Asterisk Reference Information
Version 1.6.1.19-rc2

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April 13, 2010

Contents

Chapter 1

Introduction

This document contains various pieces of information that are useful for reference purposes.

1.1 License Information

Asterisk is distributed under the GNU General Public License version 2 and is also available under alternative licenses negotiated directly with Digium, Inc. If you obtained Asterisk under the GPL, then the GPL applies to all loadable Asterisk modules used on your system as well, except as defined below. The GPL (version 2) is included in this source tree in the file COPYING.

This package also includes various components that are not part of Asterisk itself; these components are in the 'contrib' directory and its subdirectories. These components are also distributed under the GPL version 2 as well.

Digium, Inc. (formerly Linux Support Services) holds copyright and/or sufficient licenses to all components of the Asterisk package, and therefore can grant, at its sole discretion, the ability for companies, individuals, or organizations to create proprietary or Open Source (even if not GPL) modules which may be dynamically linked at runtime with the portions of Asterisk which fall under our copyright/license umbrella, or are distributed under more flexible licenses than GPL.

If you wish to use our code in other GPL programs, don't worry – there is no requirement that you provide the same exception in your GPL'd prod-

ucts (although if you've written a module for Asterisk we would strongly encourage you to make the same exception that we do).

Specific permission is also granted to link Asterisk with OpenSSL, OpenH323 and/or the UW IMAP Toolkit and distribute the resulting binary files.

In addition, Asterisk implements two management/control protocols: the Asterisk Manager Interface (AMI) and the Asterisk Gateway Interface (AGI). It is our belief that applications using these protocols to manage or control an Asterisk instance do not have to be licensed under the GPL or a compatible license, as we believe these protocols do not create a 'derivative work' as referred to in the GPL. However, should any court or other judiciary body find that these protocols do fall under the terms of the GPL, then we hereby grant you a license to use these protocols in combination with Asterisk in external applications licensed under any license you wish.

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Digium, Inc. 445 Jan Davis Drive Huntsville, AL 35806 USA

1.1.1 Hold Music

Digium has licensed the music included with the Asterisk distribution From opsound.org for use and distribution with Asterisk. It is licensed ONLY for use as hold music within an Asterisk based PBX.

1.2 Security

1.2.1 Introduction

PLEASE READ THE FOLLOWING IMPORTANT SECURITY RELATED INFORMATION. IMPROPER CONFIGURATION OF ASTERISK COULD ALLOW UNAUTHORIZED USE OF YOUR FACILITIES, POTENTIALLY INCURRING SUBSTANTIAL CHARGES.

Asterisk security involves both network security (encryption, authentication) as well as dialplan security (authorization - who can access services in your pbx). If you are setting up Asterisk in production use, please make sure you understand the issues involved.

1.2.2 Network Security

If you install Asterisk and use the "make samples" command to install a demonstration configuration, Asterisk will open a few ports for accepting VoIP calls. Check the channel configuration files for the ports and IP addresses.

If you enable the manager interface in manager.conf, please make sure that you access manager in a safe environment or protect it with SSH or other VPN solutions.

For all TCP/IP connections in Asterisk, you can set ACL lists that will permit or deny network access to Asterisk services. Please check the "permit" and "deny" configuration options in manager.conf and the VoIP channel configurations - i.e. sip.conf and iax.conf.

The IAX2 protocol supports strong RSA key authentication as well as AES encryption of voice and signalling. The SIP channel does not support encryption in this version of Asterisk.

1.2.3 Dialplan Security

First and foremost remember this:

USE THE EXTENSION CONTEXTS TO ISOLATE OUTGOING OR TOLL SERVICES FROM ANY INCOMING CONNECTIONS.

You should consider that if any channel, incoming line, etc can enter an extension context that it has the capability of accessing any extension within that context.

Therefore, you should NOT allow access to outgoing or toll services in contexts that are accessible (especially without a password) from incoming channels, be they IAX channels, FX or other trunks, or even untrusted stations within you network. In particular, never ever put outgoing toll services in the "default" context. To make things easier, you can include the "default" context within other private contexts by using:

```
include => default
```

in the appropriate section. A well designed PBX might look like this:

```
[longdistance]
exten => _91NXXNXXXXXX,1,Dial(DAHDI/g2/${EXTEN:1})
include => local

[local]
exten => _9NXXNXXX,1,Dial(DAHDI/g2/${EXTEN:1})
include => default

[default]
exten => 6123,Dial(DAHDI/1)
```

DON'T FORGET TO TAKE THE DEMO CONTEXT OUT OF YOUR DEFAULT CONTEXT. There isn't really a security reason, it just will keep people from wanting to play with your Asterisk setup remotely.

1.2.4 Log Security

Please note that the Asterisk log files, as well as information printed to the Asterisk CLI, may contain sensitive information such as passwords and call history. Keep this in mind when providing access to these resources.

1.3 Hardware

1.3.1 Introduction

A PBX is only really useful if you can get calls into it. Of course, you can use Asterisk with VoIP calls (SIP, H.323, IAX, etc.), but you can also talk to the real PSTN through various cards.

Supported Hardware is divided into two general groups: DAHDI devices and non-DAHDI devices. The DAHDI compatible hardware supports pseudo-TDM conferencing and all call features through chan_dahdi, whereas non-DAHDI compatible hardware may have different features.

1.3.2 DAHDI compatible hardware

- Digium, Inc. (Primary Developer of Asterisk) `http://www.digium.com`
  - Analog Interfaces
    - TDM400P - The TDM400P is a half-length PCI 2.2-compliant card that supports FXS and FXO station interfaces for connecting analog telephones and analog POTS lines through a PC.
    - TDM800P - The TDM800P is a half-length PCI 2.2-compliant, 8 port card using Digium's VoiceBus technology that supports FXS and FXO station interfaces for connecting analog telephones and analog POTS lines through a PC.
    - TDM2400P - The TDM2400P is a full-length PCI 2.2-compliant card for connecting analog telephones and analog POTS lines through a PC. It supports a combination of up to 6 FXS and/or FXO modules for a total of 24 lines.
  - Digital Interfaces
    - TE412P - The TE412P offers an on-board DSP-based echo cancellation module. It supports E1, T1, and J1 environments and is selectable on a per-card or per-port basis.
    - TE410P - The TE410P improves performance and scalability through bus mastering architecture. It supports E1, T1, and J1 environments and is selectable on a per-card or per-port basis.
    - TE407P - The TE407P offers an on-board DSP-based echo cancellation module. It supports E1, T1, and J1 environments and is selectable on a per-card or per-port basis.
    - TE405P - The TE405P improves performance and scalability through bus mastering architecture. It supports both E1, T1, J1 environments and is selectable on a per-card or per-port basis.
    - TE212P - The TE212P offers an on-board DSP-based echo cancellation module. It supports E1, T1, and J1 environments and is selectable on a per-card or per-port basis.
    - TE210P - The TE210P improves performance and scalability through bus mastering architecture. It supports E1, T1, and

J1 environments and is selectable on a per-card or per-port basis.

* TE207P - The TE207P offers an on-board DSP-based echo cancellation module. It supports E1, T1, and J1 environments and is selectable on a per-card or per-port basis.
* TE205P - The TE205P improves performance and scalability through bus mastering architecture. It supports both E1 and T1/J1 environments and is selectable on a per-card or per-port basis.
* TE120P - The TE120P is a single span, selectable T1, E1, or J1 card and utilizes Digium's VoiceBus^{TM}technology. It supports both voice and data modes.
* TE110P - The TE110P brings a high-performance, cost-effective, and flexible single span togglable T1, E1, J1 interface to the Digium line-up of telephony interface devices.

1.3.3 Non-DAHDI compatible hardware

- QuickNet, Inc. `http://www.quicknet.net`
  - Internet PhoneJack - Single FXS interface. Supports Linux telephony interface. DSP compression built-in.
  - Internet LineJack - Single FXS or FXO interface. Supports Linux telephony interface.

1.3.4 mISDN compatible hardware

mISDN homepage: `http://www.misdn.org/`

Any adapter with an mISDN driver should be compatible with chan_misdn. See the mISDN section for more information.

- Digium, Inc. (Primary Developer of Asterisk) `http://www.digium.com`
  - B410P - 4 Port BRI card (TE/NT)
- beroNet `http://www.beronet.com`
  - BN4S0 - 4 Port BRI card (TE/NT)

  - BN8S0 - 8 Port BRI card (TE/NT)
  - Billion Card - Single Port BRI card (TE (/NT with crossed cable))

1.3.5 Miscellaneous other interfaces

- Digium, Inc. (Primary Developer of Asterisk)
  - TC400B - The TC400B is a half-length, low-profile PCI 2.2-compliant card for transforming complex VoIP codecs (G.729) into simple codecs.
- ALSA `http://www.alsa-project.org`
  - Any ALSA compatible full-duplex sound card
- OSS `http://www.opensound.com`
  - Any OSS compatible full-duplex sound card

Chapter 2

Configuration

2.1 General Configuration Information

2.1.1 Configuration Parser

Introduction

The Asterisk configuration parser in the 1.2 version and beyond series has been improved in a number of ways. In addition to the realtime architecture, we now have the ability to create templates in configuration files, and use these as templates when we configure phones, voicemail accounts and queues.

These changes are general to the configuration parser, and works in all configuration files.

General syntax

Asterisk configuration files are defined as follows:

```
[section]
label = value
label2 = value
```

In some files, (e.g. mgcp.conf, dahdi.conf and agents.conf), the syntax is a bit different. In these files the syntax is as follows:

```
[section]
label1 = value1
label2 = value2
object => name
```

```
label3 = value3
label2 = value4
object2 => name2
```

In this syntax, we create objects with the settings defined above the object creation. Note that settings are inherited from the top, so in the example above object2 has inherited the setting for "label1" from the first object.

For template configurations, the syntax for defining a section is changed to:

```
[section](options)
label = value
```

The options field is used to define templates, refer to templates and hide templates. Any object can be used as a template.

No whitespace is allowed between the closing "]" and the parenthesis "(".

Comments

All lines that starts with semi-colon ";" is treated as comments and is not parsed.

