_delimiter.patch
 - patch -p1 < ./cli_chan_concise_delimiter.patch
 - File to patch: cli.c
 - *OPTIONAL*(1.2.14 thru 1.2.24) rewrite of waitforsilence
 - wget http://www.eflo.net/files/app_waitforsilence.c
 - mv app_waitforsilence.c apps/app_waitforsilence.c
 - *OPTIONAL* shorter enter and leave sounds for meetme
 - wget http://www.eflo.net/files/enter.h
 - wget http://www.eflo.net/files/leave.h
 - mv -f enter.h apps/enter.h
 - mv -f leave.h apps/leave.h

- make clean
- make
- make install
- make samples # this makes sample conf files (only use for new installs)

- modprobe zaptel # this loads the zaptel module
- install the module for the digium device that you are using, we are using the T100P single span T1 card so we use:
- modprobe wct1xxp

Here's the list of all digium cards and the modules you use with

them:

Card	Module
TDM400P	wctdm
X100P	wcfxo

```

TDM*      wcfxs
S100U     wcusb
T100P     wct1xxp
E100P     wct1xxp
T400P     tor2
E400P     tor2
TE110P    wcte11xp
TE410P    wct4xxp
TE405P    wct4xxp
TE411P    wct4xxp
TE406P    wct4xxp
TE210P    wct2xxp
TE205P    wct2xxp
TDM2400P  wctdm24xxp

```

- If you have chosen a Sangoma T1/E1 or analog card, you will need to follow their instructions for installation of their driver software

LATEST Sangoma Wanpipe drivers:

ftp://ftp.sangoma.com/linux/current_wanpipe/wanpipe-3.2.1.tgz

- now your asterisk installation is built and loaded and it's time to configure it.

NOTES: If you want to install zttool diagnostics you may need the newt package installed:

- wget <http://www.freedomphones.net/files/newt-0.51.6.tar.gz>
- gunzip newt-0.51.6.tar.gz
- tar xvf newt-0.51.6.tar
- cd newt-0.51.6
- ./configure
- make
- make install
- cd ../
- ln -s /usr/lib/libnewt.so.0.51.6 /usr/lib/libnewt.so.0.51

then go to your zaptel folder and do 'make zttool'

Digium/Clone X100P EXAMPLE:

Here is an example of a configuration where an X100P single FXO card is used for zaptel timing and not used for calling:

NOTE: you can get an X100P through ebay for \$10-\$30 that will work with Asterisk

```

/etc/zaptel.conf:
loadzone=us
defaultzone=us
fxsks=1

```

```

/etc/asterisk/zapata.conf:
[trunkgroups]
[channels]
context=unused
signalling=fxs_ks
channel => 1

```

```

Added this to the rc.local file:
# Load zaptel drivers for x100p
modprobe zaptel
modprobe wcfxo

```

If you will be doing native music-on-hold for your inbound calls, you will need musiconhold audio files to be converted to native formats like GSM, ULAW and ALAW:

```

cd /var/lib/asterisk/mohmp3/
mpg123 -s --rate 44100 --mono /var/lib/asterisk/mohmp3/fpm-sunshine.mp3 > /var/lib/asterisk/mohmp3/fpm-sunshine.raw

```

```
sox -r 44100 -w -s -c 1 fpm-sunshine.raw -r 8000 -c 1 fpm-sunshine.wav
sox fpm-sunshine.wav -t gsm -r 8000 -b -c 1 fpm-sunshine.gsm
sox fpm-sunshine.wav -t ul -r 8000 -b -c 1 fpm-sunshine.pcm
mpg123 -s --rate 44100 --mono /var/lib/asterisk/mohmp3/fpm-calm-river.mp3 > /var/lib/asterisk/mohmp3/fpm-
calm-river.raw
sox -r 44100 -w -s -c 1 fpm-calm-river.raw -r 8000 -c 1 fpm-calm-river.wav
sox fpm-calm-river.wav -t gsm -r 8000 -b -c 1 fpm-calm-river.gsm
sox fpm-calm-river.wav -t ul -r 8000 -b -c 1 fpm-calm-river.pcm
mpg123 -s --rate 44100 --mono /var/lib/asterisk/mohmp3/fpm-world-mix.mp3 > /var/lib/asterisk/mohmp3/fpm-
world-mix.raw
sox -r 44100 -w -s -c 1 fpm-world-mix.raw -r 8000 -c 1 fpm-world-mix.wav
sox fpm-world-mix.wav -t gsm -r 8000 -b -c 1 fpm-world-mix.gsm
sox fpm-world-mix.wav -t ul -r 8000 -b -c 1 fpm-world-mix.pcm
rm -f *.raw
mkdir ../orig-mp3
mv -f *.mp3 ../orig-mp3/
```

PHASE 5: CONFIGURING ASTERISK AND YOUR SIP PHONES

In this phase we will configure the telco lines, the SIP phones, the extensions, meetme(conference calling) rooms, dialplan extensions and the voicemail boxes. After this phase your Asterisk system should be able to place and receive calls to and from the SIP phones you have installed over the telco lines you've hooked up. There are several things that we will not be showing how to do because Asterisk is extremely flexible and has so many different ways of being configured, that if we were to try to explain them all in this document it would be 99% asterisk configuration and be 20,000 lines long, and that would just be a barrier for those who just want to get it set up. The "Wiki" and the mailing list are two very good resources for finding answers if you run into problems configuring your system, here are links to them:

The Wiki: <http://www.voip-info.org/tiki-index.php>

The Lists: <http://www.asterisk.org/index.php?menu=support>

I need to note that it is possible to install Asterisk and use astGUIClient applications with no Zaptel(Digium) cards installed, but it is not recommended even if you are not going to use Zap trunks for your inbound-outbound calls with no real Zap devices, you would need to use a dummy timer(zt_dummy) based on you USB ports to get meetme conference rooms working properly and you may have other issues along the way. We would at least recommend getting a X100 or X101 board from Digium or a clone manufacturer so there is a dedicated hardware timer in place on your system.

SUBPHASE 5.0: setting up your Asterisk configuration files

1. edit zaptel.conf

- vi /etc/zaptel.conf

There are many examples inside of the zaptel.conf file that is generated with the "make samples" command that we issued at the end of the last phase. There are many different parameters for the different telco line possibilities, because we are installing a T1 that is NON-PRI-isdn B8ZS ExtendedSuperframe(ESF) E&M Wink start and 24 channels, we will use the following settings for zaptel.conf:

span=1,1,0,esf,b8zs

e&m=1-24

loadzone = us

defaultzone=us

FOR A PRI YOU WOULD USE SOMETHING LIKE THIS:

span=2,2,0,esf,b8zs

bchan=25-47

dchan=48

2. edit zapata.conf

- vi /etc/asterisk/zapata.conf

There are also many examples of how to configure zapata.conf online. we decided to separate our T1 into two line groups to keep some incoming calls from being busy if we filled up all of our lines.

Here's what we used(you can set echocancel=no if you are using PRIs):

[channels]

group=1

language=en

signalling=em_w

usecallerid=yes

callerid=asreceived

context=default

echocancel=64

```
echocancelwhenbridged=yes
rxgain=1.0
txgain=1.0
channel => 1-2
group=2
language=en
signalling=em_w
usecallerid=yes
callerid=asreceived
context=default
echocancel=64
echocancelwhenbridged=yes
rxgain=1.0
txgain=1.0
channel => 3-24
```

FOR A PRI YOU WOULD USE SOMETHING LIKE THIS:

```
group=3
language=en
signalling=pri_net
usecallerid=yes
callerid=asreceived
callprogress=no
busydetect=no
context=default
echocancel=64
echocancelwhenbridged=yes
rxgain=1.0
txgain=1.0
channel => 25-47
```

3. edit sip.conf

```
- vi /etc/asterisk/sip.conf
```

here is where we will edit the configuration of our SIP compatible phone devices. As stated at the beginning, we will be setting up a Grandstream Budgetone 102 phone and a Sipura SPA-2000 adapter with two analog phones connected(each with it's own extension). Here are the settings we used to set each of them up:

```
[general]
port = 5060
bindaddr = 0.0.0.0
context = default
; register SIP account on remote machine if using SIP trunks
; register => testSIPtrunk:test@10.10.10.16:5060
;
; setup account for SIP trunking:
; [SIPtrunk]
; disallow=all
; allow=ulaw
; allow=alaw
; type=friend
; username=testSIPtrunk
; secret=test
; host=10.10.10.16
; dtmfmode=inband
; qualify=1000
```

```
[gs102]
disallow=all
allow=ulaw
allow=alaw
type=friend
username=gs102
```

```

secret=test
host=dynamic
dtmfmode=inband
defaultip=10.10.10.16
qualify=1000
mailbox=102
[spa2000]
disallow=all
allow=ulaw
allow=alaw
type=friend
username=spa2000
secret=test
host=dynamic
dtmfmode=inband
defaultip=10.10.10.17
qualify=1000
mailbox=2000
[spa2001]
disallow=all
allow=ulaw
allow=alaw
type=friend
username=spa2001
secret=test
host=dynamic
dtmfmode=inband
defaultip=10.10.10.17
qualify=1000
mailbox=2001

```

4. edit meetme.conf

```
- vi /etc/asterisk/meetme.conf
```

This is known as the conference calling configuration file. We are just going to add two conferences (one without a pin number and one with a pin number required for entry):

```

[rooms]
conf => 8600
conf => 8601,1234

```

5. edit iax.conf (only to do IAX trunks, not required)

```
- vi /etc/asterisk/iax.conf
```

This is the IAX configuration file, below is a very simple config for having two Asterisk servers connect natively to each other, if you will be using this, make sure to add the optional lines included after the extensions.conf section. Also, there is an account setup here for a firefly IAX softphone to use. (details on that later)

*** IMPORTANT NOTE *** if you plan to use IAX2 trunks for VICIDIAL outbound dialing you must register with the remote IAX2 server through the iax.conf file, not just in the Dial or TRUNK line of the extensions.conf dialplan.

```

[general]
bindport=4569
iaxcompat=yes
bandwidth=high
allow=all
allow=gsm ; Always allow GSM, it's cool :)
jitterbuffer=no
tos=lowdelay
register => ASTtest1:test@10.10.10.16:4569

```

```

[ASTtest2]
type=friend
accountcode=IAXtrunk2

```

```
context=default
auth=plaintext
host=dynamic
permit=0.0.0.0/0.0.0.0
secret=test
disallow=all
allow=ulaw
qualify=yes
```

```
[firefly01]
type=friend
accountcode=firefly01
context=default
auth=plaintext
host=dynamic
permit=0.0.0.0/0.0.0.0
secret=test
disallow=all
allow=gsm
qualify=yes
```

EXAMPLE - This is a config example for setting up Binfone service(<http://www.binfone.com>)

NOTE: The "1112223333" is your iax username. When you signup, Binfone creates a default IAX username for you, (a 5 digit number, usually, starting with a 1). This works for most customers. Folks that want inbound then also sign up for DIDs, each of which has its own IAX username. (Which is the 10 digit DID). Each username has its own password which is managed through their web interface.

