Start and Stop Asterisk

asterisk	Start Asterisk. You have to be root!
asterisk -c	Start Asterisk and open the Command Line Interface (CLI).
asterisk -r	Open the Command Line Interface (CLI) of an allready running Asterisk.
stop now	Stop a running Asterisk in the CLI (all open channels will be killed).
stop gracefully	Stop a running Asterisk when all active channels have completed. No new calles will be accept. CLI command.
stop when convenient	Stop a running Asterisk when all active channels have completed. New calls are accepted till then. The system will wait until nothing is going on at all. CLI command.
exit	Quits a running CLI. Does not stop Asterisk itself.

Directories and Files

`/etc/asterisk/`	All Asterisk configuration files are under directory `/etc/asterisk/`.
`/etc/asterisk/sip.conf`	Configfile for all SIP channels.
`/etc/asterisk/iax.conf`	Configfile for all IAX channels.
`/etc/asterisk/extensions.conf`	Configfile for the dialplan.
`/var/lib/asterisk/sounds/`	All Sound Prompts.
`/var/log/asterisk/`	All log files.
`/var/spool/asterisk/`	All call files.

Important CLI Commands

set verbose *5*	Sets verbose level (1-10).
set debug *5*	Sets debug level (1-10).
sip show peers	Lists all registered peers.
sip show channels	Lists all active channels.

Help within the CLI

help	Lists all available commands.
help sip	Lists all **sip** commands.
help sip show	Lists all **sip show** commands.
help sip show peers	Show help for the command **sip show peers**.

Regex in the extensions.conf

X	0-9
Z	1-9
N	2-9
[5-7]	5, 6 and 7
[15-7]	1,5, 6 and 7
.	Any character or digit.
exten => _XX,1,Answer()	A regular expression as to start with an underscore.

sip.conf

[general]	Section to configure global stuff.
port=5060	Asterisk listens to port 5060.
bindaddr=0.0.0.0	Asterisk listens to this IP only. 0.0.0.0 means to all IP addresses.
[200]	Configures the SIP channel 200 (can be alphanumeric).
type=friend	Configures the type of channel. Asterisk <= user Asterisk => peer Asterisk = friend
username=200	Username
secret=1234	Password
host=dynamic	Defines the IP address for this SIP device. dynamic = all IPs
context=from-test	Context which is started when this devices opens a channel.

extensions.conf

[general]	General configuration section.
static=yes	The configuration is static.
writeprotect=yes	It is not possible to change the dialplan from the CLI.
VAR2=23	Sets a global variable.

Applications

Dial(tech/u:p@host)	Connects to the given host/user. Can be supplemented with ,ring-timout (in seconds) and several flags (e.g. tT to permits the caller and the called party to transfer a call by pressing #).
Answer()	Answers the channel.
Hangup()	Quits the channel.
Wait(n)	Waits for n seconds.
Goto(n)	Jumps to the priority n for the same extension. Goto(100,12) jumps to the priority 12 of the extension 100. Goto(example,23,5) jumps to the priority 5 of the extension 23 in the example context.
SetGlobal(VAR1=1)	Sets the global variable var1.
Set(VAR3=Test)	Sets a channel variable.
GotoIf($[$(X}=1]?,1:5)	Jumps to priority 1 if the variable X has the value 1. If not it jumps to the priority 5.
SayNumber()	Says a number
SayAlpha()	Spells out the string.
SayDigits()	Says the single digits of a number.

Functions

LEN()	Calculates the length of a variable (e.g. LEN(${X})).
URIENCODE()	URIencodes a string.
URIDECODE()	URIdecodes a string.
CALLERID()	Sets and reads the callerid. exten => 1234,1, Set(CALLERID(name)="Smith")

VI editor

ESC	Changes from the edit modus to the command modus.
i	Changes into the edit modus (i = insert).
dd	In the command modus dd deletes the line in which the curser is located. 100dd deletes the next 100 lines.
:x	Saves a document in the command modus.
:q!	Quits vi without saving the file (call changes are lost). Command modus.

The following example configuration is for 2 SIP Phones which are connected in a mini PBX.

sip.conf

```
[general]
port=5060
bindaddr=0.0.0.0
context=misc

[10]
type=friend
context=internal
username=10
secret=1234
host=dynamic

[11]
type=friend
context=internal
username=11
secret=1234
host=dynamic
```

extensions.conf

```
[globals]

[internal]
exten => _1X,1,Dail(SIP/${EXTEN},90)
exten => _1X,2,VoiceMail(u${EXTEN})
exten => _1X,102,VoiceMail(b${EXTEN})

exten => 99,1,VoiceMailMain(${CALLERIDNUM})
```

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: http://www.amooma.de

Asterisk Reference Card

0.0.1, 2006-04-07

http://www.asterisk.org/

Getting Help

/usr/src/asterisk-1.2.x/doc/	Find all Asterisk documentation here.
asterisk.org	Reference, manuals, FAQs, HOWTOs, etc. at
Mailing lists at http://lists.digium.com/mailman/listinfo/	The community is always helpful, search for **users**.
voip-info.org Wiki	Reference, manuals, FAQs, HOWTOs, etc.

Installation

apt-get install gcc libc6 m4 openssl zlibc libkrb5-dev libncurses5 libncurses5-dev libssl-dev zlib1g-dev make wget	Get all packages which are necessary to compile Asterisk on a vanilla Debian.
cd /usr/src && wget http://ftp.digium.com/pub/asterisk/asterisk-1.2.6.tar.gz	Download Asterisk
cd /usr/src && tar xvzf asterisk-1.2.6.tar.gz && cd asterisk-1.2.6 && make && make install && make samples	Untar, compile and install Asterisk (including sample config files).