The ";--" is a marker for a multi-line comment. Everything after that marker will be treated as a comment until the end-marker "--;" is found. Parsing begins directly after the end-marker.

```
;This is a comment
label = value
;-- This is
a comment --;

;-- Comment --; exten=> 1000,1,dial(SIP/lisa)
```

Including other files

In all of the configuration files, you may include the content of another file with the #include statement. The content of the other file will be included at the row that the #include statement occurred.

```
#include myusers.conf
```

You may also include the output of a program with the #exec directive, if you enable it in asterisk.conf

In asterisk.conf, add the execincludes = yes statement in the options section:

```
[options]
execincludes=yes
```

The exec directive is used like this:

```
#exec /usr/local/bin/myasteriskconfigurator.sh
```

Adding to an existing section

```
[section]
label = value

[section](+)
label2 = value2
```

In this case, the plus sign indicates that the second section (with the same name) is an addition to the first section. The second section can be in another file (by using the #include statement). If the section name referred to before the plus is missing, the configuration will fail to load.

Defining a template-only section

```
[section](!)
label = value
```

The exclamation mark indicates to the config parser that this is a only a template and should not itself be used by the Asterisk module for configuration. The section can be inherited by other sections (see section "Using templates" below) but is not used by itself.

Using templates (or other configuration sections)

```
[section](name[,name])
label = value
```

The name within the parenthesis refers to other sections, either templates or standard sections. The referred sections are included before the configuration engine parses the local settings within the section as though their entire contents (and anything they were previously based upon) were included in the new section. For example consider the following:

```
[foo]
disallow=all
allow=ulaw
allow=alaw
```

```
[bar]
allow=gsm
allow=g729
permit=192.168.2.1

[baz](foo,bar)
type=friend
permit=192.168.3.1
context=incoming
host=bnm
```

The [baz] section will be processed as though it had been written in the following way:

```
[baz]
disallow=all
allow=ulaw
allow=alaw
allow=gsm
allow=g729
permit=192.168.2.1
type=friend
permit=192.168.3.1
context=incoming
host=bnm
```

It should also be noted that there are no guaranteed overriding semantics, meaning that if you define something in one template, you should not expect to be able to override it by defining it again in another template.

Additional Examples

(in top-level sip.conf)

```
[defaults](!)
type=friend
nat=yes
qualify=on
dtmfmode=rfc2833
disallow=all
allow=alaw

#include accounts/*/sip.conf
```

(in `accounts/customer1/sip.conf`)

```
[def-customer1](!,defaults)
secret=this_is_not_secret
context=from-customer1
callerid=Customer 1 <300>
accountcode=0001
```

```
[phone1](def-customer1)
mailbox=phone1@customer1

[phone2](def-customer1)
mailbox=phone2@customer1
```

This example defines two phones - phone1 and phone2 with settings inherited from "def-customer1". The "def-customer1" is a template that inherits from "defaults", which also is a template.

2.1.2 Asterisk.conf

Asterisk Main Configuration File

Below is a sample of the main Asterisk configuration file, asterisk.conf. Note that this file is not provided in sample form, because the Makefile creates it when needed and does not touch it when it already exists.

```
[directories]
; Make sure these directories have the right permissions if not
; running Asterisk as root

; Where the configuration files (except for this one) are located
astetcdir => /etc/asterisk

; Where the Asterisk loadable modules are located
astmoddir => /usr/lib/asterisk/modules

; Where additional 'library' elements (scripts, etc.) are located
astvarlibdir => /var/lib/asterisk

; Where AGI scripts/programs are located
astagidir => /var/lib/asterisk/agi-bin

; Where spool directories are located
; Voicemail, monitor, dictation and other apps will create files here
; and outgoing call files (used with pbx_spool) must be placed here
astspooldir => /var/spool/asterisk

; Where the Asterisk process ID (pid) file should be created
astrundir => /var/run/asterisk

; Where the Asterisk log files should be created
astlogdir => /var/log/asterisk


[options]
;Under "options" you can enter configuration options
;that you also can set with command line options

; Verbosity level for logging (-v)
```

```
verbose = 0

; Debug: "No" or value (1-4)
debug = 3

; Background execution disabled (-f)
nofork=yes | no

; Always background, even with -v or -d (-F)
alwaysfork=yes | no

; Console mode (-c)
console= yes | no

; Execute with high priority (-p)
highpriority = yes | no

; Initialize crypto at startup (-i)
initcrypto = yes | no

; Disable ANSI colors (-n)
nocolor = yes | no

; Dump core on failure (-g)
dumpcore = yes | no

; Run quietly (-q)
quiet = yes | no

; Force timestamping in CLI verbose output (-T)
timestamp = yes | no

; User to run asterisk as (-U) NOTE: will require changes to
; directory and device permissions
runuser = asterisk

; Group to run asterisk as (-G)
rungroup = asterisk

; Enable internal timing support (-I)
internal_timing = yes | no

; These options have no command line equivalent

; Cache record() files in another directory until completion
cache_record_files = yes | no
record_cache_dir = <dir>

; Build transcode paths via SLINEAR
transcode_via_sln = yes | no

; send SLINEAR silence while channel is being recorded
transmit_silence_during_record = yes | no

; The maximum load average we accept calls for
maxload = 1.0
```

```
; The maximum number of concurrent calls you want to allow
maxcalls = 255

; Stop accepting calls when free memory falls below this amount specified in MB
minmemfree = 256

; Allow #exec entries in configuration files
execincludes = yes | no

; Don't over-inform the Asterisk sysadm, he's a guru
dontwarn = yes | no

; System name. Used to prefix CDR uniqueid and to fill \${SYSTEMNAME}
systemname = <a_string>

; Should language code be last component of sound file name or first?
; when off, sound files are searched as <path>/<lang>/<file>
; when on, sound files are search as <lang>/<path>/<file>
; (only affects relative paths for sound files)
languageprefix = yes | no

; Locking mode for voicemail
;  - lockfile: default, for normal use
;  - flock: for where the lockfile locking method doesn't work
;           eh. on SMB/CIFS mounts
lockmode = lockfile | flock

; Entity ID.  This is in the form of a MAC address.  It should be universally
; unique.  It must be unique between servers communicating with a protocol
; that uses this value.  The only thing that uses this currently is DUNDi,
; but other things will use it in the future.
; entityid=00:11:22:33:44:55

[files]
; Changing the following lines may compromise your security
; Asterisk.ctl is the pipe that is used to connect the remote CLI
; (asterisk -r) to Asterisk. Changing these settings change the
; permissions and ownership of this file.
; The file is created when Asterisk starts, in the "astrundir" above.

;astctlpermissions = 0660
;astctlowner = root
;astctlgroup = asterisk
;astctl = asterisk.ctl
```

2.1.3 CLI Prompt

Changing the CLI Prompt

The CLI prompt is set with the ASTERISK_PROMPT UNIX environment variable that you set from the Unix shell before starting Asterisk

You may include the following variables, that will be replaced by the

current value by Asterisk:

- %d - Date (year-month-date)
- %s - Asterisk system name (from asterisk.conf)
- %h - Full hostname
- %H - Short hostname
- %t - Time
- %u - Username
- %g - Groupname
- %% - Percent sign
- %# - '#' if Asterisk is run in console mode, '>' if running as remote console
- %Cn[;n] - Change terminal foreground (and optional background) color to specified A full list of colors may be found in `include/asterisk/term.h`

On systems which implement getloadavg(3), you may also use:

- %l1 - Load average over past minute
- %l2 - Load average over past 5 minutes
- %l3 - Load average over past 15 minutes

2.1.4 Extensions

The Asterisk dialplan

The Asterisk dialplan is divided into contexts. A context is simply a group of extensions. For each "line" that should be able to be called, an extension must be added to a context. Then, you configure the calling "line" to have access to this context.

If you change the dialplan, you can use the Asterisk CLI command "dialplan reload" to load the new dialplan without disrupting service in your PBX.

Extensions are routed according to priority and may be based on any set of characters (a-z), digits, #, and *. Please note that when matching a pattern, "N", "X", and "Z" are interpreted as classes of digits.

For each extension, several actions may be listed and must be given a unique priority. When each action completes, the call continues at the next priority (except for some modules which use explicitly GOTO's).

Extensions frequently have data they pass to the executing application (most frequently a string). You can see the available dialplan applications by entering the "core show applications" command in the CLI.

In this version of Asterisk, dialplan functions are added. These can be used as arguments to any application. For a list of the installed functions in your Asterisk, use the "core show functions" command.

Example dialplan

The example dial plan, in the `configs/extensions.conf.sample` file is installed as extensions.conf if you run "make samples" after installation of Asterisk. This file includes many more instructions and examples than this file, so it's worthwhile to read it.

Special extensions

There are some extensions with important meanings:

- s
  - What to do when an extension context is entered (unless overridden by the low level channel interface) This is used in macros, and some special cases. "s" is not a generic catch-all wildcard extension.
- i
  - What to do if an invalid extension is entered
- h

  - The hangup extension, executed at hangup

- t
  - What to do if nothing is entered in the requisite amount of time.
- T
  - This is the extension that is executed when the 'absolute' timeout is reached. See "core show function TIMEOUT" for more information on setting timeouts.
- e
  - This extension will substitute as a catchall for any of the 'i', 't', or 'T' extensions, if any of them do not exist and catching the error in a single routine is desired. The function EXCEPTION may be used to query the type of exception or the location where it occurred.

And finally, the extension context "default" is used when either a) an extension context is deleted while an extension is in use, or b) a specific starting extension handler has not been defined (unless overridden by the low level channel interface).

2.1.5 IP Quality of Service

Introduction

Asterisk support different QoS settings on application level on various protocol on any of signaling and media. Type of Service (TOS) byte can be set on outgoing IP packets for various protocols. The TOS byte is used by the network to provide some level of Quality of Service (QoS) even if the network is congested with other traffic.

Also asterisk running on Linux can set 802.1p CoS marks in VLAN packets for all used VoIP protocols. It is useful when you are working in switched environment. In fact asterisk only set priority for Linux socket. For mapping this priority and VLAN CoS mark you need to use this command:

```
vconfig set_egress_map [vlan-device] [skb-priority] [vlan-qos]
```

In table behind shown all voice channels and other modules of asterisk, that support QoS settings for network traffic and type of traffic which can have QoS settings.