NOTE: If you will be using the G729 codec through binfone there is now a dedicated G729 gateway that only handles G729 calls. Please use this address to register to if you plan on using G729 as your codec: iax-g729.binfone.com

iax.conf:

```
[general]
register => 1112223333:PASSWORD@iax.binfone.com

[1112223333]
auth=md5
type=friend
username=1112223333
secret=PASSWORD
host=iax.binfone.com
context=incoming-IAX-context-in-extensions.conf
```

extensions.conf:

```
[global]
TRUNKBINPHONE=IAX2/1112223333:PASSWORD@iax.binfone.com ; IAX trunk interface

[default]
exten => _91NXXNXXXXXX,1,AGI(AGI(agi://127.0.0.1:4577/call_log)
exten => _91NXXNXXXXXX,2,Dial(${TRUNKIAX}/${EXTEN},55,o)
exten => _91NXXNXXXXXX,3,Hangup

[incoming]
exten => 1112223333,1,AGI(AGI(agi://127.0.0.1:4577/call_log)
exten => 1112223333,2,Dial(sip/g102,55,o)
exten => 1112223333,3,Hangup
```

dnsmgr.conf: # It is very helpful to enable dnsmgr

```
[general]
enable=yes ; enable creation of managed DNS lookups
refreshinterval=300 ; refresh managed DNS lookups every seconds
```

END EXAMPLE

6. edit voicemail.conf

```
- vi /etc/asterisk/voicemail.conf
```

This is where we set up the voicemail boxes for the extensions that we have set up:

```
[general]
format=wav49|gsm|wav
serveremail=asterisk
attach=yes
skipms=3000
maxsilence=10
silencethreshold=128
maxlogins=3
[zonemessages]
eastern=America/New_York|'vm-received' Q 'digits/at' IMp
central=America/Chicago|'vm-received' Q 'digits/at' IMp
central24=America/Chicago|'vm-received' q 'digits/at' H 'digits/hundred' M 'hours'
[default]
102 => 102,Grandstream Mailbox,root@localhost
2000 => 2000,Sipura Mailbox 1
2001 => 2001,Sipura Mailbox 2
3001 => 3001,Firefly Mailbox 1
```

7. edit manager.conf

```
- vi /etc/asterisk/manager.conf
```

This is where we set up remote logins to the asterisk manager interface, to allow sending of Action commands from remote connections to the Asterisk server, this will be important for the astguiclient applications so let's set that up now:

```
[general]
enabled = yes
port = 5038
bindaddr = 0.0.0.0
[cron]
secret = 1234
read = system,call,log,verbose,command,agent,user
write = system,call,log,verbose,command,agent,user
[updatecron]
secret = 1234
read = command
write = command
[listencron]
secret = 1234
read = system,call,log,verbose,command,agent,user
write = command
[sendcron]
secret = 1234
read = command
write = system,call,log,verbose,command,agent,user
```

8. edit logger.conf

```
- vi /etc/asterisk/logger.conf
```

This file determines the messages that are logged to the console and the /var/log/asterisk/messages file. We usually turn on full logging to the messages file to more easily diagnose any problems that we may run into, the problem with this is that it does produce very large

files, so be warned:

[logfiles]

console => notice,warning,error

messages => notice,warning,error,debug,verbose

9. edit extensions.conf

- vi /etc/asterisk/extensions.conf

This is known as the dialplan. Since we are installing a Long-Distance T1 with one 800 number on it, we will need to put that 800 number in the plan, as well as how to dial out through the T1 lines and we will need to add an entry for each of the phones that we have just set up in the sip.conf file. There are many examples both in the sample file and online for what to put in your dialplan, here is the simplified dialplan that we are using:

#####----- START extensions.conf example -----#####

[general]

static=yes

writeprotect=no

[globals]

CONSOLE=Console/dsp ; Console interface for demo

TRUNK=Zap/g1 ; Trunk interface

TRUNKX=Zap/g2 ; 2nd trunk interface

TRUNKIAX=IAX2/ASTtest1:test@10.10.10.16:4569; IAX trunk interface

[default]

; Extension 8600 + 8601 conference rooms

exten => 8600,1,Meetme,8600

exten => 8601,1,Meetme,8601

; Extension 102 - Grandstream hardphone

exten => 102,1,Playback,transfer|skip ; "Please hold while..."

exten => 102,2,Dial,sip/g102|20|to ; Ring, 20 secs max

exten => 102,3,VoiceMail,u102 ; Send to voicemail...

; Extension 2000 Sipura line 1

exten => 2000,1,Dial,sip/spa2000|30|to ; Ring, 30 secs max

exten => 2000,2,VoiceMail,u2000 ; Send to voicemail...

; Extension 2001 Sipura line 2

exten => 2001,1,Dial,sip/spa2001|30|to ; Ring, 30 secs max

exten => 2001,2,VoiceMail,u2001 ; Send to voicemail...

; Extension 2020 rings both sipura lines

exten => 2001,1,Dial,sip/spa2000&sip/spa2001|30|to ; Ring, 30 secs max

exten => 2001,2,VoiceMail,u2000 ; Send to voicemail...

; Extension 301 rings the firefly softphone

exten => 301,1,Dial,(IAX2/firefly01@firefly01/s)

exten => 301,2,Hangup

; Extension 3429 - Inbound 800 number (1-800-555-3429)

exten => _**3429,1,Ringing

exten => _**3429,2,Answer

exten => _**3429,3,Dial,sip/spa2000&sip/spa2001|30|to ; Ring, 30 secs max

exten => _**3429,4,VoiceMail,u2000 ; Send to voicemail...

; Extension 3429 - with ANI [callerID]

exten => _*NXXNXXXXXXX*3429,1,Ringing

exten => _*NXXNXXXXXXX*3429,2,Answer

exten => _*NXXNXXXXXXX*3429,3,Dial,sip/spa2000&sip/spa2001|30|to ; Ring, 30 secs max

exten => _*NXXNXXXXXXX*3429,4,VoiceMail,u2000 ; Send to voicemail...

; dial a long distance outbound number to the UK

exten => _901144XXXXXXXXXXXX,1,Dial(\${TRUNKX}/\${EXTEN:1},55,tTo)

exten => _901144XXXXXXXXXXXX,2,Hangup

```

; dial a long distance outbound number to Australia
exten => _901161XXXXXXXXX,1,Dial(${TRUNKX}/${EXTEN:1},,tTo)
exten => _901161XXXXXXXXX,2,Hangup

; dial an 800 outbound number
exten => _91800XXXXXXXX,1,Dial(${TRUNK}/${EXTEN:1},,tTo)
exten => _91800XXXXXXXX,2,Hangup
exten => _91888XXXXXXXX,1,Dial(${TRUNK}/${EXTEN:1},,tTo)
exten => _91888XXXXXXXX,2,Hangup
exten => _91877XXXXXXXX,1,Dial(${TRUNK}/${EXTEN:1},,tTo)
exten => _91877XXXXXXXX,2,Hangup
exten => _91866XXXXXXXX,1,Dial(${TRUNK}/${EXTEN:1},,tTo)
exten => _91866XXXXXXXX,2,Hangup

; dial a local 727 outbound number with area code
exten => _9727NXXXXXXXX,1,Dial(${TRUNK}/1${EXTEN:1},,tTo)
exten => _9727NXXXXXXXX,2,Hangup

; dial a local 813 outbound number with area code
exten => _9813NXXXXXXXX,1,Dial(${TRUNK}/${EXTEN:1},,tTo)
exten => _9813NXXXXXXXX,2,Hangup

; dial a long distance outbound number
exten => _91NXXNXXXXXXXX,1,Dial(${TRUNKX}/${EXTEN:1},,tTo)
exten => _91NXXNXXXXXXXX,2,Hangup

; dial a local outbound number (modified because of only LD T1)
exten => _9NXXXXXXXX,1,Dial(${TRUNK}/1727${EXTEN:1},,tTo)
exten => _9NXXXXXXXX,2,Hangup

; barge monitoring extension
exten => 8159,1,ZapBarge
exten => 8159,2,Hangup

; # timeout invalid rules
exten => #,1,Playback(invalid) ; "Thanks for trying the demo"
exten => #,2,Hangup ; Hang them up.
exten => t,1,Goto(#,1) ; If they take too long, give up
exten => i,1,Playback(invalid) ; "That's not valid, try again"

; Give voicemail at extension 8500
exten => 8500,1,VoiceMailMain
exten => 8500,2,Goto(s,6)

; ASTERISK AGENTS LOGINS FOR QUEUES (NOT part of VICIDIAL)
; the following assumes phone agent login and exten are 3 digits and the same
; also assumes that 3-digit login is present in agents.conf and queueus.conf
;Agent Logout then stay onhook, DIAL 54 + 3-digit ID
exten => _54XXX,1,AgentCallbackLogin(())
; the following are used to login and logout of Asterisk Queues from phone
;Agent Login then stay offhook on the phone, DIAL 55 + 3-digit ID
exten => _55XXX,1,AgentLogin(${EXTEN:1})
;Agent Login then stay onhook, phones will ring, DIAL 56 + 3-digit ID
exten => _56XXX,1,AgentCallbackLogin(||${EXTEN:1}@default)

#####----- END extensions.conf example -----#####

### OPTIONAL IAX trunk extensions entries for long distance dialing over IAX
exten => _91NXXNXXXXXXXX,1,AGI(AGI(agi://127.0.0.1:4577/call_log)
exten => _91NXXNXXXXXXXX,2,Dial(${TRUNKIAX}/${EXTEN},55,o)
exten => _91NXXNXXXXXXXX,3,Hangup

```

```
### OPTIONAL SIP trunk extensions entries for long distance dialing over SIP
    exten => _91NXXNXXXXXX,1,AGI(AGI(agi://127.0.0.1:4577/call_log)
    exten => _91NXXNXXXXXX,2,Dial(sip/${EXTEN}@SIPtrunk,55,o)
    exten => _91NXXNXXXXXX,3,Hangup
```

OPTIONAL IAX Load Balance extens to allow for Overflow and Balanced VDAD

In this setup, the serverIP is the prefix followed by agent conf_exten

FOR MORE INFORMATION, READ THE LOAD_BALANCING.txt DOCUMENT

server 1 extens:

```
    exten => _010*010*010*016*8600XXX,1,Dial(${TRUNKIAX2}/${EXTEN:16},55,o)
```

server 2 extens:

```
    exten => _010*010*010*015*8600XXX,1,Dial(${TRUNKIAX1}/${EXTEN:16},55,o)
```

10. edit dnsmgr.conf: # It is very helpful to enable dnsmgr

[general]

enable=yes ; enable creation of managed DNS lookups

refreshinterval=300 ; refresh managed DNS lookups every seconds

11. Now that you have configured Asterisk, it is time to try to start it for the first time.

- First make sure that your T1 line(or other telco line) is connected to the digium card.

- next type the following at the command prompt: "ztcfg -vvvvv"

- you should see a confirmation that the Zaptel device has loaded

- now you can launch asteris with the following command:

"asterisk -vvvvvvvvvvvc"

- you should see a lot of messages scroll by and at the end you should

be given a CLI> prompt if everything loaded OK. To get out of Asterisk you can

type "stop now". Now that you are sure it is running you can either run it in a

separate terminal window or use the start_asterisk_boot.pl that you will

install with astguiclient to start Asterisk.

SUBPHASE 5.1: setting up your SIP phones

You will need to follow the instructions for the phones that you are using with your system, but here's the way to set up a Grandstream 102 and a Sipura SPA-2000

1. Here are basic instructions for setting up a Grandstream BT 102:

- On the phone plug it in to power only at first and follow these instructions:

- wait for the phone to boot up and press the MENU button

- go through the menu screens with the menu key and the up/down arrow

keys to move from setting to setting. We are setting the following

values:

- DHCP OFF

- IP Addr: 010.010.010.016

- Subnet: 255.255.000.000

- router: 010.010.010.001

- dnS: 010.010.010.001

- tftp: we leave this blank for now

- menu 7 we change the codec to G-711u

- now wait 10 seconds and unplug the power and plug it back in

- you can also plug the network cable into the LAN port on the phone

- at this time you can go to your workstation and open a new web browser

- go to http://10.10.10.16/ the password is "admin"

- here is where you will enter in the configuration details for the

phone to register with the Asterisk server

- SIP server: 10.10.10.15

- SIP user ID: gs102
- Authenticate ID: gs102
- Password: test
- Name: gs102
- Voice Mail UserID: 102
- Send DTMF: in-audio
- NTP Server: tick.mit.edu
- then click update, click review changes, and click reboot
- your phone should now be able to register with the Asterisk server. If you still have your console screen up you should see a registration message appear telling you that gs102 has registered.