Channel Drivers

	Signaling	Audio	Video	Text
chan_sip	+	+	+	+
chan_skinny	+	+	+	
chan_mgcp	+	+		
chan_unistim	+	+		
chan_h323		+		
chan_iax2	+			

Other

dundi.conf	+ (tos setting)
iaxprov.conf	+ (tos setting)

IP TOS values

The allowable values for any of the tos* parameters are: CS0, CS1, CS2, CS3, CS4, CS5, CS6, CS7, AF11, AF12, AF13, AF21, AF22, AF23, AF31, AF32, AF33, AF41, AF42, AF43 and ef (expedited forwarding),

The tos* parameters also take numeric values.

Note, that on Linux system you can use ef value in case your asterisk is running from a user other then root only when you have compiled asterisk with libcap.

The lowdelay, throughput, reliability, mincost, and none values are removed in current releases.

802.1p CoS values

As far as 802.1p uses 3 bites from VLAN header, there are parameter can take integer values from 0 to 7.

Recommended values

Recommended values shown above and also included in sample configuration files:

	tos	cos
Signaling	cs3	3
Audio	ef	5
Video	af41	4
Text	af41	3
Other	ef	

IAX2

In iax.conf, there is a "tos" parameter that sets the global default TOS for IAX packets generated by chan_iax2. Since IAX connections combine signalling, audio, and video into one UDP stream, it is not possible to set the TOS separately for the different types of traffic.

In iaxprov.conf, there is a "tos" parameter that tells the IAXy what TOS to set on packets it generates. As with the parameter in iax.conf, IAX packets generated by an IAXy cannot have different TOS settings based upon the type of packet. However different IAXy devices can have different TOS settings.

SIP

In sip.conf, there are three parameters that control the TOS settings: "tos_sip", "tos_audio", "tos_video" and "tos_text". tos_sip controls what TOS SIP call signaling packets are set to. tos_audio, tos_video and tos_text controls what TOS RTP audio, video or text accordingly packets are set to.

There are four parameters to control 802.1p CoS: "cos_sip", "cos_audio", "cos_video" and "cos_text". It behavior the same as written above.

Other RTP channels

chan_mgcp, chan_h323, chan_skinny and chan_unistim also support TOS and CoS via setting tos and cos parameters in correspond to module config files. Naming style and behavior same as for chan_sip.

Reference

IEEE 802.1Q Standard: `http://standards.ieee.org/getieee802/download/802.1Q-1998.pdf` Related protocols: IEEE 802.3, 802.2, 802.1D, 802.1Q

RFC 2474 - "Definition of the Differentiated Services Field (DS field) in the IPv4 and IPv6 Headers", Nichols, K., et al, December 1998.

IANA Assignments, DSCP registry Differentiated Services Field Codepoints `http://www.iana.org/assignments/dscp-registry`

To get the most out of setting the TOS on packets generated by Asterisk, you will need to ensure that your network handles packets with a TOS properly. For Cisco devices, see the previously mentioned "Enterprise QoS Solution Reference Network Design Guide". For Linux systems see the "Linux Advanced Routing & Traffic Control HOWTO" at `http://www.lartc.org/`.

For more information on Quality of Service for VoIP networks see the "Enterprise QoS Solution Reference Network Design Guide" version 3.3 from Cisco at: `http://www.cisco.com/application/pdf/en/us/guest/netsol/ns432/c649/ccmigration_09186a008049b062.pdf`

2.1.6 MP3 Support

MP3 Music On Hold

Use of the mpg123 for your music on hold is no longer recommended and is now officially deprecated. You should now use one of the native formats for your music on hold selections.

However, if you still need to use mp3 as your music on hold format, a format driver for reading MP3 audio files is available in the asterisk-addons SVN repository on svn.digium.com or in the asterisk-addons release at `http://downloads.digium.com/pub/telephony/asterisk/`.

2.1.7 ICES

The advent of icecast into Asterisk allows you to do neat things like have a caller stream right into an ice-cast stream as well as using chan_local to place things like conferences, music on hold, etc. into the stream.

You'll need to specify a config file for the ices encoder. An example is included in `contrib/asterisk-ices.xml`.

2.2 Database Support

2.2.1 Realtime Database Configuration

Introduction

The Asterisk Realtime Architecture is a new set of drivers and functions implemented in Asterisk.

The benefits of this architecture are many, both from a code management standpoint and from an installation perspective.

The ARA is designed to be independent of storage. Currently, most drivers are based on SQL, but the architecture should be able to handle other storage methods in the future, like LDAP.

The main benefit comes in the database support. In Asterisk v1.0 some functions supported MySQL database, some PostgreSQL and other ODBC. With the ARA, we have a unified database interface internally in Asterisk, so if one function supports database integration, all databases that has a realtime driver will be supported in that function.

Currently there are three realtime database drivers:

- ODBC: Support for UnixODBC, integrated into Asterisk The UnixODBC subsystem supports many different databases, please check `www.unixodbc.org` for more information.
- MySQL: Found in the asterisk-addons subversion repository on `svn.digium.com`
- PostgreSQL: Native support for Postgres, integrated into Asterisk

Two modes: Static and Realtime

The ARA realtime mode is used to dynamically load and update objects. This mode is used in the SIP and IAX2 channels, as well as in the voicemail system. For SIP and IAX2 this is similar to the v1.0 MYSQL_FRIENDS functionality. With the ARA, we now support many more databases for dynamic configuration of phones.

The ARA static mode is used to load configuration files. For the Asterisk modules that read configurations, there's no difference between a static file in the file system, like extensions.conf, and a configuration loaded from a database.

You just have to always make sure the var_metric values are properly set and ordered as you expect in your database server if you're using the static mode with ARA (either sequentially or with the same var_metric value for everybody).

If you have an option that depends on another one in a given configuration file (i.e, 'musiconhold' depending on 'agent' from agents.conf) but their var_metric are not sequential you'll probably get default values being assigned for those options instead of the desired ones. You can still use the same var_metric for all entries in your DB, just make sure the entries are recorded in an order that does not break the option dependency.

That doesn't happen when you use a static file in the file system. Although this might be interpreted as a bug or limitation, it is not.

Realtime SIP friends

The SIP realtime objects are users and peers that are loaded in memory when needed, then deleted. This means that Asterisk currently can't handle voicemail notification and NAT keepalives for these peers. Other than that, most of the functionality works the same way for realtime friends as for the ones in static configuration.

With caching, the device stays in memory for a specified time. More information about this is to be found in the sip.conf sample file.

If you specify a separate family called "sipregs" SIP registration data will be stored in that table and not in the "sippeers" table.

Realtime H.323 friends

Like SIP realtime friends, H.323 friends also can be configured using dynamic realtime objects.

New function in the dial plan: The Realtime Switch

The realtime switch is more than a port of functionality in v1.0 to the new architecture, this is a new feature of Asterisk based on the ARA. The realtime switch lets your Asterisk server do database lookups of extensions in realtime from your dial plan. You can have many Asterisk servers sharing a dynamically updated dial plan in real time with this solution.

Note that this switch does NOT support Caller ID matching, only extension name or pattern matching.

Capabilities

The realtime Architecture lets you store all of your configuration in databases and reload it whenever you want. You can force a reload over the AMI, Asterisk Manager Interface or by calling Asterisk from a shell script with

asterisk -rx "reload"

You may also dynamically add SIP and IAX devices and extensions and making them available without a reload, by using the realtime objects and the realtime switch.

Configuration in extconfig.conf

You configure the ARA in extconfig.conf (yes, it's a strange name, but is was defined in the early days of the realtime architecture and kind of stuck).

The part of Asterisk that connects to the ARA use a well defined family name to find the proper database driver. The syntax is easy:

```
<family> => <realtime driver>,<db name>[,<table>]
```

The options following the realtime driver identified depends on the driver. Defined well-known family names are:

- sippeers, sipusers - SIP peers and users
- iaxpeers, iaxusers - IAX2 peers and users

- voicemail - Voicemail accounts
- queues - Queues
- queue_members - Queue members
- extensions - Realtime extensions (switch)

Voicemail storage with the support of ODBC described in file `docs/odbcstorage.tex` (11.1).

Limitations

Currently, realtime extensions do not support realtime hints. There is a workaround available by using func_odbc. See the sample func_odbc.conf for more information.

FreeTDS supported with connection pooling

In order to use a FreeTDS-based database with realtime, you need to turn connection pooling on in res_odbc.conf. This is due to a limitation within the FreeTDS protocol itself. Please note that this includes databases such as MS SQL Server and Sybase. This support is new in the current release.

2.2.2 FreeTDS

The cdr_tds module now works with most modern release versions of FreeTDS (from at least 0.60 through 0.82). Although versions of FreeTDS prior to 0.82 will work, we recommend using the latest available version for performance and stability reasons.

The latest release of FreeTDS is available from http://www.freetds.org/

2.3 Privacy

So, you want to avoid talking to pesky telemarketers/charity seekers/poll takers/magazine renewers/etc?

2.3.1 First of all

the FTC "Don't call" database, this alone will reduce your telemarketing call volume considerably. (see: `https://www.donotcall.gov/default.aspx`) But, this list won't protect from the Charities, previous business relationships, etc.

2.3.2 Next, Fight against autodialers!!

Zapateller detects if callerid is present, and if not, plays the da-da-da tones that immediately precede messages like, "I'm sorry, the number you have called is no longer in service."

Most humans, even those with unlisted/callerid-blocked numbers, will not immediately slam the handset down on the hook the moment they hear the three tones. But autodialers seem pretty quick to do this.

I just counted 40 hangups in Zapateller over the last year in my CDR's. So, that is possibly 40 different telemarketers/charities that have hopefully slashed my back-waters, out-of-the-way, humble home phone number from their lists.

I highly advise Zapateller for those seeking the nirvana of "privacy".

2.3.3 Next, Fight against the empty CALLERID!

A considerable percentage of the calls you don't want, come from sites that do not provide CallerID.

Null callerid's are a fact of life, and could be a friend with an unlisted number, or some charity looking for a handout. The PrivacyManager application can help here. It will ask the caller to enter a 10-digit phone number. They get 3 tries(configurable), and this is configurable, with control being passed to next priority where you can check the channelvariable PRIVACYMGRSTATUS. If the callerid was valid this variable will have the value SUCCESS, otherwise it will have the value FAILED.

PrivacyManager can't guarantee that the number they supply is any good, tho, as there is no way to find out, short of hanging up and calling them back. But some answers are obviously wrong. For instance, it seems a common practice for telemarketers to use your own number instead of giving you theirs. A simple test can detect this. More advanced tests would be to look for -555- numbers, numbers that count up or down, numbers of all the same digit, etc.

PrivacyManager can be told about a context where you can have patterns that describe valid phone numbers. If none of the patterns match the input, it will be considered a non-valid phonenumber and the user can try again until the retry counter is reached. This helps in resolving the issues stated in the previous paragraph.

My logs show that 39 have hung up in the PrivacyManager script over the last year.

(Note: Demanding all unlisted incoming callers to enter their CID may not always be appropriate for all users. Another option might be to use call screening. See below.)

2.3.4 Next, use a WELCOME MENU !

Experience has shown that simply presenting incoming callers with a set of options, no matter how simple, will deter them from calling you. In the vast majority of situations, a telemarketer will simply hang up rather than make a choice and press a key.

This will also immediately foil all autodialers that simply belch a message in your ear and hang up.

Example usage of Zapateller and PrivacyManager

```
[homeline]
exten => s,1,Answer
exten => s,2,SetVar,repeatcount=0
exten => s,3,Zapateller,nocallerid
exten => s,4,PrivacyManager
     ;; do this if they don't enter a number to Privacy Manager
exten => s,5,GotoIf($[ "${PRIVACYMGRSTATUS}" = "FAILED" ]?s,105)
exten => s,6,GotoIf($[ "${CALLERID(num)}"  = "7773334444" & "${CALLERID(name)}" : "Privacy Manager" ]?callerid-liar,
exten => s,7,Dial(SIP/yourphone)
exten => s,105,Background(tt-allbusy)
exten => s,106,Background(tt-somethingwrong)
exten => s,107,Background(tt-monkeysintro)
exten => s,108,Background(tt-monkeys)
exten => s,109,Background(tt-weasels)
exten => s,110,Hangup
```

I suggest using Zapateller at the beginning of the context, before anything else, on incoming calls.This can be followed by the PrivacyManager App.

Make sure, if you do the PrivacyManager app, that you take care of the error condition! or their non-compliance will be rewarded with access to the system. In the above, if they can't enter a 10-digit number in 3 tries, they get

the humorous "I'm sorry, but all household members are currently helping other telemarketers...", "something is terribly wrong", "monkeys have carried them away...", various loud monkey screechings, "weasels have...", and a hangup. There are plenty of other paths to my torture scripts, I wanted to have some fun.

In nearly all cases now, the telemarketers/charity-seekers that usually get thru to my main intro, hang up. I guess they can see it's pointless, or the average telemarketer/charity-seeker is instructed not to enter options when encountering such systems. Don't know.

2.3.5 Next: Torture Them!

I have developed an elaborate script to torture Telemarketers, and entertain friends. (See `http://www.voip-info.org/wiki-Asterisk+Telemarketer+Torture`)

While mostly those that call in and traverse my teletorture scripts are those we know, and are doing so out of curiosity, there have been these others from Jan 1st,2004 thru June 1st, 2004: (the numbers may or may not be correct.)

- 603890zzzz – hung up telemarket options.
- "Integrated Sale" – called a couple times. hung up in telemarket options
- "UNITED STATES GOV" – maybe a military recruiter, trying to lure one of my sons.
- 800349zzzz – hung up in charity intro
- 800349zzzz – hung up in charity choices, intro, about the only one who actually travelled to the bitter bottom of the scripts!
- 216377zzzz – hung up the magazine section
- 626757zzzz = "LIR " (pronounced "Liar"?) hung up in telemarket intro, then choices
- 757821zzzz – hung up in new magazine subscription options.

That averages out to maybe 1 a month. That puts into question whether the ratio of the amount of labor it took to make the scripts versus the benefits of lower call volumes was worth it, but, well, I had fun, so what the heck.

but, that's about it. Not a whole lot. But I haven't had to say "NO" or "GO AWAY" to any of these folks for about a year now ...!

2.3.6 Using Call Screening

Another option is to use call screening in the Dial command. It has two main privacy modes, one that remembers the CID of the caller, and how the callee wants the call handled, and the other, which does not have a "memory".

Turning on these modes in the dial command results in this sequence of events, when someone calls you at an extension:

1. The caller calls the Asterisk system, and at some point, selects an option or enters an extension number that would dial your extension.

2. Before ringing your extension, the caller is asked to supply an introduction. The application asks them: "After the tone, say your name". They are allowed 4 seconds of introduction.

3. After that, they are told "Hang on, we will attempt to connect you to your party. Depending on your dial options, they will hear ringing indications, or get music on hold. I suggest music on hold.

4. Your extension is then dialed. When (and if) you pick up, you are told that a caller presenting themselves as <their recorded intro is played> is calling, and you have options, like being connected, sending them to voicemail, torture, etc.

5. You make your selection, and the call is handled as you chose.

There are some variations, and these will be explained in due course.

To use these options, set your Dial to something like:

```
exten => 3,3,Dial(DAHDI/5r3&DAHDI/6r3,35,tmPA(beep))
    or
exten => 3,3,Dial(DAHDI/5r3&DAHDI/6r3,35,tmP(something)A(beep))
    or
exten => 3,3,Dial(DAHDI/5r3&DAHDI/6r3,35,tmpA(beep))
```

The 't' allows the dialed party to transfer the call using '#'. It's optional.

The 'm' is for music on hold. I suggest it. Otherwise, the calling party gets to hear all the ringing, and lack thereof. It is generally better to use Music On Hold. Lots of folks hang up after the 3rd or 4th ring, and you might lose the call before you can enter an option!

The 'P' option alone will database everything using the extension as a default 'tree'. To get multiple extensions sharing the same database, use P(some-shared-key). Also, if the same person has multiple extensions, use P(unique-id) on all their dial commands.

Use little 'p' for screening. Every incoming call will include a prompt for the callee's choice.

the A(beep), will generate a 'beep' that the callee will hear if they choose to talk to the caller. It's kind of a prompt to let the callee know that he has to say 'hi'. It's not required, but I find it helpful.

When there is no CallerID, P and p options will always record an intro for the incoming caller. This intro will be stored temporarily in the `/var/lib/asterisk/sounds/priv-callerintros` dir, under the name NOCALLERID_<extension> <channelname> and will be erased after the callee decides what to do with the call.

Of course, NOCALLERID is not stored in the database. All those with no CALLERID will be considered "Unknown".

2.3.7 The 'N' and 'n' options

Two other options exist, that act as modifiers to the privacy options 'P' and 'p'. They are 'N' and 'n'. You can enter them as dialing options, but they only affect things if P or p are also in the options.

'N' says, "Only screen the call if no CallerID is present". So, if a callerID were supplied, it will come straight thru to your extension.

'n' says, "Don't save any introductions". Folks will be asked to supply an introduction ("At the tone, say your name") every time they call. Their introductions will be removed after the callee makes a choice on how to handle the call. Whether the P option or the p option is used, the incoming caller will have to supply their intro every time they call.

2.3.8 Recorded Introductions

Philosophical Side Note

The 'P' option stores the CALLERID in the database, along with the callee's choice of actions, as a convenience to the CALLEE, whereas introductions are stored and re-used for the convenience of the CALLER.

Introductions

Unless instructed to not save introductions (see the 'n' option above), the screening modes will save the recordings of the caller's names in the directory `/var/lib/asterisk/sounds/priv-callerintros`, if they have a CallerID. Just the 10-digit callerid numbers are used as filenames, with a ".gsm" at the end.

Having these recordings around can be very useful, however...

First of all, if a callerid is supplied, and a recorded intro for that number is already present, the caller is spared the inconvenience of having to supply their name, which shortens their call a bit.

Next of all, these intros can be used in voicemail, played over loudspeakers, and perhaps other nifty things. For instance:

```
exten => s,6,Set(PATH=/var/lib/asterisk/sounds/priv-callerintros)
exten => s,7,System(/usr/bin/play ${PATH}/${CALLERID(num)}.gsm&,0)
```

When a call comes in at the house, the above priority gets executed, and the callers intro is played over the phone systems speakers. This gives us a hint who is calling.

(Note: the ,0 option at the end of the System command above, is a local mod I made to the System command. It forces a 0 result code to be returned, whether the play command successfully completed or not. Therefore, I don't have to ensure that the file exists or not. While I've turned this mod into the developers, it hasn't been incorporated yet. You might want to write an AGI or shell script to handle it a little more intelligently)