2. Here are the basic instructions for setting up a Sipura SPA-2000 analog adapter with 2 lines.

- Plug power and two analog phones into the adapter.
- pick up the phone plugged into line1 and press **** to enter admin menu
- press 101# then 0# to disable DHCP
- press 111# then 10*10*10*17# to change the IP address
- press 121# then 255*255*0*0# to change the subnet mask
- press 131# then 10*10*10*1# to change the default gateway
- hang up the phone, unplug the power, plug in the network cable and plug in the power cable
- now you can go to the admin website: <http://10.10.10.17/admin/advanced>
- you will need to make these setting changes:
- click on the "Line 1" tab at the top and change the following values:
 - Proxy: 10.10.10.15
 - Display Name: spa2000
 - userID: spa2000
 - password: test
 - authID: spa2000
 - change the dialplan to the following:
 (*xx|xxx|xxxx|xxxxx|xxxxxx|xxxxxxx|xxxxxxxx|xxxxxxxxxx|xxxxxxxxxxx|xxxxxxxxxxxx|xxxxxxxxxxxxx|xxxxxxxxxxxxxx|xxxxxxxxxxxxxxx|xxxxxxxxxxxxxx.)
- then click the "submit all changes" button at the bottom of the page
- and your first phone line should work now
- to register the second line, simply click on the "Line 2" tab and go through the above steps except use spa2001 instead of spa2000 for the use IDs

3. Now both of your phone devices are set up and you can try making phone calls between the three phones

SUBPHASE 5.2: setting up an IAX2 phone

This is optional and we won't go into too much detail about this, but currently there are several IAX hard and softphones on the market and more are coming every month. Follow the instructions with the IAX phone you have chosen and follow the steps below:

1. Add an entry into your iax.conf file like below if you have not already


```
[firefly01]
type=friend
accountcode=firefly01
context=default
auth=plaintext
host=dynamic
permit=0.0.0.0/0.0.0.0
secret=test
qualify=yes
```
2. Add an entry into your extensions.conf file like below if it is not in there


```
; Extension 3001 rings IAX phone
```

```
exten => 301,1,Dial(IAX2/firefly01@firefly01/s)
exten => 301,2,VoiceMail,u301 ; Send to voicemail...
```

3. Download Firefly 3rd party, or IDEfisk for Windows or Linux:

- <http://www.virbiage.com/firefly/download/firefly-thirdparty.exe>

MIRRORS:

- <http://mirror.isp.net.au/ftp/pub/firefly/firefly-thirdparty.exe>
- <http://www.freedomphones.net/files/firefly-thirdparty.exe>

IDEFISK:

- <http://www.asteriskguru.com/idefisk/>
 - Install the application
 - Launch Firefly Softphone
 - click the "I wish to connect to a 3rd party network" button
 - Enter in network name: Asterisk
 - Select IAX2 as the protocol
 - enter in your server address: "10.10.10.15" in our case
 - enter login and pass: "firefly01" and "test" for in our case
 - click OK and you should be logged in and can place calls

SUBPHASE 5.3: setting up a Zap phone

This is optional and we won't go into too much detail about this either, there are a few ways to use Zap devices as phones on your Asterisk system: Zaptel phone cards, Channel Banks going through Zaptel T1 card, outside line call coming in going through Zaptel line card. There isn't much to do but set your Zaptel config files up and put entries into your extensions.conf file:

1. Add an entry into your extensions.conf file like below

```
; Extension 4001 rings Zap phone
exten => 4001,1,Dial,Zap/1|30| ; ring Zap device 1
exten => 4001,2,VoiceMail,u4001 ; Send to voicemail...
```

PHASE 6: INSTALLING ASTGUICLIENT AND VICIDIAL

Now that Asterisk is installed and running we can add the astGUIClient and VICIDIAL components to the system.

SUBPHASE 6.0: putting the files in place

There are two methods for downloading astGUIClient/VICIDIAL, a release and SVN

1. Go to <http://astguiclient.sf.net/> and download the latest astguiclient package(as of this writing it is 2.0.4)

- for 2.0.X release:
 - mkdir /usr/src/astguiclient
 - cd /usr/src/astguiclient
 - wget http://internap.dl.sourceforge.net/sourceforge/astguiclient/astguiclient_2.0.4.zip
 - unzip astguiclient_2.0.4.zip
 - perl install.pl (make sure you are in the directory with the install.pl file)
 - for SVN 2.0 trunk:
 - mkdir /usr/src/astguiclient
 - cd /usr/src/astguiclient
 - svn checkout svn://svn.eflo.net:43690/agc_2-X/trunk
 - cd trunk
 - perl install.pl
 - select to do interactive setup and customize to your server
 - there is one more file you need that's not included with the download package, it's the conf.gsm file(this is the half-hour music file that we use to put people on hold). I have a free classical music file that is available free for download at the following two sites:
 - <http://www.freedomphones.net/conf.gsm>
 - <http://astguiclient.sf.net/conf.gsm>
- Once you have downloaded it, you will need to copy it to this folder:
 /var/lib/asterisk/sounds/
- Then you will need to execute this command to copy it as the park file
'cp /var/lib/asterisk/sounds/conf.gsm /var/lib/asterisk/sounds/park.gsm'
- Here are the steps spelled out:
- cd /var/lib/asterisk/sounds
 - wget <http://www.freedomphones.net/conf.gsm>
 - cp conf.gsm park.gsm
- you are done

SUBPHASE 6.1: creating the MySQL "asterisk" database

we will create the database and add a few initial records so that we can use the administrative web interface. Since this is a new install it is easier to use our new mysql script file to add the tables to the database:

1. at the command prompt type go to the mysql client:
/usr/local/mysql/bin/mysql
2. type the following into the mysql client prompt:
(make sure you put your IP address in place of "10.10.10.15" in the queries below)
#####----- BEGIN Mysql data entry(you can copy and paste this into terminal) #
create database asterisk;

```
GRANT SELECT,INSERT,UPDATE,DELETE,LOCK TABLES on asterisk.* TO cron@%' IDENTIFIED BY '1234';  
GRANT SELECT,INSERT,UPDATE,DELETE,LOCK TABLES on asterisk.* TO cron@localhost IDENTIFIED BY '1234';
```

NOTE: if using MySQL 4.1.12 or higher you may need to run this query too:
UPDATE mysql.user set password=OLD_PASSWORD('1234') where user='cron';

use asterisk;

\\. /usr/share/astguiclient/MySQL_AST_CREATE_tables.sql
or you may need to run this if you get an error:
\\. /usr/src/astguiclient/trunk/extras/MySQL_AST_CREATE_tables.sql

insert into servers (server_id,server_description,server_ip,active,asterisk_version)values('TESTasterisk','Test install of Asterisk server', '10.10.10.15','Y','1.2.17');

insert into server_updater values('10.10.10.15','');

insert into phones (extension, dialplan_number, voicemail_id, phone_ip, computer_ip, server_ip, login, pass, status, active, phone_type, fullname, company, picture, messages, old_messages, protocol)
values('gs102','102','102','10.10.10.16','10.10.9.16','10.10.10.15','gs102','test', 'ADMIN','Y','Grandstream BT 102','Test Admin Phone','TEST','','0','0','SIP');

insert into vicidial_users
(user,pass,full_name,user_level,user_group,load_leads,campaign_detail,ast_admin_access,modify_users)
values('6666','1234','Admin','9','ADMIN','1','1','1','1');

insert into conferences values('8600001','10.10.10.15','');
insert into conferences values('8600002','10.10.10.15','');
insert into conferences values('8600003','10.10.10.15','');
insert into conferences values('8600004','10.10.10.15','');
insert into conferences values('8600005','10.10.10.15','');
insert into conferences values('8600006','10.10.10.15','');
insert into conferences values('8600007','10.10.10.15','');
insert into conferences values('8600008','10.10.10.15','');
insert into conferences values('8600009','10.10.10.15','');
insert into conferences values('8600010','10.10.10.15','');
insert into conferences values('8600011','10.10.10.15','');
insert into conferences values('8600012','10.10.10.15','');
insert into conferences values('8600013','10.10.10.15','');
insert into conferences values('8600014','10.10.10.15','');
insert into conferences values('8600015','10.10.10.15','');
insert into conferences values('8600016','10.10.10.15','');
insert into conferences values('8600017','10.10.10.15','');
insert into conferences values('8600018','10.10.10.15','');
insert into conferences values('8600019','10.10.10.15','');
insert into conferences values('8600020','10.10.10.15','');
insert into conferences values('8600021','10.10.10.15','');
insert into conferences values('8600022','10.10.10.15','');
insert into conferences values('8600023','10.10.10.15','');
insert into conferences values('8600024','10.10.10.15','');
insert into conferences values('8600025','10.10.10.15','');
insert into conferences values('8600026','10.10.10.15','');
insert into conferences values('8600027','10.10.10.15','');
insert into conferences values('8600028','10.10.10.15','');
insert into conferences values('8600029','10.10.10.15','');

quit

run this command from command line:
- /usr/share/astguiclient/ADMIN_area_code_populate.pl

#####----- END Mysql data entry -----#####

NOTE: If you will be using channelbanks for agent phones you can use the

/extras/single_channelbank_phones.sql file to help enter the phones table entries.

NOTE: If you will be using IAX or SIP phones for agent phones you can use the /extras/sip-iax_phones.sql file to help enter the phones table entries.

NOTE: if you are not installing using default user/pass or have MySQL on another server, you will need to edit either the /etc/astguiclient.conf file or the dbconnect.php files in the astguiclient, vicidial and agc directories of your webroot.

3. Enter the vicidial administration page:

<http://10.10.10.15/vicidial/admin.php>

NOTE: if you click on the Logout button you must leave the user/pass empty and click OK

- Here you will enter the login and password that you inserted into the mysql database in the vicidial_users table (subphase 6.1 [6666/1234])

- Now that you are logged into the astGUIclient administration system we can add a new phone entry for each of the sipura lines we created.

- click on the "Admin" link at the top, then the "ADD PHONE" link below that and enter in the proper information for each of the new phone lines.

Here's what we entered for spa2000:

- Phone extension: spa2000

- Dialplan Number: 2000

- Voicemail Box: 2000

- Phone IP address: 10.10.10.17

- Computer IP address: 10.10.9.17

- Server IP: 10.10.10.15

- Login: spa2000

- Password: test

- Status: ACTIVE

- Active Account: Y

- Phone Type: Sipura SPA-2000 line 1

- Full Name: Sipura line 1 test

- Company: TEST

- Picture:

- for the next phone simply replace 2000 with 2001 in the above example

- now your phones are all set up in the astguiclient system and you can use this website to add new phones to be used with astguiclient and monitor the number of calls people are making.

- now your database is set up for the astguiclient conferences which will allow you to have over 6 remote parties that you called from your GUI client application in one conference.

- click on the "LIST ALL SERVERS" link at the top then click on the server to modify. Verify that the GMT time zone and all other fields are what you want them to be. There is a setting(Max VICIDIAL Trunks) that can be modified to limit the number of VICIDIAL outbound trunks that will be allowed to use on this server.