And one other thing. You can easily supply your callers with an option to listen to, and re-record their introductions. Here's what I did in the home system's extensions.conf. (assume that a Goto(home-introduction,s,1) exists somewhere in your main menu as an option):

```
[home-introduction]
exten => s,1,Background(intro-options) ;; Script:
```

```
     ;; To hear your Introduction, dial 1.
     ;;    to record a new introduction, dial 2.
     ;;    to return to the main menu, dial 3.
     ;;    to hear what this is all about, dial 4.
exten => 1,1,Playback,priv-callerintros/${CALLERID(num)}
exten => 1,2,Goto(s,1)
exten => 2,1,Goto(home-introduction-record,s,1)
exten => 3,1,Goto(homeline,s,7)
exten => 4,1,Playback(intro-intro)
     ;; Script:
     ;; This may seem a little strange, but it really is a neat
     ;; thing, both for you and for us. I've taped a short introduction
     ;; for many of the folks who normally call us. Using the Caller ID
     ;; from each incoming call, the system plays the introduction
     ;; for that phone number over a speaker, just as the call comes in.
     ;; This helps the folks
     ;; here in the house more quickly determine who is calling.
     ;; and gets the right ones to gravitate to the phone.
     ;; You can listen to, and record a new intro for your phone number
     ;; using this menu.
exten => 4,2,Goto(s,1)
exten => t,1,Goto(s,1)
exten => i,1,Background(invalid)
exten => i,2,Goto(s,1)
exten => o,1,Goto(s,1)

[home-introduction-record]
exten => s,1,Background(intro-record-choices)    ;; Script:
     ;; If you want some advice about recording your
     ;; introduction, dial 1.
     ;; otherwise, dial 2, and introduce yourself after
     ;; the beep.
exten => 1,1,Playback(intro-record)
     ;; Your introduction should be short and sweet and crisp.
     ;; Your introduction will be limited to 4 seconds.
     ;; This is NOT meant to be a voice mail message, so
     ;; please, don't say anything about why you are calling.
     ;; After we are done making the recording, your introduction
     ;; will be saved for playback.
     ;; If you are the only person that would call from this number,
     ;; please state your name.  Otherwise, state your business
     ;; or residence name instead. For instance, if you are
     ;; friend of the family, say, Olie McPherson, and both
     ;; you and your kids might call here a lot, you might
     ;; say: "This is the distinguished Olie McPherson Residence!"
     ;; If you are the only person calling, you might say this:
     ;; "This is the illustrious Kermit McFrog! Pick up the Phone, someone!!"
     ;; If you are calling from a business, you might pronounce a more sedate introduction,like,
     ;; "Fritz from McDonalds calling.", or perhaps the more original introduction:
     ;; "John, from the Park County Morgue. You stab 'em, we slab 'em!".
     ;; Just one caution: the kids will hear what you record every time
     ;; you call. So watch your language!
     ;; I will begin recording after the tone.
     ;; When you are done, hit the # key. Gather your thoughts and get
     ;; ready. Remember, the # key will end the recording, and play back
     ;; your intro. Good Luck, and Thank you!"
exten => 1,2,Goto(2,1)
```

```
exten => 2,1,Background(intro-start)
     ;;  OK, here we go! After the beep, please give your introduction.
exten => 2,2,Background(beep)
exten => 2,3,Record(priv-callerintros/${CALLERID(num)}:gsm,4)
exten => 2,4,Background(priv-callerintros/${CALLERID(num)})
exten => 2,5,Goto(home-introduction,s,1)
exten => t,1,Goto(s,1)
exten => i,1,Background(invalid)
exten => i,2,Goto(s,1)
exten => o,1,Goto(s,1)
```

In the above, you'd most likely reword the messages to your liking, and maybe do more advanced things with the 'error' conditions (i,o,t priorities), but I hope it conveys the idea.

Chapter 3

Channel Variables

3.1 Introduction

There are two levels of parameter evaluation done in the Asterisk dial plan in extensions.conf.

1. The first, and most frequently used, is the substitution of variable references with their values.

2. Then there are the evaluations of expressions done in $[..]. This will be discussed below.

Asterisk has user-defined variables and standard variables set by various modules in Asterisk. These standard variables are listed at the end of this document.

3.2 Parameter Quoting

```
exten => s,5,BackGround,blabla
```

The parameter (blabla) can be quoted ("blabla"). In this case, a comma does not terminate the field. However, the double quotes will be passed down to the Background command, in this example.

Also, characters special to variable substitution, expression evaluation, etc (see below), can be quoted. For example, to literally use a $ on the string

"$1231", quote it with a preceding \. Special characters that must be quoted to be used, are [ ] $ " \. (to write \itself, use \).

These Double quotes and escapes are evaluated at the level of the asterisk config file parser.

Double quotes can also be used inside expressions, as discussed below.

3.3 Variables

Parameter strings can include variables. Variable names are arbitrary strings. They are stored in the respective channel structure.

To set a variable to a particular value, do:

```
exten => 1,2,Set(varname=value)
```

You can substitute the value of a variable everywhere using ${variablename}. For example, to stringwise append $lala to $blabla and store result in $koko, do:

```
exten => 1,2,Set(koko=${blabla}${lala})
```

There are two reference modes - reference by value and reference by name. To refer to a variable with its name (as an argument to a function that requires a variable), just write the name. To refer to the variable's value, enclose it inside ${}. For example, Set takes as the first argument (before the =) a variable name, so:

```
exten => 1,2,Set(koko=lala)
exten => 1,3,Set(${koko}=blabla)
```

stores to the variable "koko" the value "lala" and to variable "lala" the value "blabla".

In fact, everything contained ${here} is just replaced with the value of the variable "here".

3.4 Variable Inheritance

Variable names which are prefixed by "_" will be inherited to channels that are created in the process of servicing the original channel in which the variable was set. When the inheritance takes place, the prefix will be removed in the channel inheriting the variable. If the name is prefixed by "__" in the

channel, then the variable is inherited and the "__" will remain intact in the new channel.

In the dialplan, all references to these variables refer to the same variable, regardless of having a prefix or not. Note that setting any version of the variable removes any other version of the variable, regardless of prefix.

3.4.1 Example

```
Set(__FOO=bar) ; Sets an inherited version of "FOO" variable
Set(FOO=bar)   ; Removes the inherited version and sets a local
               ; variable.
```

However, NoOp(${__FOO}) is identical to NoOp(${FOO})

3.5 Selecting Characters from Variables

The format for selecting characters from a variable can be expressed as:

```
${variable_name[:offset[:length]]}
```

If you want to select the first N characters from the string assigned to a variable, simply append a colon and the number of characters to skip from the beginning of the string to the variable name.

```
; Remove the first character of extension, save in "number" variable
exten => _9X.,1,Set(number=${EXTEN:1})
```

Assuming we've dialed 918005551234, the value saved to the 'number' variable would be 18005551234. This is useful in situations when we require users to dial a number to access an outside line, but do not wish to pass the first digit.

If you use a negative offset number, Asterisk starts counting from the end of the string and then selects everything after the new position. The following example will save the numbers 1234 to the 'number' variable, still assuming we've dialed 918005551234.

```
; Remove everything before the last four digits of the dialed string
exten => _9X.,1,Set(number=${EXTEN:-4})
```

We can also limit the number of characters from our offset position that we wish to use. This is done by appending a second colon and length value to the variable name. The following example will save the numbers 555 to the 'number' variable.

```
; Only save the middle numbers 555 from the string 918005551234
exten => _9X.,1,Set(number=${EXTEN:5:3})
```

The length value can also be used in conjunction with a negative offset. This may be useful if the length of the string is unknown, but the trailing digits are. The following example will save the numbers 555 to the 'number' variable, even if the string starts with more characters than expected (unlike the previous example).

```
; Save the numbers 555 to the 'number' variable
exten => _9X.,1,Set(number=${EXTEN:-7:3})
```

If a negative length value is entered, Asterisk will remove that many characters from the end of the string.

```
; Set pin to everything but the trailing #.
exten => _XXXX#,1,Set(pin=${EXTEN:0:-1})
```

3.6 Expressions

Everything contained inside a bracket pair prefixed by a $ (like $[this]) is considered as an expression and it is evaluated. Evaluation works similar to (but is done on a later stage than) variable substitution: the expression (including the square brackets) is replaced by the result of the expression evaluation.

For example, after the sequence:

```
exten => 1,1,Set(lala=$[1 + 2])
exten => 1,2,Set(koko=$[2 * ${lala}])
```

the value of variable koko is "6".

and, further:

```
exten => 1,1,Set,(lala=$[  1 +    2   ]);
```

will parse as intended. Extra spaces are ignored.

3.6.1 Spaces Inside Variables Values

If the variable being evaluated contains spaces, there can be problems.

For these cases, double quotes around text that may contain spaces will force the surrounded text to be evaluated as a single token. The double quotes will be counted as part of that lexical token.

As an example:

```
exten => s,6,GotoIf($[ "${CALLERID(name)}" : "Privacy Manager" ]?callerid-liar,s,1:s,7)
```

The variable CALLERID(name) could evaluate to "DELOREAN MOTORS" (with a space) but the above will evaluate to:

```
"DELOREAN MOTORS" : "Privacy Manager"
```

and will evaluate to 0.

The above without double quotes would have evaluated to:

```
DELOREAN MOTORS : Privacy Manager
```

and will result in syntax errors, because token DELOREAN is immediately followed by token MOTORS and the expression parser will not know how to evaluate this expression, because it does not match its grammar.

3.6.2 Operators

Operators are listed below in order of increasing precedence. Operators with equal precedence are grouped within { } symbols.

- `expr1 | expr2`

  Return the evaluation of expr1 if it is neither an empty string nor zero; otherwise, returns the evaluation of expr2.

- `expr1 & expr2`

  Return the evaluation of expr1 if neither expression evaluates to an empty string or zero; otherwise, returns zero.

- `expr1 {=, >, >=, <, <=, !=} expr2`

  Return the results of floating point comparison if both arguments are numbers; otherwise, returns the results of string comparison using the locale-specific collation sequence. The result of each comparison is 1 if the specified relation is true, or 0 if the relation is false.

- `expr1 {+, -} expr2`

  Return the results of addition or subtraction of floating point-valued arguments.

- `expr1 {*, /, %} expr2`

  Return the results of multiplication, floating point division, or remainder of arguments.

- `- expr1`

  Return the result of subtracting expr1 from 0. This, the unary minus operator, is right associative, and has the same precedence as the ! operator.

- `! expr1`

  Return the result of a logical complement of expr1. In other words, if expr1 is null, 0, an empty string, or the string "0", return a 1. Otherwise, return a 0. It has the same precedence as the unary minus operator, and is also right associative.

- `expr1 : expr2`

  The ':' operator matches expr1 against expr2, which must be a regular expression. The regular expression is anchored to the beginning of the string with an implicit '^'.

  If the match succeeds and the pattern contains at least one regular expression subexpression '...', the string corresponing to '\1' is returned; otherwise the matching operator returns the number of characters matched. If the match fails and the pattern contains a regular expression subexpression the null string is returned; otherwise 0.

  Normally, the double quotes wrapping a string are left as part of the string. This is disastrous to the : operator. Therefore, before the regex match is made, beginning and ending double quote characters are stripped from both the pattern and the string.

- `expr1 =~ expr2`

  Exactly the same as the ':' operator, except that the match is not anchored to the beginning of the string. Pardon any similarity to seemingly similar operators in other programming languages! The ":" and "=~" operators share the same precedence.

- `expr1 ? expr2 :: expr3`

Traditional Conditional operator. If expr1 is a number that evaluates to 0 (false), expr3 is result of the this expression evaluation. Otherwise, expr2 is the result. If expr1 is a string, and evaluates to an empty string, or the two characters (""), then expr3 is the result. Otherwise, expr2 is the result. In Asterisk, all 3 exprs will be "evaluated"; if expr1 is "true", expr2 will be the result of the "evaluation" of this expression. expr3 will be the result otherwise. This operator has the lowest precedence.

- `expr1 ~~ expr2`

  Concatenation operator. The two exprs are evaluated and turned into strings, stripped of surrounding double quotes, and are turned into a single string with no invtervening spaces. This operator is new to trunk after 1.6.0; it is not needed in existing extensions.conf code. Because of the way asterisk evaluates *and*[ ] constructs (recursively, bottom- up), no []*or* is ever present when the contents of a *or*[] is evaluated. Thus, tokens are usually already merged at evaluation time. But, in AEL, various exprs are evaluated raw, and *and*[] are gathered and treated as tokens. And in AEL, no two tokens can sit side by side without an intervening operator. So, in AEL, concatenation must be explicitly specified in expressions. This new operator will play well into future plans, where expressions ($[]constructs, andvariablereferences($ constructs) are merged into a single grammar.

Parentheses are used for grouping in the usual manner.

Operator precedence is applied as one would expect in any of the C or C derived languages.

3.6.3 Floating Point Numbers

In 1.6 and above, we shifted the $[...] expressions to be calculated via floating point numbers instead of integers. We use 'long double' numbers when possible, which provide around 16 digits of precision with 12 byte numbers.

To specify a floating point constant, the number has to have this format: D.D, where D is a string of base 10 digits. So, you can say 0.10, but you can't say .10 or 20.– we hope this is not an excessive restriction!

Floating point numbers are turned into strings via the '%g'/'%Lg' format of the printf function set. This allows numbers to still 'look' like integers to those counting on integer behavior. If you were counting on 1/4 evaluating to

0, you need to now say TRUNC(1/4). For a list of all the truncation/rounding capabilities, see the next section.

3.6.4 Functions

In 1.6 and above, we upgraded the $[] expressions to handle floating point numbers. Because of this, folks counting on integer behavior would be disrupted. To make the same results possible, some rounding and integer truncation functions have been added to the core of the Expr2 parser. Indeed, dialplan functions can be called from $[..] expressions without the ${...} operators. The only trouble might be in the fact that the arguments to these functions must be specified with a comma. If you try to call the MATH function, for example, and try to say 3 + MATH(7*8), the expression parser will evaluate 7*8 for you into 56, and the MATH function will most likely complain that its input doesn't make any sense.

We also provide access to most of the floating point functions in the C library. (but not all of them).

While we don't expect someone to want to do Fourier analysis in the dialplan, we don't want to preclude it, either.

Here is a list of the 'builtin' functions in Expr2. All other dialplan functions are available by simply calling them (read-only). In other words, you don't need to surround function calls in $[...] expressions with ${...}. Don't jump to conclusions, though! – you still need to wrap variable names in curly braces!

1. COS(x) x is in radians. Results vary from -1 to 1.
2. SIN(x) x is in radians. Results vary from -1 to 1.
3. TAN(x) x is in radians.
4. ACOS(x) x should be a value between -1 and 1.
5. ASIN(x) x should be a value between -1 and 1.
6. ATAN(x) returns the arc tangent in radians; between -PI/2 and PI/2.
7. ATAN2(x,y) returns a result resembling y/x, except that the signs of both args are used to determine the quadrant of the result. Its result is in radians, between -PI and PI.

8. POW(x,y) returns the value of x raised to the power of y.
9. SQRT(x) returns the square root of x.
10. FLOOR(x) rounds x down to the nearest integer.
11. CEIL(x) rounds x up to the nearest integer.
12. ROUND(x) rounds x to the nearest integer, but round halfway cases away from zero.
13. RINT(x) rounds x to the nearest integer, rounding halfway cases to the nearest even integer.
14. TRUNC(x) rounds x to the nearest integer not larger in absolute value.
15. REMAINDER(x,y) computes the remainder of dividing x by y. The return value is x - n*y, where n is the value x/y, rounded to the nearest integer. If this quotient is 1/2, it is rounded to the nearest even number.
16. EXP(x) returns e to the x power.
17. EXP2(x) returns 2 to the x power.
18. LOG(x) returns the natural logarithm of x.
19. LOG2(x) returns the base 2 log of x.
20. LOG10(x) returns the base 10 log of x.

3.6.5 Examples

```
"One Thousand Five Hundred" =~ "(T[^ ]+)"
  returns: Thousand

"One Thousand Five Hundred" =~ "T[^ ]+"
  returns: 8

"One Thousand Five Hundred" : "T[^ ]+"
  returns: 0

"8015551212" : "(...)"
  returns: 801

"3075551212":"...(...)"
  returns: 555
```

```
 ! "One Thousand Five Hundred" =~ "T[^ ]+"
   returns: 0 (because it applies to the string, which is non-null,
                   which it turns to "0", and then looks for the pattern
                   in the "0", and doesn't find it)

 !( "One Thousand Five Hundred" : "T[^ ]+" )
   returns: 1  (because the string doesn't start with a word starting
                    with T, so the match evals to 0, and the ! operator
                    inverts it to 1 ).

 2 + 8 / 2
   returns 6. (because of operator precedence; the division is done first, then the addition).

 2+8/2
   returns 6. Spaces aren't necessary.

(2+8)/2
   returns 5, of course.

(3+8)/2
   returns 5.5 now.

TRUNC((3+8)/2)
   returns 5.

FLOOR(2.5)
   returns 2

FLOOR(-2.5)
   returns -3

CEIL(2.5)
   returns 3.

CEIL(-2.5)
   returns -2.

ROUND(2.5)
   returns 3.

ROUND(3.5)
   returns 4.

ROUND(-2.5)
   returns -3

RINT(2.5)
   returns 2.

RINT(3.5)
   returns 4.

RINT(-2.5)
   returns -2.

RINT(-3.5)
   returns -4.
```

```
TRUNC(2.5)
   returns 2.

TRUNC(3.5)
   returns 3.

TRUNC(-3.5)
   returns -3.
```

Of course, all of the above examples use constants, but would work the same if any of the numeric or string constants were replaced with a variable reference ${CALLERID(num)}, for instance.

3.6.6 Numbers Vs. Strings

Tokens consisting only of numbers are converted to 'long double' if possible, which are from 80 bits to 128 bits depending on the OS, compiler, and hardware. This means that overflows can occur when the numbers get above 18 digits (depending on the number of bits involved). Warnings will appear in the logs in this case.

3.6.7 Conditionals

There is one conditional application - the conditional goto :

```
exten => 1,2,GotoIf(condition?label1:label2)
```

If condition is true go to label1, else go to label2. Labels are interpreted exactly as in the normal goto command.

"condition" is just a string. If the string is empty or "0", the condition is considered to be false, if it's anything else, the condition is true. This is designed to be used together with the expression syntax described above, eg :

```
exten => 1,2,GotoIf($[${CALLERID(all)} = 123456]?2,1:3,1)
```

Example of use :

```
exten => s,2,Set(vara=1)
exten => s,3,Set(varb=$[${vara} + 2])
exten => s,4,Set(varc=$[${varb} * 2])
exten => s,5,GotoIf($[${varc} = 6]?99,1:s,6)
```

3.6.8 Parse Errors

Syntax errors are now output with 3 lines.

If the extensions.conf file contains a line like:

```
exten => s,6,GotoIf($[ "${CALLERID(num)}"  = "3071234567" & &  "${CALLERID(name)}" : "Privacy Manager" ]?callerid-li
```

You may see an error in `/var/log/asterisk/messages` like this:

```
Jul 15 21:27:49 WARNING[1251240752]: ast_yyerror(): syntax error: parse error, unexpected TOK_AND, expecting TOK_MIN
"3072312154"  = "3071234567" & & "Steves Extension" : "Privacy Manager"
              ^
```

The log line tells you that a syntax error was encountered. It now also tells you (in grand standard bison format) that it hit an "AND" (&) token unexpectedly, and that was hoping for for a MINUS (-), LP (left parenthesis), or a plain token (a string or number).

The next line shows the evaluated expression, and the line after that, the position of the parser in the expression when it became confused, marked with the "^" character.

3.6.9 NULL Strings

Testing to see if a string is null can be done in one of two different ways:

```
exten => _XX.,1,GotoIf($["${calledid}" != ""]?3)
  or
exten => _XX.,1,GotoIf($[foo${calledid} != foo]?3)
```

The second example above is the way suggested by the WIKI. It will work as long as there are no spaces in the evaluated value.

The first way should work in all cases, and indeed, might now be the safest way to handle this situation.

3.6.10 Warning

If you need to do complicated things with strings, asterisk expressions is most likely NOT the best way to go about it. AGI scripts are an excellent option to this need, and make available the full power of whatever language you desire, be it Perl, C, C++, Cobol, RPG, Java, Snobol, PL/I, Scheme, Common Lisp, Shell scripts, Tcl, Forth, Modula, Pascal, APL, assembler, etc.

3.6.11 Incompatabilities

The asterisk expression parser has undergone some evolution. It is hoped that the changes will be viewed as positive.

The "original" expression parser had a simple, hand-written scanner, and a simple bison grammar. This was upgraded to a more involved bison grammar, and a hand-written scanner upgraded to allow extra spaces, and to generate better error diagnostics. This upgrade required bison 1.85, and part of the user community felt the pain of having to upgrade their bison version.

The next upgrade included new bison and flex input files, and the makefile was upgraded to detect current version of both flex and bison, conditionally compiling and linking the new files if the versions of flex and bison would allow it.

If you have not touched your extensions.conf files in a year or so, the above upgrades may cause you some heartburn in certain circumstances, as several changes have been made, and these will affect asterisk's behavior on legacy extension.conf constructs. The changes have been engineered to minimize these conflicts, but there are bound to be problems.