4. ****OPTIONAL**** For IAX clients you will need to use full phone name as the extension on the admin page entry: "firefly01@firefly01" for our IAX phone example previously. And do not forget to set the protocol on this page to IAX2

5. ****OPTIONAL**** For Zap clients you will need to use full Zap Channel name as the extension on the admin page entry: "1-1" for our Zap phone example previously. And do not forget to set the protocol on this page to Zap

SUBPHASE 6.2: making additions to your Asterisk conf files

Now that the database is set up and our phones have entries in the system we can make the additions to the running Asterisk system that will allow astguiclient

to work with it.

1. Add the call_log entries to all incoming/outgoing extensions entries:
- here is how our sample dialplan changes for adding call_log entries(only effected extension groups are show):

#####----- START extensions.conf changes for call_log -----#####

This 'h' exten is VERY important for VICIDIAL usage,

you will have problems if it is not in your dialplan!

exten => h,1,DeadAGI(agi://127.0.0.1:4577/call_log--HVcauses--PRI-----NODEBUG-----\${HANGUPCAUSE}-----
\${DIALSTATUS}-----\${DIALEDTIME}-----\${ANSWEREDTIME}))

; Extension 3429 - Inbound 800 number (1-800-555-3429)

exten => _**3429,1,Ringing

exten => _**3429,2,AGI(agi://127.0.0.1:4577/call_log)

exten => _**3429,3,Answer

exten => _**3429,4,Dial,sip/spa2000&sip/spa2001|30|to

exten => _**3429,5,Voicemail,u2000

; Extension 3429 - with ANI [callerID]

exten => _NXXNXXXXXXX*3429,1,Ringing

exten => _NXXNXXXXXXX*3429,2,AGI(agi://127.0.0.1:4577/call_log)

exten => _NXXNXXXXXXX*3429,3,Answer

exten => _NXXNXXXXXXX*3429,4,Dial,sip/spa2000&sip/spa2001|30|to

exten => _NXXNXXXXXXX*3429,5,Voicemail,u2000

; Extension 7275551212 - Inbound local number from PRI with 10 digit delivery

exten => 7275551212,1,Ringing

exten => 7275551212,2,Wait(1)

exten => 7275551212,3,AGI(agi://127.0.0.1:4577/call_log--fullCID--\${EXTEN}-----\${CALLERID}-----\${CALLERIDNUM}-----
\${CALLERIDNAME})

exten => 7275551212,4,Answer

exten => 7275551212,5,Dial,sip/spa2000&sip/spa2001|30|to

exten => 7275551212,6,Voicemail,u2000

; This 'o' Dial flag is VERY important for VICIDIAL on outbound calls,

; dial a long distance outbound number to the UK

exten => _901144XXXXXXXXXX,1,AGI(agi://127.0.0.1:4577/call_log)

exten => _901144XXXXXXXXXX,2,Dial(\${TRUNKX}/\${EXTEN:1},55,tTo)

exten => _901144XXXXXXXXXX,3,Hangup

; dial a long distance outbound number to Australia

exten => _901161XXXXXXXXXX,1,AGI(agi://127.0.0.1:4577/call_log)

exten => _901161XXXXXXXXXX,2,Dial(\${TRUNKX}/\${EXTEN:1},,tTo)

exten => _901161XXXXXXXXXX,3,Hangup

; dial an 800 outbound number

exten => _91800NXXXXXX,1,AGI(agi://127.0.0.1:4577/call_log)

exten => _91800NXXXXXX,2,Dial(\${TRUNK}/\${EXTEN:1},,tTo)

exten => _91800NXXXXXX,3,Hangup

exten => _91888NXXXXXX,1,AGI(agi://127.0.0.1:4577/call_log)

exten => _91888NXXXXXX,2,Dial(\${TRUNK}/\${EXTEN:1},,tTo)

exten => _91888NXXXXXX,3,Hangup

exten => _91877NXXXXXX,1,AGI(agi://127.0.0.1:4577/call_log)

exten => _91877NXXXXXX,2,Dial(\${TRUNK}/\${EXTEN:1},,tTo)

exten => _91877NXXXXXX,3,Hangup

exten => _91866NXXXXXX,1,AGI(agi://127.0.0.1:4577/call_log)

exten => _91866NXXXXXX,2,Dial(\${TRUNK}/\${EXTEN:1},,tTo)

exten => _91866NXXXXXX,3,Hangup

; dial a local 727 outbound number with area code

exten => _9727NXXXXXX,1,AGI(agi://127.0.0.1:4577/call_log)

```
exten => _9727NXXXXXX,2,Dial(${TRUNK}/1${EXTEN:1},,tTo)
exten => _9727NXXXXXX,3,Hangup
```

```
; dial a local 813 outbound number with area code
exten => _9813NXXXXXX,1,AGI(agi://127.0.0.1:4577/call_log)
exten => _9813NXXXXXX,2,Dial(${TRUNK}/${EXTEN:1},,tTo)
exten => _9813NXXXXXX,3,Hangup
```

```
; This 'o' Dial flag is VERY important for VICIDIAL on outbound calls,
; dial a long distance outbound number
exten => _91NXXNXXXXXX,1,AGI(agi://127.0.0.1:4577/call_log)
exten => _91NXXNXXXXXX,2,Dial(${TRUNKX}/${EXTEN:1},,tTo)
exten => _91NXXNXXXXXX,3,Hangup
```

```
; dial a local outbound number (modified because of only LD T1)
exten => _9NXXXXXX,1,AGI(agi://127.0.0.1:4577/call_log)
exten => _9NXXXXXX,2,Dial(${TRUNK}/1727${EXTEN:1},,tTo)
exten => _9NXXXXXX,3,Hangup
```

#####----- END extensions.conf changes -----#####

2. Add the call_inbound entries to all incoming extensions entries that you want CallerID popups on:

- here is how our sample dialplan changes for adding call_inbound entries(only effected extension groups are show):

#####----- START extensions.conf changes for call_inbound -----#####
; parameters for call_inbound.agi (7 fields separated by five dashes "-----"):
; 1. the extension of the phone to ring as defined in the asterisk.phones table
; 2. the phone number that was called, for the live_inbound/_log entry
; 3. a text description of the number that was called in
; 4-7. optional fields, they are also passed as fields in the GUI to web browser

```
; Extension 3429 - Inbound 800 number (1-800-555-3429)
exten => _**3429,1,Ringing
exten => _**3429,2,AGI(agi://127.0.0.1:4577/call_log)
exten => _**3429,3,AGI(call_inbound.agi,spa2000-----8005553429-----Inbound 800-----x-----y-----z-----w)
exten => _**3429,4,Answer
exten => _**3429,5,Dial,sip/spa2000&sip/spa2001|30|to
exten => _**3429,6,Voicemail,u2000
; Extension 3429 - with ANI [callerID]
exten => _*NXXNXXXXXX*3429,1,Ringing
exten => _*NXXNXXXXXX*3429,2,AGI(agi://127.0.0.1:4577/call_log)
exten => _*NXXNXXXXXX*3429,3,AGI(call_inbound.agi,spa2000-----8005553429-----Inbound 800-----x-----y-----z-----w)
exten => _*NXXNXXXXXX*3429,3,Answer
exten => _*NXXNXXXXXX*3429,4,Dial,sip/spa2000&sip/spa2001|30|to
exten => _*NXXNXXXXXX*3429,5,Voicemail,u2000
```

```
exten => _010*010*010*015*8600XXX,1,Goto(default,{EXTEN:16},1)
exten => _010*010*010*015*8600XXX*,1,Goto(default,{EXTEN:16},1)
exten => _8600XXX*,1,AGI(agi-VDADfixCXFER.agi)
```

; parameters for agi-VDAD_ALL_inbound.agi (9 fields separated by five dashes "-----"):

```
; 1. the method of call handling for the script:
; - CID - CID received, add record with phone number
; - CIDLOOKUP - Lookup CID to find record in whole system
; - CIDLOOKUPRL - Restrict lookup to one list
; - CIDLOOKUPRC - Restrict lookup to one campaign's lists
; - CLOSER - Closer calls from VICIDIAL frontends
; - ANI - ANI received, add record with phone number
; - ANILOOKUP - Lookup ANI to find record in whole system
```

```
;
; - ANILOOKUPRL - Restrict lookup to one list
;
; - 3DIGITID - Enter 3 digit code to go to agent
;
; - 4DIGITID - Enter 4 digit code to go to agent
;
; - 5DIGITID - Enter 5 digit code to go to agent
;
; - 10DIGITID - Enter 10 digit code to go to agent
;
2. the method of searching for an available agent:
;
; - LO - Load Balance Overflow only (priority to home server)
;
; - LB - Load Balance total system
;
; - SO - Home server only
;
3. the full name of the IN GROUP to be used in vicidial for the inbound call
;
4. the phone number that was called, for the log entry
;
5. the callerID or lead_id of the person that called(usually overridden)
;
6. the park extension audio file name if used
;
7. the status of the call initially(usually not used)
;
8. the list_id to insert the new lead under if it is new (and CID/ANI available)
;
9. the phone dialing code to insert with the new lead if new (and CID/ANI available)
;
10. the campaign_id to search within lists if CIDLOOKUPRC
;
inbound VICIDIAL call with CID delivery through T1 PRI
exten => 1234,1,Answer ; Answer the line
exten => 1234,2,AGI(agi-VDAD_ALL_inbound.agi,CID-----LB-----CL_GALLERIA-----7274515134-----Closer-----park-----
-999-----1)
exten => 1234,3,Hangup
```

```
; inbound VICIDIAL transfer calls [can arrive through PRI T1 crossover, IAX or SIP channel]
exten => _90009.,1,Answer ; Answer the line
exten => _90009.,2,AGI(agi-VDAD_ALL_inbound.agi,CLOSER-----LB-----CL_TESTCAMP-----7275551212-----Closer-----
park-----999-----1)
exten => _90009.,3,Hangup
exten => _990009.,1,Answer ; Answer the line
exten => _990009.,2,AGI(agi-VDAD_ALL_inbound.agi,CLOSER-----LB-----CL_TESTCAMP-----7275551212-----Closer-----
park-----999-----1)
exten => _990009.,3,Hangup
```

follow these instructions if you plan to have VICIDIAL agents take inbound or closer calls:

1. in VICIDIAL web admin "add a new in-group" (the above examples would be "CL_GALLERIA")
 - group IDs cannot contain spaces ' ' or dashes '-' or plusses '+'
 - if you are using a HEX color value make sure to include the hash '#' at the beginning
2. create a new campaign in VICIDIAL called "CLOSER" and set "allow inbound blended" to Y
3. check the CL_GALLERIA checknox in the "Allowed In-Groups" section
4. have agents log in to the CLOSER campaign and select the CL_GALLERIA in-group
5. they will now start receiving inbound calls
6. as calls come in, each call is inserted into the vicidial_list table under the list specified in the AGI string, In the above example that would be list 999
7. if you want to take closer calls from the campaign "TEST" you will need to create an in-group called "CL_TEST_" for internal closing(on the same system) or "CL_TEST_L" for local closing(closer on different system from fronter) and then the fronter will click on the "internal closer" button to send the call to a closer

* NOTE, you need to set the dial_level of the CLOSER campaign to 1 or higher for inbound/closers to work