The following list gives some (and most likely, not all) of areas of possible concern with "legacy" extension.conf files:

1. Tokens separated by space(s). Previously, tokens were separated by spaces. Thus, ' 1 + 1 ' would evaluate to the value '2', but '1+1' would evaluate to the string '1+1'. If this behavior was depended on, then the expression evaluation will break. '1+1' will now evaluate to '2', and something is not going to work right. To keep such strings from being evaluated, simply wrap them in double quotes: ' "1+1" '

2. The colon operator. In versions previous to double quoting, the colon operator takes the right hand string, and using it as a regex pattern, looks for it in the left hand string. It is given an implicit ôperator at the beginning, meaning the pattern will match only at the beginning of the left hand string. If the pattern or the matching string had double quotes around them, these could get in the way of the pattern match. Now, the wrapping double quotes are stripped from both the pattern and the left hand string before applying the pattern. This was done because it recognized that the new way of scanning the expression doesn't use spaces to separate tokens, and the average regex expression is full of

operators that the scanner will recognize as expression operators. Thus, unless the pattern is wrapped in double quotes, there will be trouble. For instance, ${VAR1} : (Who|What*)+ may have have worked before, but unless you wrap the pattern in double quotes now, look out for trouble! This is better: "${VAR1}" : "(Who|What*)+" and should work as previous.

3. Variables and Double Quotes Before these changes, if a variable's value contained one or more double quotes, it was no reason for concern. It is now!

4. LE, GE, NE operators removed. The code supported these operators, but they were not documented. The symbolic operators, <=, >=, and != should be used instead.

5. Added the unary '-' operator. So you can 3+ -4 and get -1.

6. Added the unary '!' operator, which is a logical complement. Basically, if the string or number is null, empty, or '0', a '1' is returned. Otherwise a '0' is returned.

7. Added the '= ' operator, just in case someone is just looking for match anywhere in the string. The only diff with the ':' is that match doesn't have to be anchored to the beginning of the string.

8. Added the conditional operator 'expr1 ? true_expr : false_expr' First, all 3 exprs are evaluated, and if expr1 is false, the 'false_expr' is returned as the result. See above for details.

9. Unary operators '-' and '!' were made right associative.

3.6.12 Debugging Hints

There are two utilities you can build to help debug the $[ ] in your extensions.conf file.

The first, and most simplistic, is to issue the command:

```
make testexpr2
```

in the top level asterisk source directory. This will build a small executable, that is able to take the first command line argument, and run it thru the

expression parser. No variable substitutions will be performed. It might be safest to wrap the expression in single quotes...

```
testexpr2 '2*2+2/2'
```

is an example.

And, in the utils directory, you can say:

```
make check_expr
```

and a small program will be built, that will check the file mentioned in the first command line argument, for any expressions that might be have problems when you move to flex-2.5.31. It was originally designed to help spot possible incompatibilities when moving from the pre-2.5.31 world to the upgraded version of the lexer.

But one more capability has been added to check_expr, that might make it more generally useful. It now does a simple minded evaluation of all variables, and then passes the $[] exprs to the parser. If there are any parse errors, they will be reported in the log file. You can use check_expr to do a quick sanity check of the expressions in your extensions.conf file, to see if they pass a crude syntax check.

The "simple-minded" variable substitution replaces ${varname} variable references with '555'. You can override the 555 for variable values, by entering in var=val arguments after the filename on the command line. So...

```
check_expr /etc/asterisk/extensions.conf CALLERID(num)=3075551212 DIALSTATUS=TORTURE EXTEN=121
```

will substitute any ${CALLERID(num)} variable references with 3075551212, any ${DIALSTATUS} variable references with 'TORTURE', and any ${EXTEN} references with '121'. If there is any fancy stuff going on in the reference, like ${EXTEN:2}, then the override will not work. Everything in the ${...} has to match. So, to substitute ${EXTEN:2} references, you'd best say:

```
check_expr /etc/asterisk/extensions.conf CALLERID(num)=3075551212 DIALSTATUS=TORTURE EXTEN:2=121
```

on stdout, you will see something like:

```
OK -- $[ "${DIALSTATUS}"  = "TORTURE" | "${DIALSTATUS}" = "DONTCALL" ] at line 416
```

In the expr2_log file that is generated, you will see:

```
line 416, evaluation of $[  "TORTURE"  = "TORTURE" | "TORTURE" = "DONTCALL"  ] result: 1
```

check_expr is a very simplistic algorithm, and it is far from being guaranteed to work in all cases, but it is hoped that it will be useful.

3.7 Asterisk standard channel variables

There are a number of variables that are defined or read by Asterisk. Here is a list of them. More information is available in each application's help text. All these variables are in UPPER CASE only.

Variables marked with a * are builtin functions and can't be set, only read in the dialplan. Writes to such variables are silently ignored.

```
${CDR(accountcode)}    * Account code (if specified)
${BLINDTRANSFER}         The name of the channel on the other side of a blind tran
${BRIDGEPEER}            Bridged peer
${BRIDGEPVTCALLID}       Bridged peer PVT call ID (SIP Call ID if a SIP call)
${CALLERID(ani)}       * Caller ANI (PRI channels)
${CALLERID(ani2)}      * ANI2 (Info digits) also called Originating line informati
${CALLERID(all)}       * Caller ID
${CALLERID(dnid)}      * Dialed Number Identifier
${CALLERID(name)}      * Caller ID Name only
${CALLERID(num)}       * Caller ID Number only
${CALLERID(rdnis)}     * Redirected Dial Number ID Service
${CALLINGANI2}         * Caller ANI2 (PRI channels)
${CALLINGPRES}         * Caller ID presentation for incoming calls (PRI channels)
${CALLINGTNS}          * Transit Network Selector (PRI channels)
${CALLINGTON}          * Caller Type of Number (PRI channels)
${CHANNEL}             * Current channel name
${CONTEXT}             * Current context
${DATETIME}            * Current date time in the format: DDMMYYYY-HH:MM:SS
                         (Deprecated; use ${STRFTIME(${EPOCH},,%d%m%Y-%H:%M:%S)})
${DB_RESULT}             Result value of DB_EXISTS() dial plan function
${EPOCH}               * Current unix style epoch
${EXTEN}               * Current extension
${ENV(VAR)}              Environmental variable VAR
${GOTO_ON_BLINDXFR}      Transfer to the specified context/extension/priority
                         after a blind transfer (use ^ characters in place of
                         | to separate context/extension/priority when setting
                         this variable from the dialplan)
${HANGUPCAUSE}         * Asterisk cause of hangup (inbound/outbound)
${HINT}                * Channel hints for this extension
${HINTNAME}            * Suggested Caller*ID name for this extension
```

```
${INVALID_EXTEN}         The invalid called extension (used in the "i" extension)
${LANGUAGE}            * Current language (Deprecated; use ${LANGUAGE()})
${LEN(VAR)}            * String length of VAR (integer)
${PRIORITY}            * Current priority in the dialplan
${PRIREDIRECTREASON}     Reason for redirect on PRI, if a call was directed
${TIMESTAMP}           * Current date time in the format: YYYYMMDD-HHMMSS
                         (Deprecated; use ${STRFTIME(${EPOCH},,%Y%m%d-%H%M%S)})
${TRANSFER_CONTEXT}      Context for transferred calls
${FORWARD_CONTEXT}       Context for forwarded calls
${UNIQUEID}            * Current call unique identifier
${SYSTEMNAME}          * value of the systemname option of asterisk.conf
${ENTITYID}            * Global Entity ID set automatically, or from asterisk.conf
```

3.7.1 Application return values

Many applications return the result in a variable that you read to get the result of the application. These status fields are unique for each application. For the various status values, see each application's help text.

```
${AGISTATUS}          * agi()
${AQMSTATUS}          * addqueuemember()
${AVAILSTATUS}        * chanisavail()
${CHECKGROUPSTATUS}   * checkgroup()
${CHECKMD5STATUS}     * checkmd5()
${CPLAYBACKSTATUS}    * controlplayback()
${DIALSTATUS}         * dial()
${DBGETSTATUS}        * dbget()
${ENUMSTATUS}         * enumlookup()
${HASVMSTATUS}        * hasnewvoicemail()
${LOOKUPBLSTATUS}     * lookupblacklist()
${OSPAUTHSTATUS}      * ospauth()
${OSPLOOKUPSTATUS}    * osplookup()
${OSPNEXTSTATUS}      * ospnext()
${OSPFINISHSTATUS}    * ospfinish()
${PARKEDAT}           * parkandannounce()
${PLAYBACKSTATUS}     * playback()
${PQMSTATUS}          * pausequeuemember()
${PRIVACYMGRSTATUS}   * privacymanager()
```

```
${QUEUESTATUS}       * queue()
${RQMSTATUS}         * removequeuemember()
${SENDIMAGESTATUS}   * sendimage()
${SENDTEXTSTATUS}    * sendtext()
${SENDURLSTATUS}     * sendurl()
${SYSTEMSTATUS}      * system()
${TRANSFERSTATUS}    * transfer()
${TXTCIDNAMESTATUS}  * txtcidname()
${UPQMSTATUS}        * unpausequeuemember()
${VMSTATUS}          * voicmail()
${VMBOXEXISTSSTATUS} * vmboxexists()
${WAITSTATUS}        * waitforsilence()
```

3.7.2 Various application variables

```
${CURL}                 * Resulting page content for curl()
${ENUM}                 * Result of application EnumLookup
${EXITCONTEXT}            Context to exit to in IVR menu (app background())
                          or in the RetryDial() application
${MONITOR}              * Set to "TRUE" if the channel is/has been monitored (app
${MONITOR_EXEC}           Application to execute after monitoring a call
${MONITOR_EXEC_ARGS}      Arguments to application
${MONITOR_FILENAME}       File for monitoring (recording) calls in queue
${QUEUE_PRIO}             Queue priority
${QUEUE_MAX_PENALTY}      Maximum member penalty allowed to answer caller
${QUEUE_MIN_PENALTY}      Minimum member penalty allowed to answer caller
${QUEUESTATUS}            Status of the call, one of:
                          (TIMEOUT | FULL | JOINEMPTY | LEAVEEMPTY | JOINUNAVAIL |
${RECORDED_FILE}        * Recorded file in record()
${TALK_DETECTED}        * Result from talkdetect()
${TOUCH_MONITOR}          The filename base to use with Touch Monitor (auto record
${TOUCH_MONITOR_PREF}   * The prefix for automonitor recording filenames.
${TOUCH_MONITOR_FORMAT}   The audio format to use with Touch Monitor (auto record)
${TOUCH_MONITOR_OUTPUT} * Recorded file from Touch Monitor (auto record)
${TOUCH_MONITOR_MESSAGE_START} Recorded file to play for both channels at start of
${TOUCH_MONITOR_MESSAGE_STOP} Recorded file to play for both channels at end of mo
${TXTCIDNAME}           * Result of application TXTCIDName
${VPB_GETDTMF}            chan_vpb
```