#####----- END extensions.conf changes for call_inbound -----#####

3. Add the ZapBarge entries for all zap lines:

- here is how our sample dialplan changes for adding zapbarge line-specific entries(this is a pure addition, nothing is being modified):

```
#####----- START extensions.conf additions for direct zapbarges -----#####
; ZapBarge direct channel extensions
exten => 8612001,1,ZapBarge(1)
exten => 8612002,1,ZapBarge(2)
exten => 8612003,1,ZapBarge(3)
```

```
exten => 8612004,1,ZapBarge(4)
exten => 8612005,1,ZapBarge(5)
exten => 8612006,1,ZapBarge(6)
exten => 8612007,1,ZapBarge(7)
exten => 8612008,1,ZapBarge(8)
exten => 8612009,1,ZapBarge(9)
exten => 8612010,1,ZapBarge(10)
exten => 8612011,1,ZapBarge(11)
exten => 8612012,1,ZapBarge(12)
exten => 8612013,1,ZapBarge(13)
exten => 8612014,1,ZapBarge(14)
exten => 8612015,1,ZapBarge(15)
exten => 8612016,1,ZapBarge(16)
exten => 8612017,1,ZapBarge(17)
exten => 8612018,1,ZapBarge(18)
exten => 8612019,1,ZapBarge(19)
exten => 8612020,1,ZapBarge(20)
exten => 8612021,1,ZapBarge(21)
exten => 8612022,1,ZapBarge(22)
exten => 8612023,1,ZapBarge(23)
exten => 8612024,1,ZapBarge(24)
#####----- END extensions.conf additions for direct zapbarges -----#####
```

4. Add the meetme entries for astguiclient and VICIDIAL conferences to meetme.conf:

- here is how our sample meetme.conf file changes for adding conference entries (this is a pure addition, nothing is being modified):

```
#####----- START meetme.conf additions for conferences -----#####
```

```
conf => 8600001
conf => 8600002
conf => 8600003
conf => 8600004
conf => 8600005
conf => 8600006
conf => 8600007
conf => 8600008
conf => 8600009
conf => 8600010
conf => 8600011
conf => 8600012
conf => 8600013
conf => 8600014
conf => 8600015
conf => 8600016
conf => 8600017
conf => 8600018
conf => 8600019
conf => 8600020
conf => 8600021
conf => 8600022
conf => 8600023
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conf => 8600054
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conf => 8600114
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conf => 8600117
conf => 8600118
conf => 8600119
conf => 8600120
conf => 8600121
conf => 8600122
conf => 8600123
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conf => 8600191
conf => 8600192
conf => 8600193
conf => 8600194
conf => 8600195
conf => 8600196
conf => 8600197
conf => 8600198
conf => 8600199
conf => 8600200
#####----- END meetme.conf additions for conferences -----#####
```

5. Add the conference entries for astguiclient conferences:

- here is how our sample dialplan changes for adding conference entries (this is a pure addition, nothing is being modified):

```
#####----- START extensions.conf additions for agc conferences -----#####
exten => 8600001,1,Meetme,8600001|q
exten => 8600002,1,Meetme,8600002|q
exten => 8600003,1,Meetme,8600003|q
exten => 8600004,1,Meetme,8600004|q
exten => 8600005,1,Meetme,8600005|q
exten => 8600006,1,Meetme,8600006|q
exten => 8600007,1,Meetme,8600007|q
exten => 8600008,1,Meetme,8600008|q
exten => 8600009,1,Meetme,8600009|q
exten => 8600010,1,Meetme,8600010|q
exten => 8600011,1,Meetme,8600011|q
exten => 8600012,1,Meetme,8600012|q
exten => 8600013,1,Meetme,8600013|q
exten => 8600014,1,Meetme,8600014|q
exten => 8600015,1,Meetme,8600015|q
exten => 8600016,1,Meetme,8600016|q
exten => 8600017,1,Meetme,8600017|q
exten => 8600018,1,Meetme,8600018|q
exten => 8600019,1,Meetme,8600019|q
exten => 8600020,1,Meetme,8600020|q
exten => 8600021,1,Meetme,8600021|q
exten => 8600022,1,Meetme,8600022|q
exten => 8600023,1,Meetme,8600023|q
exten => 8600024,1,Meetme,8600024|q
exten => 8600025,1,Meetme,8600025|q
exten => 8600026,1,Meetme,8600026|q
exten => 8600027,1,Meetme,8600027|q
exten => 8600028,1,Meetme,8600028|q
exten => 8600029,1,Meetme,8600029|q
#####----- END extensions.conf additions for agc conferences -----#####
```

6. Add the conference entries for VICIDIAL conferences:

- here is how our sample dialplan changes for adding VICIDIAL conference entries(this is a pure addition, nothing is being modified):

NOTE: see below these entries for app_conference instructions is used

#####----- START extensions.conf changes for VD conf -----#####

exten => _X48600XXX,1,MeetMeAdmin(\${EXTEN:2},T,\${EXTEN:0:1})

exten => _X48600XXX,2,Hangup

exten => _X38600XXX,1,MeetMeAdmin(\${EXTEN:2},t,\${EXTEN:0:1})

exten => _X38600XXX,2,Hangup

exten => _X28600XXX,1,MeetMeAdmin(\${EXTEN:2},m,\${EXTEN:0:1})

exten => _X28600XXX,2,Hangup

exten => _X18600XXX,1,MeetMeAdmin(\${EXTEN:2},M,\${EXTEN:0:1})

exten => _X18600XXX,2,Hangup

exten => _55558600XXX,1,MeetMeAdmin(\${EXTEN:4},K)

exten => _55558600XXX,2,Hangup

exten => 8300,1,Hangup

exten => 8600051,1,Meetme,8600051

exten => 8600052,1,Meetme,8600052

exten => 8600053,1,Meetme,8600053

exten => 8600054,1,Meetme,8600054

exten => 8600055,1,Meetme,8600055

exten => 8600056,1,Meetme,8600056

exten => 8600057,1,Meetme,8600057

exten => 8600058,1,Meetme,8600058

exten => 8600059,1,Meetme,8600059

exten => 8600060,1,Meetme,8600060

exten => 8600061,1,Meetme,8600061

exten => 8600062,1,Meetme,8600062

exten => 8600063,1,Meetme,8600063

exten => 8600064,1,Meetme,8600064

exten => 8600065,1,Meetme,8600065

exten => 8600066,1,Meetme,8600066

exten => 8600067,1,Meetme,8600067

exten => 8600068,1,Meetme,8600068

exten => 8600069,1,Meetme,8600069

exten => 8600070,1,Meetme,8600070

exten => 8600071,1,Meetme,8600071

exten => 8600072,1,Meetme,8600072

exten => 8600073,1,Meetme,8600073

exten => 8600074,1,Meetme,8600074

exten => 8600075,1,Meetme,8600075

exten => 8600076,1,Meetme,8600076

exten => 8600077,1,Meetme,8600077

exten => 8600078,1,Meetme,8600078

exten => 8600079,1,Meetme,8600079

exten => 8600080,1,Meetme,8600080

exten => 8600081,1,Meetme,8600081

exten => 8600082,1,Meetme,8600082

exten => 8600083,1,Meetme,8600083

exten => 8600084,1,Meetme,8600084

exten => 8600085,1,Meetme,8600085

exten => 8600086,1,Meetme,8600086

exten => 8600087,1,Meetme,8600087

exten => 8600088,1,Meetme,8600088

exten => 8600089,1,Meetme,8600089

exten => 8600090,1,Meetme,8600090

exten => 8600091,1,Meetme,8600091

exten => 8600092,1,Meetme,8600092

exten => 8600093,1,Meetme,8600093
exten => 8600094,1,Meetme,8600094
exten => 8600095,1,Meetme,8600095
exten => 8600096,1,Meetme,8600096
exten => 8600097,1,Meetme,8600097
exten => 8600098,1,Meetme,8600098
exten => 8600099,1,Meetme,8600099
exten => 8600100,1,Meetme,8600100
exten => 8600101,1,Meetme,8600101
exten => 8600102,1,Meetme,8600102
exten => 8600103,1,Meetme,8600103
exten => 8600104,1,Meetme,8600104
exten => 8600105,1,Meetme,8600105
exten => 8600106,1,Meetme,8600106
exten => 8600107,1,Meetme,8600107
exten => 8600108,1,Meetme,8600108
exten => 8600109,1,Meetme,8600109
exten => 8600110,1,Meetme,8600110
exten => 8600111,1,Meetme,8600111
exten => 8600112,1,Meetme,8600112
exten => 8600113,1,Meetme,8600113
exten => 8600114,1,Meetme,8600114
exten => 8600115,1,Meetme,8600115
exten => 8600116,1,Meetme,8600116
exten => 8600117,1,Meetme,8600117
exten => 8600118,1,Meetme,8600118
exten => 8600119,1,Meetme,8600119
exten => 8600120,1,Meetme,8600120
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exten => 8600127,1,Meetme,8600127
exten => 8600128,1,Meetme,8600128
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exten => 8600144,1,Meetme,8600144
exten => 8600145,1,Meetme,8600145
exten => 8600146,1,Meetme,8600146
exten => 8600147,1,Meetme,8600147
exten => 8600148,1,Meetme,8600148
exten => 8600149,1,Meetme,8600149
exten => 8600150,1,Meetme,8600150
exten => 8600151,1,Meetme,8600151
exten => 8600152,1,Meetme,8600152
exten => 8600153,1,Meetme,8600153
exten => 8600154,1,Meetme,8600154

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exten => 8600155,1,Meetme,8600155
exten => 8600156,1,Meetme,8600156
exten => 8600157,1,Meetme,8600157
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exten => 8600160,1,Meetme,8600160
exten => 8600161,1,Meetme,8600161
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exten => 8600189,1,Meetme,8600189
exten => 8600190,1,Meetme,8600190
exten => 8600191,1,Meetme,8600191
exten => 8600192,1,Meetme,8600192
exten => 8600193,1,Meetme,8600193
exten => 8600194,1,Meetme,8600194
exten => 8600195,1,Meetme,8600195
exten => 8600196,1,Meetme,8600196
exten => 8600197,1,Meetme,8600197
exten => 8600198,1,Meetme,8600198
exten => 8600199,1,Meetme,8600199
exten => 8600200,1,Meetme,8600200
; quiet entry and leaving conferences for VICIDIAL
exten => _78600XXX,1,Meetme,${EXTEN:1}|q
; quiet monitor extensions for meetme rooms (for room managers)
exten => _68600XXX,1,Meetme,${EXTEN:1}|mq

```

#####----- END extensions.conf changes for VD conf -----#####

NOTE: If you want to do DTMF passthru with app_conference be sure to add the "i" and "t" flags to the 8600XX lines: Conference(8600051|it)

7. Add the more entries for astGUIClient specific uses:

- here are some more dialplan additions needed to use astGUIClient(this is a pure addition, nothing is being modified):

```

#####----- START extensions.conf other additions -----#####
; park channel for client GUI parking, hangup after 30 minutes

```

```
; create a GSM formatted audio file named "park.gsm" that is 30 minutes long
; and put it in /var/lib/asterisk/sounds
exten => 8301,1,Answer
exten => 8301,2,AGI(park_CID.agi)
exten => 8301,3,Playback(park)
exten => 8301,4,Hangup
exten => 8303,1,Answer
exten => 8303,2,AGI(park_CID.agi)
exten => 8303,3,Playback(conf)
exten => 8303,4,Hangup
```

```
; park channel for client GUI conferencing, hangup after 30 minutes
; create a GSM formatted audio file named "conf.gsm" that is 30 minutes long
; and put it in /var/lib/asterisk/sounds
exten => 8302,1,Answer
exten => 8302,2,Playback(conf)
exten => 8302,3,Hangup
```

```
exten => 8304,1,Answer
exten => 8304,2,Playback(ding)
exten => 8304,3,Hangup
```

```
; default audio for safe harbor 2-second-after-hello message then hangup
; create a GSM formatted audio file complies with safe harbor rules
; and put it in /var/lib/asterisk/sounds then change filename below
exten => 8307,1,Answer
exten => 8307,2,Playback(vm-goodbye)
exten => 8307,3,Hangup
```

```
; this is used for recording conference calls, the client app sends the filename
; value as a callerID recordings go to /var/spool/asterisk/monitor (WAV)
exten => 8309,1,Answer
exten => 8309,2,Monitor(wav,${CALLERIDNAME})
exten => 8309,3,Wait,3600
exten => 8309,4,Hangup
```

```
; this is used for recording conference calls, the client app sends the filename
; value as a callerID recordings go to /var/spool/asterisk/monitor (GSM)
exten => 8310,1,Answer
exten => 8310,2,Monitor(gsm,${CALLERIDNAME})
exten => 8310,3,Wait,3600
exten => 8310,4,Hangup
```

```
; this is used for playing a message to an answering machine forwarded from AMD in VICIDIAL
; replace conf with the message file you want to leave
exten => 8320,1,WaitForSilence(2000,2) ; AMD got machine. leave message after recording
exten => 8320,2,Playback(conf)
exten => 8320,3,AGI(VD_amd_post.agi,${EXTEN})
exten => 8320,4,Hangup
```

```
; this is used to allow the GUI to send you directly into voicemail
; don't forget to set GUI variable $voicemail_exten to this extension
exten => 8501,1,VoicemailMain(s${CALLERIDNUM})
exten => 8501,2,Hangup
```

```
; this is used to allow the GUI to send live calls directly into voicemail
; don't forget to set GUI variable $voicemail_dump_exten to this extension
exten => _8502666666666666.,1,Wait(2)
exten => _8502666666666666.,2,Voicemail(${EXTEN:14})
exten => _8502666666666666.,3,Hangup
```

```
; this is used for sending DTMF signals within conference calls, the client app
```

```
; sends the digits to be played in the callerID field
; sound files must be placed in /var/lib/asterisk/sounds
exten => 8500998,1,Answer
exten => 8500998,2,Playback(silence)
exten => 8500998,3,AGI(agi-dtmf.agi)
exten => 8500998,4,Hangup
```

```
; prompt recording AGI script, ID is 4321
exten => 8168,1,Answer
exten => 8168,2,AGI(agi-record_prompts.agi)
exten => 8168,3,Hangup
```

```
; playback of recorded prompts
exten => _851XXXXX,1,Answer
exten => _851XXXXX,2,Playback(${EXTEN})
exten => _851XXXXX,3,Hangup
```

VDAD STANDARD TRANSFER ENTRIES

```
; VICIDIAL_auto_dialer transfer script for no-agent campaigns:
exten => 8364,1,Playback(sip-silence)
exten => 8364,2,AGI(agi://127.0.0.1:4577/call_log)
exten => 8364,3,AGI(agi-VDADtransferBROADCAST.agi,${EXTEN})
exten => 8364,4,AGI(agi-VDADtransferBROADCAST.agi,${EXTEN})
exten => 8364,5,Hangup
```

```
; VICIDIAL_auto_dialer transfer script:
exten => 8365,1,AGI(agi://127.0.0.1:4577/call_log)
exten => 8365,2,AGI(agi-VDADtransfer.agi,${EXTEN})
exten => 8365,3,AGI(agi-VDADtransfer.agi,${EXTEN})
exten => 8365,4,AGI(agi-VDADtransfer.agi,${EXTEN})
exten => 8365,5,Hangup
```

```
; VICIDIAL_auto_dialer transfer script SURVEY at beginning:
exten => 8366,1,AGI(agi://127.0.0.1:4577/call_log)
exten => 8366,2,AGI(agi-VDADtransferSURVEY.agi,${EXTEN})
exten => 8366,3,AGI(agi-VDADtransferSURVEY.agi,${EXTEN})
exten => 8366,4,AGI(agi-VDADtransferSURVEY.agi,${EXTEN})
exten => 8366,5,Hangup
```

```
; VICIDIAL_auto_dialer transfer script Load Balance Overflow:
exten => 8367,1,AGI(agi://127.0.0.1:4577/call_log)
exten => 8367,2,AGI(agi-VDAD_LO_transfer.agi,${EXTEN})
exten => 8367,3,AGI(agi-VDAD_LO_transfer.agi,${EXTEN})
exten => 8367,4,AGI(agi-VDAD_LO_transfer.agi,${EXTEN})
exten => 8367,5,Hangup
```

```
; VICIDIAL_auto_dialer transfer script Load Balanced:
exten => 8368,1,AGI(agi://127.0.0.1:4577/call_log)
exten => 8368,2,AGI(agi-VDAD_LB_transfer.agi,${EXTEN})
exten => 8368,3,AGI(agi-VDAD_LB_transfer.agi,${EXTEN})
exten => 8368,4,AGI(agi-VDAD_LB_transfer.agi,${EXTEN})
exten => 8368,5,Hangup
```

```
; VICIDIAL_auto_dialer transfer script AMD with Load Balanced:
exten => 8369,1,AGI(agi://127.0.0.1:4577/call_log)
exten => 8369,2,AMD(3500|1500|300|5000|120|50|5|256)
exten => 8369,3,AGI(agi-VDAMD.agi,${EXTEN})
exten => 8369,4,AGI(agi-VDAD_LB_transfer.agi,${EXTEN})
exten => 8369,5,AGI(agi-VDAD_LB_transfer.agi,${EXTEN})
exten => 8369,6,AGI(agi-VDAD_LB_transfer.agi,${EXTEN})
exten => 8369,7,Hangup
```

```
; VICIDIAL auto-dial reminder script
exten => 8372,1,AGI(agi://127.0.0.1:4577/call_log)
exten => 8372,2,AGI(agi-VDADautoREMINDER.agi,{EXTEN})
exten => 8372,3,AGI(agi-VDADautoREMINDER.agi,{EXTEN})
exten => 8372,4,AGI(agi-VDADautoREMINDER.agi,{EXTEN})
exten => 8372,5,Hangup
```

VDAD SIP UNREGISTERED TRANSFER ENTRIES

Use these entries IN PLACE OF the entries above if you are using SIP trunks

and are not registering your provider in sip.conf

```
; VICIDIAL_auto_dialer transfer script:
```

```
exten => 8365,1,Playback(sip-silence)
exten => 8365,2,AGI(agi://127.0.0.1:4577/call_log)
exten => 8365,3,AGI(agi-VDADtransfer.agi,{EXTEN})
exten => 8365,4,AGI(agi-VDADtransfer.agi,{EXTEN})
exten => 8365,5,Hangup
```

```
; VICIDIAL_auto_dialer transfer script SURVEY at beginning:
```

```
exten => 8366,1,Playback(sip-silence)
exten => 8366,2,AGI(agi://127.0.0.1:4577/call_log)
exten => 8366,3,AGI(agi-VDADtransferSURVEY.agi,{EXTEN})
exten => 8366,4,AGI(agi-VDADtransferSURVEY.agi,{EXTEN})
exten => 8366,5,Hangup
```

```
; VICIDIAL_auto_dialer transfer script Load Balance Overflow:
```

```
exten => 8367,1,Playback(sip-silence)
exten => 8367,2,AGI(agi://127.0.0.1:4577/call_log)
exten => 8367,3,AGI(agi-VDAD_LO_transfer.agi,{EXTEN})
exten => 8367,4,AGI(agi-VDAD_LO_transfer.agi,{EXTEN})
exten => 8367,5,Hangup
```

```
; VICIDIAL_auto_dialer transfer script Load Balanced:
```

```
exten => 8368,1,Playback(sip-silence)
exten => 8368,2,AGI(agi://127.0.0.1:4577/call_log)
exten => 8368,3,AGI(agi-VDAD_LB_transfer.agi,{EXTEN})
exten => 8368,4,AGI(agi-VDAD_LB_transfer.agi,{EXTEN})
exten => 8368,5,Hangup
```

```
; VICIDIAL_auto_dialer transfer script AMD with Load Balanced:
```

```
exten => 8369,1,Playback(sip-silence)
exten => 8369,2,AGI(agi://127.0.0.1:4577/call_log)
exten => 8369,3,AMD(3500|1500|300|5000|120|50|5|256)
exten => 8369,4,AGI(VD_amd.agi,{EXTEN})
exten => 8369,5,AGI(agi-VDAD_LB_transfer.agi,{EXTEN})
exten => 8369,6,AGI(agi-VDAD_LB_transfer.agi,{EXTEN})
exten => 8369,7,Hangup
```

```
; VICIDIAL auto-dial reminder script
```

```
exten => 8372,1,Playback(sip-silence)
exten => 8372,2,AGI(agi://127.0.0.1:4577/call_log)
exten => 8372,3,AGI(agi-VDADautoREMINDER.agi,{EXTEN})
exten => 8372,4,AGI(agi-VDADautoREMINDER.agi,{EXTEN})
exten => 8372,5,Hangup
```

```
; MANDATORY VDAD extens:
```

```
; In this setup, the serverIP is the prefix followed by agent conf_exten
```

```
; These lines are REQUIRED for VICIDIAL to work properly
```

```
; local server extens:
```

```
exten => _010*010*010*015*8600XXX,1,Goto(default,{EXTEN:16},1)
exten => _010*010*010*015*8600XXX*,1,Goto(default,{EXTEN:16},1)
exten => _010*010*010*015*78600XXX,1,Goto(default,{EXTEN:16},1)
exten => _010*010*010*015*78600XXX*,1,Goto(default,{EXTEN:16},1)
```

```
exten => _8600XXX*,1,AGI(agi-VDADfixCXFER.agi)
exten => _78600XXX*,1,AGI(agi-VDADfixCXFER.agi)
```

; OPTIONAL server 2 extens, needed for load balancing:

```
exten => _010*010*010*016*8600XXX,1,Dial(${TRUNKIAX2}/${EXTEN:16},55,o)
exten => _010*010*010*016*8600XXX*,1,Dial(${TRUNKIAX2}/${EXTEN:16},55,o)
exten => _010*010*010*016*78600XXX,1,Dial(${TRUNKIAX2}/${EXTEN:16},55,o)
exten => _010*010*010*016*78600XXX*,1,Dial(${TRUNKIAX2}/${EXTEN:16},55,o)
```

SUBPHASE 6.3: adding entries to your MySQL "asterisk" database for vicidial applications

We need to add a few initial values to the vicidial tables in the "asterisk" database in order to start setting up the vicidial dialer system for use.

1. at the command prompt type this to go to the mysql client:

```
/usr/local/mysql/bin/mysql
```

2. type the following into the mysql client prompt:

(make sure you put your IP address in place of "10.10.10.15" in the queries below)

```
#####----- BEGIN Mysql data entry(you can copy and paste this into terminal) #
```

```
insert into vicidial_conferences values('8600051','10.10.10.15','');
insert into vicidial_conferences values('8600052','10.10.10.15','');
insert into vicidial_conferences values('8600053','10.10.10.15','');
insert into vicidial_conferences values('8600054','10.10.10.15','');
insert into vicidial_conferences values('8600055','10.10.10.15','');
insert into vicidial_conferences values('8600056','10.10.10.15','');
insert into vicidial_conferences values('8600057','10.10.10.15','');
insert into vicidial_conferences values('8600058','10.10.10.15','');
insert into vicidial_conferences values('8600059','10.10.10.15','');
insert into vicidial_conferences values('8600060','10.10.10.15','');
insert into vicidial_conferences values('8600061','10.10.10.15','');
insert into vicidial_conferences values('8600062','10.10.10.15','');
insert into vicidial_conferences values('8600063','10.10.10.15','');
insert into vicidial_conferences values('8600064','10.10.10.15','');
insert into vicidial_conferences values('8600065','10.10.10.15','');
insert into vicidial_conferences values('8600066','10.10.10.15','');
insert into vicidial_conferences values('8600067','10.10.10.15','');
insert into vicidial_conferences values('8600068','10.10.10.15','');
insert into vicidial_conferences values('8600069','10.10.10.15','');
insert into vicidial_conferences values('8600070','10.10.10.15','');
insert into vicidial_conferences values('8600071','10.10.10.15','');
insert into vicidial_conferences values('8600072','10.10.10.15','');
insert into vicidial_conferences values('8600073','10.10.10.15','');
insert into vicidial_conferences values('8600074','10.10.10.15','');
insert into vicidial_conferences values('8600075','10.10.10.15','');
insert into vicidial_conferences values('8600076','10.10.10.15','');
insert into vicidial_conferences values('8600077','10.10.10.15','');
insert into vicidial_conferences values('8600078','10.10.10.15','');
insert into vicidial_conferences values('8600079','10.10.10.15','');
insert into vicidial_conferences values('8600080','10.10.10.15','');
insert into vicidial_conferences values('8600081','10.10.10.15','');
insert into vicidial_conferences values('8600082','10.10.10.15','');
insert into vicidial_conferences values('8600083','10.10.10.15','');
insert into vicidial_conferences values('8600084','10.10.10.15','');
insert into vicidial_conferences values('8600085','10.10.10.15','');
insert into vicidial_conferences values('8600086','10.10.10.15','');
insert into vicidial_conferences values('8600087','10.10.10.15','');
insert into vicidial_conferences values('8600088','10.10.10.15','');
insert into vicidial_conferences values('8600089','10.10.10.15','');
insert into vicidial_conferences values('8600090','10.10.10.15','');
```

[illegible]

```

insert into vicidial_conferences values('8600153','10.10.10.15','');
insert into vicidial_conferences values('8600154','10.10.10.15','');
insert into vicidial_conferences values('8600155','10.10.10.15','');
insert into vicidial_conferences values('8600156','10.10.10.15','');
insert into vicidial_conferences values('8600157','10.10.10.15','');
insert into vicidial_conferences values('8600158','10.10.10.15','');
insert into vicidial_conferences values('8600159','10.10.10.15','');
insert into vicidial_conferences values('8600160','10.10.10.15','');
insert into vicidial_conferences values('8600161','10.10.10.15','');
insert into vicidial_conferences values('8600162','10.10.10.15','');
insert into vicidial_conferences values('8600163','10.10.10.15','');
insert into vicidial_conferences values('8600164','10.10.10.15','');
insert into vicidial_conferences values('8600165','10.10.10.15','');
insert into vicidial_conferences values('8600166','10.10.10.15','');
insert into vicidial_conferences values('8600167','10.10.10.15','');
insert into vicidial_conferences values('8600168','10.10.10.15','');
insert into vicidial_conferences values('8600169','10.10.10.15','');
insert into vicidial_conferences values('8600170','10.10.10.15','');
insert into vicidial_conferences values('8600171','10.10.10.15','');
insert into vicidial_conferences values('8600172','10.10.10.15','');
insert into vicidial_conferences values('8600173','10.10.10.15','');
insert into vicidial_conferences values('8600174','10.10.10.15','');
insert into vicidial_conferences values('8600175','10.10.10.15','');
insert into vicidial_conferences values('8600176','10.10.10.15','');
insert into vicidial_conferences values('8600177','10.10.10.15','');
insert into vicidial_conferences values('8600178','10.10.10.15','');
insert into vicidial_conferences values('8600179','10.10.10.15','');
insert into vicidial_conferences values('8600180','10.10.10.15','');
insert into vicidial_conferences values('8600181','10.10.10.15','');
insert into vicidial_conferences values('8600182','10.10.10.15','');
insert into vicidial_conferences values('8600183','10.10.10.15','');
insert into vicidial_conferences values('8600184','10.10.10.15','');
insert into vicidial_conferences values('8600185','10.10.10.15','');
insert into vicidial_conferences values('8600186','10.10.10.15','');
insert into vicidial_conferences values('8600187','10.10.10.15','');
insert into vicidial_conferences values('8600188','10.10.10.15','');
insert into vicidial_conferences values('8600189','10.10.10.15','');
insert into vicidial_conferences values('8600190','10.10.10.15','');
insert into vicidial_conferences values('8600191','10.10.10.15','');
insert into vicidial_conferences values('8600192','10.10.10.15','');
insert into vicidial_conferences values('8600193','10.10.10.15','');
insert into vicidial_conferences values('8600194','10.10.10.15','');
insert into vicidial_conferences values('8600195','10.10.10.15','');
insert into vicidial_conferences values('8600196','10.10.10.15','');
insert into vicidial_conferences values('8600197','10.10.10.15','');
insert into vicidial_conferences values('8600198','10.10.10.15','');
insert into vicidial_conferences values('8600199','10.10.10.15','');
insert into vicidial_conferences values('8600200','10.10.10.15','');

```

#####----- END Mysql data entry -----#####

SUBPHASE 6.4: setting up asterisk and helper applications for startup

1. Make several entries in the rc.local of your system:

- on the command line type:
 - vi /etc/rc.d/rc.local
 - add the following entries(here's what we used):

```

### uncomment If kernel RAM drive is enabled
# mke2fs -m 0 /dev/ram0

```



```
# mount /dev/ram0 /var/spool/asterisk/monitor

### Path to local asterisk-perl_0.08 module if needed
PERL5LIB="/usr/share/astguiclient/libs"; export PERL5LIB

### start time server
/usr/local/bin/ntpdate -u 18.145.0.30
/usr/sbin/ntpd

### start up the MySQL server
/usr/local/mysql/bin/mysqld_safe --user=mysql --skip-name-resolve --skip-host-cache &

### start up the MySQL 4.1.X server (with old passwords)
/usr/local/mysql/bin/safe_mysqld --old-passwords --skip-name-resolve --skip-host-cache &

### start up the apache web server
/usr/local/apache2/bin/apachectl start

### roll the Asterisk logs upon reboot
/usr/share/astguiclient/ADMIN_restart_roll_logs.pl

### clear the server-related records from the database
/usr/share/astguiclient/AST_reset_mysql_vars.pl

### load digium tormenta 4xT1 drivers into system
modprobe zaptel
modprobe wct1xxp
/sbin/ztcfg -vvvvvvvvvvv

### sybsys local login
touch /var/lock/subsys/local

### sleep for 20 seconds before launching Asterisk
sleep 20

### start up asterisk
/usr/share/astguiclient/start_asterisk_boot.pl

- you are done
```

SUBPHASE 6.5: setting up astguiclient scripts for continuous running

1. Make several entries in the crontab of your system:

- on the command line type:
 - cd /usr/share/astguiclient
 - crontab -e
 - add the following entries(here's what we used):
- (Note if you have not edited the AST_CRON_mix_recordings.pl script to FTP to a different server, then you should use the AST_CRON_mix_recordings_BASIC.pl file in it's place)

```
### recording mixing for Asterisk run every 5 minutes
1,6,11,16,21,26,31,36,41,46,51,56 * * * 1,2,3,4,5,6 /usr/share/astguiclient/AST_CRON_mix_recordings_BASIC.pl

### keepalive script for astguiclient processes
* * * * * /usr/share/astguiclient/ADMIN_keepalive_ALL.pl

### kill Hangup script for Asterisk updaters
* * * * * /usr/share/astguiclient/AST_manager_kill_hung_congested.pl
```

```

### updater for voicemail
* * * * * /usr/share/astguiclient/AST_vm_update.pl

### updater for conference validator
* * * * * /usr/share/astguiclient/AST_conf_update.pl

### flush queue DB table every hour for entries older than 1 hour
11 * * * * /usr/share/astguiclient/AST_flush_DBqueue.pl -q

### fix the vicidial_agent_log once every hour
33 * * * * /usr/share/astguiclient/AST_cleanup_agent_log.pl

### updater for VICIDIAL hopper
* * * * * /usr/share/astguiclient/AST_VDhopper.pl -q

### adjust the GMT offset for the leads in the vicidial_list table
1 1,7 * * * * /usr/share/astguiclient/ADMIN_adjust_GMTnow_on_leads.pl --debug --postal-code-gmt

### reset several temporary-info tables in the database
2 1 * * * /usr/share/astguiclient/AST_reset_mysql_vars.pl

### optimize the database tables within the asterisk database
3 1 * * * /usr/share/astguiclient/AST_DB_optimize.pl

## adjust time on the server with ntp
30 * * * * /usr/local/bin/ntpdate -u 18.145.0.30 2>/dev/null 1>&2

### VICIDIAL agent time log weekly and daily summary report generation
2 0 * * 0 /usr/share/astguiclient/AST_agent_week.pl
22 0 * * * /usr/share/astguiclient/AST_agent_day.pl

### VICIDIAL campaign export scripts (OPTIONAL)
#32 0 * * * /usr/share/astguiclient/AST_VDsales_export.pl
#42 0 * * * /usr/share/astguiclient/AST_sourceID_summary_export.pl

### remove old recordings more than 7 days old
24 0 * * * /usr/bin/find /var/spool/asterisk/monitor -maxdepth 2 -type f -mtime +7 -print | xargs rm -f

### remove old vicidial logs and asterisk logs more than 2 days old
28 0 * * * /usr/bin/find /var/log/astguiclient -maxdepth 1 -type f -mtime +2 -print | xargs rm -f
29 0 * * * /usr/bin/find /var/log/asterisk -maxdepth 3 -type f -mtime +2 -print | xargs rm -f

```

- once your system starts up you can attach to the screen running asterisk by typing "screen -r " find which screen by typing "screen -r" and looking for the lowest screen number. Then to detach again from the screen while you are in it type 'Ctrl+a' then 'd'

- you are done

NOTES:

- The AST_agent_day.pl and AST_agent_week.pl scripts create an ASCII fixed-length report of all agent activity on the system
- The AST_VDsales_export.pl script allows for the exporting(into several different formats) of specified vicidial_list data based on status and campaign as well as inbound group
- The AST_sourceID_summary_export.pl script summarizes leads in the system by source_id and generates a text report

SUBPHASE 6.6: adding test leads to the VICIDIAL database and configuring a VICIDIAL campaign and users

First we will add a few test leads to the vicidial_list table so that we can test our system. There is also an application included with the distribution that will accept a delimited file of leads placed in the /usr/share/astguiclient/VICIDIAL/LEADS_IN/ directory and load it into the database automatically(VICIDIAL_IN_new_leads_file.pl [a sample lead file in the proper format is included with this release:

- test_VICIDIAL_lead_file.txt])

If you want to use the lead import script I suggest looking at the code to make sure it is entering what you want it to. We are not going to go over that particular script in this document.

Also, there is a web-based lead loader that was made available as of the 1.1.1 release and is accessible from the VICIDIAL admin.php web page(click on the "LOAD NEW LEADS" link at the top of the admin page). To get to this page you must have permissions in the vicidial_user table(Load Leads set to 1) . Instructions on it's use are included on the page through the help question mark link.

NOTE: in PHP you must have "fileuploads" enabled for this page to work.

NOTE: it is important to have your proper country code in the phone_code field of your leads so that the GMT offset encoding will properly code the time zones for your leads. For the USA/Canada/Caribbean this would be 1. For the UK this is 44 and Mexico is 52 and so on.

Second we need to add the disposition statuses into the system, all of these queries are below:

(Note: you may want to replace 7275551212 with a real number to test in these records)

1. at the command prompt type go to the mysql client:

/usr/local/mysql/bin/mysql

2. type the following into the mysql client prompt:

(make sure you put your IP address in place of "10.10.10.15" in the queries below)

#####----- BEGIN Mysql data entry(you can copy and paste this into terminal) #

```
insert into vicidial_list values('','2004-01-06','','NEW','','TEST01','101','TESTCAMP','N','1','7275551212','Mr','Matt','X','lead01','1234 Fake St.','','Clearwater','FL','','33760','USA','M','1970-01-01','','test@test.com','suprise','comments go here','0');
insert into vicidial_list values('','2004-01-06','','NEW','','TEST01','101','TESTCAMP','N','1','7275551212','Mr','Matt','X','lead02','1234 Fake St.','','Clearwater','FL','','33760','USA','M','1970-01-01','','test@test.com','suprise','comments go here','0');
insert into vicidial_list values('','2004-01-07','','NEW','','TEST01','101','TESTCAMP','N','1','7275551212','Mr','Matt','X','lead03','1234 Fake St.','','Clearwater','FL','','33760','USA','M','1970-01-01','','test@test.com','suprise','comments go here','0');
insert into vicidial_list values('','2004-01-07','','NEW','','TEST01','101','TESTCAMP','N','1','7275551212','Mr','Matt','X','lead04','1234 Fake St.','','Clearwater','FL','','33760','USA','M','1970-01-01','','test@test.com','suprise','comments go here','0');
insert into vicidial_list values('','2004-01-07','','NEW','','TEST01','101','TESTCAMP','N','1','7275551212','Mr','Matt','X','lead05','1234 Fake St.','','Clearwater','FL','','33760','USA','M','1970-01-01','','test@test.com','suprise','comments go here','0');
insert into vicidial_list values('','2004-01-07','','NEW','','TEST01','101','TESTCAMP','N','1','7275551212','Mr','Matt','X','lead06','1234 Fake St.','','Clearwater','FL','','33760','USA','M','1970-01-01','','test@test.com','suprise','comments go here','0');
insert into vicidial_list values('','2004-01-07','','NEW','','TEST01','101','TESTCAMP','N','1','7275551212','Mr','Matt','X','lead07','1234 Fake St.','','Clearwater','FL','','33760','USA','M','1970-01-01','','test@test.com','suprise','comments go here','0');
```

these first 5 must be in all VICIDIAL systems for it to work properly

```
insert into vicidial_statuses values('NEW','New Lead','N','N','UNDEFINED');
```

```
insert into vicidial_statuses values('QUEUE','Lead To Be Called','N','N','UNDEFINED');
```

```

insert into vicidial_statuses values('INCALL','Lead Being Called','N','N','UNDEFINED');
insert into vicidial_statuses values('DROP','Agent Not Available','N','Y','UNDEFINED');
insert into vicidial_statuses values('NA','No Answer AutoDial','N','N','UNDEFINED');

insert into vicidial_statuses values('CALLBK','Call Back','Y','Y','UNDEFINED');
insert into vicidial_statuses values('A','Answering Machine','Y','N','UNDEFINED');
insert into vicidial_statuses values('AA','Answering Machine Auto','N','N','UNDEFINED');
insert into vicidial_statuses values('AM','Answering Machine Sent to Mesg','N','N','UNDEFINED');
insert into vicidial_statuses values('AL','Answering Machine Msg Played','N','N','UNDEFINED');
insert into vicidial_statuses values('B','Busy','Y','N','UNDEFINED');
insert into vicidial_statuses values('DC','Disconnected Number','Y','N','UNDEFINED');
insert into vicidial_statuses values('DEC','Declined Sale','Y','Y','UNDEFINED');
insert into vicidial_statuses values('DNC','DO NOT CALL','Y','Y','UNDEFINED');
insert into vicidial_statuses values('SALE','Sale Made','Y','Y','UNDEFINED');
insert into vicidial_statuses values('N','No Answer','Y','N','UNDEFINED');
insert into vicidial_statuses values('NI','Not Interested','Y','Y','UNDEFINED');
insert into vicidial_statuses values('NP','No Pitch No Price','Y','Y','UNDEFINED');
insert into vicidial_statuses values('XFER','Call Transferred','Y','Y','UNDEFINED');

```

```

quit
#####----- END Mysql data entry -----#####

```

Now that the sample leads and disposition codes have been entered, we can go into the VICIDIAL administration website and set up our campaigns, lists and users. But first let's make sure that they have the right GMT offset:

run this on the command line:

```
- /usr/share/astguiclient/ADMIN_adjust_GMTnow_on_leads.pl --postal-code-gmt
```

3. Enter the astguiclient administration page:

<http://10.10.10.15/vicidial/admin.php>

(use the username and password created when we entered a record into the vicidial_users table in SUBPHASE 6.1, In our case this is 6666 and 1234)

NOTE: if you click on the Logout button you must leave the user/pass empty and click OK

- Now that you are logged into the vicidial administration system we can add new user entries for each of the new users and enter new campaigns and new lists.

- The first step is to enter your new users, Click on the ADD A NEW USER and fill in the appropriate information for each new user you want to add.

- Next, you need to create a new campaign, click on the ADD A NEW CAMPAIGN link and fill in what you want the campaign to be called as well as a description

- Next, you need to define a new list, click on the ADD A NEW LIST link and fill in what you want the list to be called as well as a using the list ID of the leads that we loaded in the previous step "101" and select the new campaign from the pull-down menu that we just created.

- Now that you have created your list, make it active by changing active to Y

- now modify your campaign and change the first status to be called to NEW and submit. Now your system is ready to dial.

- you are done

SUBPHASE 6.7: VICIDIAL remote agents:

With v1.0 of VICIDIAL we have the ability to use a simple web form to give remote agents a way to receive calls to whatever number they happen to be at, and they can view/edit call details and see a call log all through a web page (vdremote.php) or <http://10.10.10.15/vicidial/vdremote.php> on this installation. Remote Agents is only recommended for inbound calls because of the extra time needed to dial a number out to transfer the call to. To set up remote agents, just go to the vicidial admin.php page and ADD NEW REMOTE AGENTS(Make sure the

userID start also has a user login so they can get to the vremote page). You will see that you can set up a remote agent entry to take multiple lines if you wish meaning that, for example, if you need to send all of your calls to another location because of a massive snowstorm (and none of your agents showed up at work) you just log in that remote agent record with say 10 lines and then all of those calls will be directed to the same number you set up for the remote agent. Then again you could just get your agents to log in from home if they have a phone and computer

SUBPHASE 6.8: astGUIclient web-only client:

With 1.1.1 release of astguiclient we have completely rewritten the astGUIclient client app in AJAX(PHP/Javascript/XMLHttpRequest) to enable a full, real-time GUI interface using only a web browser. The browser requirements for this are:

- Firefox 0.9 or greater (Firefox 1.0.7 is the recommended browser)
- Mozilla 1.7 or greater
- Netscape 8 or greater
- Opera 8.5 or greater
- Microsoft Internet Explorer 6.0

This new version also has more flexibility and functionality than the perl/Tk version as well as being prettier. We have successfully tested this on many platforms and in remote locations. It functioned wonderfully off-site with one of our IAX hardphones and offers a lot of promise for road warriors who need a lot of options on their phone usage like conferencing and a detailed call log.

To log into this app you will need a login setup in the vicidial_users table with a user_level of 1 or greater as well as an entry for the phone you are using in the phones table. You will first get a login prompt for the vicidial login then you will have the phone login where you enter the Login and Password for that phone entry. From there the app should display and you will see the MAIN screen with your phone information, voicemail display and your inbound/outbound phone call log.

The example web page you would go to on this installation would be:
<http://10.10.10.15/agc/astguiclient.php>

The inbound log and callerID popup is dependant on having a call_inbound.agi entry in your dialplan before you phone is dialed (see subphase 6.2 step 2)

Another thing to note is that you can have the agc folder (with the .php files in it) copied to multiple web servers, you just need to make sure that the MySQL database connection works (check the settings in the dbconnect.php file that is in the agc directory). We have had astguiclient.php running on 3 separate web servers for the same DB server and Asterisk server. This is an easy way to allow for auto failover and/or redundancy. Also, this client will work over SSL connections (https) for encrypted communications with the server.

New in astGUIclient release 1.1.7 is multi-language support. multi-language versions of web-clients and admin pages are available in the LANG_www directory and can be unzipped into your webroot directory.

SUBPHASE 6.9: VICIDIAL web-only client:

NOTE: There is a VICIDIAL Agent manual available from <http://www.eflo.net>

With 1.1.6 release of astguiclient we have finished the rewrite of the VICIDIAL client app in AJAX(PHP/Javascript/XMLHttpRequest) to enable a full, real-time

GUI interface using only a web browser like we have done with astGUIclient.

The browser requirements for this are:

- Firefox 0.9 or greater (Firefox 1.0.7 is the recommended browser)
- Mozilla 1.7 or greater
- Netscape 8 or greater
- Opera 8.5 or greater
- Microsoft Internet Explorer 6.0

This version is fully functional and has been tested in our production call center with no problems. On RedHat 9.0 Linux workstations we have noticed a decrease in system load running the web-client as compared to the Perl/Tk client. Also, the memory usage stayed at about 22MB for a full shift of activity, so no memory leaks with Firefox.

To log into this app you will need a login setup in the vicidial_users table with a user_level of 1 or greater as well as an entry for the phone you are using in the phones table. You will first get a login prompt for the vicidial login then you will have the phone login where you enter the Login and Password for that phone entry. From there the app should display and you will see the VICIDIAL screen with your phone information.

The example web page you would go to on this installation would be:

<http://10.10.10.15/agc/vicidial.php>

One more feature that the VICIDIAL web-client offers is the ability to set up an EXTERNAL phone extension in the astguiclient admin section so that you can have agents log in to vicidial.php wherever they have access to a phone with an external phone number and a web browser. To do this follow these steps:

- "ADD PHONE" in the admin.php web page and enter whatever name you want
- For the dialplan number field put in the full digits that you would dial from the Asterisk server to get to that agent's external phone(with 91 if used)
- For the Protocol select EXTERNAL
- make sure the agent knows the login and password set for this phone entry.

Then the agent will go to the vicidial.php page and enter in their phone login/pass, their vicidial user/pass/campaign and their phone should ring in a few seconds, and they are logged in and ready to take calls.

Another thing to note is that you can have the agc folder(with the .php files in it) copied to multiple web servers, you just need to make sure that the MySQL database connection works (check the settings in the dbconnect.php file that is in the agc directory). We have had astguiclient.php running on 3 separate web servers for the same DB server and Asterisk server. This is an easy way to allow for auto failover and/or redundancy. Also, this client will work over SSL connections(https) for encrypted communications with the server.

New in astGUIclient release 1.1.7 is multi-language support. multi-language versions of web-clients and admin pages are available in the LANG_www directory and can be unzipped into your webroot directory.

Admin Note: If you want to enable your agents to login with only their user/pass you can hardcode the phone_login and phone_pass into the query string(URL) and use a bookmark on their desktop, taking one more step out of their login process example: <http://10.10.10.15/agc/vicidial.php?pl=gs102&pp=test>

It is recommended if you are in a call center environment that you would disable the "Saved Form Information" option in Firefox settings. This is a checkbox in the Privacy settings under the Options menu.

PHASE 7.0: You are done with installation

If you have problems and it is not working right (and are NOT celebrating right now), feel free to take a look at the FAQ for solutions to common installation errors, read the SCRATCH_INSTALL document, visit the VICIDIAL forum or send an email to the mailing list:

<http://www.eflo.net/VICIDIALforum/index.php>

<https://lists.sourceforge.net/lists/listinfo/astguiclient-users>

Also, check out our weblog: <http://astguiclient.blogspot.com/>

**** IMPORTANT - In order for vicidial/astguiclient to function correctly please read the REQUIREMENTS.txt for a minimum requirements list. ***

End-user Manuals for Agents and Managers are available from <http://www.eflo.net>