3.7.3 The MeetMe Conference Bridge

```
${MEETME_RECORDINGFILE}      Name of file for recording a conference with
                             the "r" option
${MEETME_RECORDINGFORMAT}    Format of file to be recorded
${MEETME_EXIT_CONTEXT}       Context for exit out of meetme meeting
${MEETME_AGI_BACKGROUND}     AGI script for Meetme (DAHDI only)
${MEETMESECS}              * Number of seconds a user participated in a MeetMe cor
${CONF_LIMIT_TIMEOUT_FILE}   File to play when time is up.  Used with the L() opti
${CONF_LIMIT_WARNING_FILE}   File to play as warning if 'y' is defined.
                             The default is to say the time remaining.  Used with
```

3.7.4 The VoiceMail() application

```
${VM_CATEGORY}      Sets voicemail category
${VM_NAME}        * Full name in voicemail
${VM_DUR}         * Voicemail duration
${VM_MSGNUM}      * Number of voicemail message in mailbox
${VM_CALLERID}    * Voicemail Caller ID (Person leaving vm)
${VM_CIDNAME}     * Voicemail Caller ID Name
${VM_CIDNUM}      * Voicemail Caller ID Number
${VM_DATE}        * Voicemail Date
${VM_MESSAGEFILE} * Path to message left by caller
```

3.7.5 The VMAuthenticate() application

```
${AUTH_MAILBOX}   * Authenticated mailbox
${AUTH_CONTEXT}   * Authenticated mailbox context
```

3.7.6 DUNDiLookup()

```
${DUNDTECH}       * The Technology of the result from a call to DUNDiLookup()
${DUNDDEST}       * The Destination of the result from a call to DUNDiLookup()
```

3.7.7 chan_dahdi

```
${ANI2}             * The ANI2 Code provided by the network on the incoming call
                      (ie, Code 29 identifies call as a Prison/Inmate Call)
```

```
${CALLTYPE}           * Type of call (Speech, Digital, etc)
${CALLEDTON}          * Type of number for incoming PRI extension
                        i.e. 0=unknown, 1=international, 2=domestic, 3=net_specifi
                        4=subscriber, 6=abbreviated, 7=reserved
${CALLINGSUBADDR}     * Caller's PRI Subaddress
${FAXEXTEN}           * The extension called before being redirected to "fax"
${PRIREDIRECTREASON}  * Reason for redirect, if a call was directed
${SMDI_VM_TYPE}       * When an call is received with an SMDI message, the 'type'
                        of message 'b' or 'u'
```

3.7.8 chan_sip

```
${SIPCALLID}         * SIP Call-ID: header verbatim (for logging or CDR matching)
${SIPDOMAIN}         * SIP destination domain of an inbound call (if appropriate)
${SIPUSERAGENT}      * SIP user agent (deprecated)
${SIPURI}            * SIP uri
${SIP_CODEC}           Set the SIP codec for a call
${SIP_URI_OPTIONS}   * additional options to add to the URI for an outgoing call
${RTPAUDIOQOS}         RTCP QoS report for the audio of this call
${RTPVIDEOQOS}         RTCP QoS report for the video of this call
```

3.7.9 chan_agent

```
${AGENTMAXLOGINTRIES}  Set the maximum number of failed logins
${AGENTUPDATECDR}      Whether to update the CDR record with Agent channel data
${AGENTGOODBYE}        Sound file to use for "Good Bye" when agent logs out
${AGENTACKCALL}        Whether the agent should acknowledge the incoming call
${AGENTAUTOLOGOFF}     Auto logging off for an agent
${AGENTWRAPUPTIME}     Setting the time for wrapup between incoming calls
${AGENTNUMBER}       * Agent number (username) set at login
${AGENTSTATUS}       * Status of login ( fail | on | off )
${AGENTEXTEN}        * Extension for logged in agent
```

3.7.10 The Dial() application

```
${DIALEDPEERNAME}     * Dialed peer name
${DIALEDPEERNUMBER}   * Dialed peer number
${DIALEDTIME}         * Time for the call (seconds). Is only set if call was answe
```

```
${ANSWEREDTIME}       * Time from answer to hangup (seconds)
${DIALSTATUS}         * Status of the call, one of:
                        (CHANUNAVAIL | CONGESTION | BUSY | NOANSWER
                        | ANSWER | CANCEL | DONTCALL | TORTURE)
${DYNAMIC_FEATURES}   * The list of features (from the [applicationmap] section of
                        features.conf) to activate during the call, with feature
                        names separated by '#' characters
${LIMIT_PLAYAUDIO_CALLER}  Soundfile for call limits
${LIMIT_PLAYAUDIO_CALLEE}  Soundfile for call limits
${LIMIT_WARNING_FILE}      Soundfile for call limits
${LIMIT_TIMEOUT_FILE}      Soundfile for call limits
${LIMIT_CONNECT_FILE}      Soundfile for call limits
${OUTBOUND_GROUP}          Default groups for peer channels (as in SetGroup)
                           * See "show application dial" for more information
```

3.7.11 The chanisavail() application

```
${AVAILCHAN}          * the name of the available channel if one was found
${AVAILORIGCHAN}      * the canonical channel name that was used to create the cha
${AVAILSTATUS}        * Status of requested channel
```

3.7.12 Dialplan Macros

```
${MACRO_EXTEN}        * The calling extensions
${MACRO_CONTEXT}      * The calling context
${MACRO_PRIORITY}     * The calling priority
${MACRO_OFFSET}         Offset to add to priority at return from macro
```

3.7.13 The ChanSpy() application

```
${SPYGROUP}           * A ':' (colon) separated list of group names.
                        (To be set on spied on channel and matched against the g(g
```

3.7.14 OSP

```
${OSPINHANDLE}          OSP handle of in_bound call
${OSPINTIMELIMIT}       Duration limit for in_bound call
${OSPOUTHANDLE}         OSP handle of out_bound call
```

${OSPTECH}	OSP technology
${OSPDEST}	OSP destination
${OSPCALLING}	OSP calling number
${OSPOUTTOKEN}	OSP token to use for out_bound call
${OSPOUTTIMELIMIT}	Duration limit for out_bound call
${OSPRESULTS}	Number of remained destinations

Chapter 4

AEL: Asterisk Extension Language

4.1 Introduction

AEL is a specialized language intended purely for describing Asterisk dial plans.

The current version was written by Steve Murphy, and is a rewrite of the original version.

This new version further extends AEL, and provides more flexible syntax, better error messages, and some missing functionality.

AEL is really the merger of 4 different 'languages', or syntaxes:

- The first and most obvious is the AEL syntax itself. A BNF is provided near the end of this document.
- The second syntax is the Expression Syntax, which is normally handled by Asterisk extension engine, as expressions enclosed in $[...]. The right hand side of assignments are wrapped in $[...] by AEL, and so are the if and while expressions, among others.
- The third syntax is the Variable Reference Syntax, the stuff enclosed in ${..} curly braces. It's a bit more involved than just putting a variable name in there. You can include one of dozens of 'functions', and their arguments, and there are even some string manipulation notation in there.
- The last syntax that underlies AEL, and is not used directly in AEL, is the Extension Language Syntax. The extension language is what

you see in extensions.conf, and AEL compiles the higher level AEL language into extensions and priorities, and passes them via function calls into Asterisk. Embedded in this language is the Application/AGI commands, of which one application call per step, or priority can be made. You can think of this as a "macro assembler" language, that AEL will compile into.

Any programmer of AEL should be familiar with its syntax, of course, as well as the Expression syntax, and the Variable syntax.

4.2 Asterisk in a Nutshell

Asterisk acts as a server. Devices involved in telephony, like DAHDI cards, or Voip phones, all indicate some context that should be activated in their behalf. See the config file formats for IAX, SIP, dahdi.conf, etc. They all help describe a device, and they all specify a context to activate when somebody picks up a phone, or a call comes in from the phone company, or a voip phone, etc.

4.2.1 Contexts

Contexts are a grouping of extensions.

Contexts can also include other contexts. Think of it as a sort of merge operation at runtime, whereby the included context's extensions are added to the contexts making the inclusion.

4.2.2 Extensions and priorities

A Context contains zero or more Extensions. There are several predefined extensions. The "s" extension is the "start" extension, and when a device activates a context the "s" extension is the one that is going to be run. Other extensions are the timeout "t" extension, the invalid response, or "i" extension, and there's a "fax" extension. For instance, a normal call will activate the "s" extension, but an incoming FAX call will come into the "fax" extension, if it exists. (BTW, asterisk can tell it's a fax call by the little "beep" that the calling fax machine emits every so many seconds.).

Extensions contain several priorities, which are individual instructions to perform. Some are as simple as setting a variable to a value. Others are as

complex as initiating the Voicemail application, for instance. Priorities are executed in order.

When the 's" extension completes, asterisk waits until the timeout for a response. If the response matches an extension's pattern in the context, then control is transferred to that extension. Usually the responses are tones emitted when a user presses a button on their phone. For instance, a context associated with a desk phone might not have any "s" extension. It just plays a dialtone until someone starts hitting numbers on the keypad, gather the number, find a matching extension, and begin executing it. That extension might Dial out over a connected telephone line for the user, and then connect the two lines together.

The extensions can also contain "goto" or "jump" commands to skip to extensions in other contexts. Conditionals provide the ability to react to different stimuli, and there you have it.

4.2.3 Macros

Think of a macro as a combination of a context with one nameless extension, and a subroutine. It has arguments like a subroutine might. A macro call can be made within an extension, and the individual statements there are executed until it ends. At this point, execution returns to the next statement after the macro call. Macros can call other macros. And they work just like function calls.

4.2.4 Applications

Application calls, like "Dial()", or "Hangup()", or "Answer()", are available for users to use to accomplish the work of the dialplan. There are over 145 of them at the moment this was written, and the list grows as new needs and wants are uncovered. Some applications do fairly simple things, some provide amazingly complex services.

Hopefully, the above objects will allow you do anything you need to in the Asterisk environment!

4.3 Getting Started

The AEL parser (res_ael.so) is completely separate from the module that parses extensions.conf (pbx_config.so). To use AEL, the only thing that has to be done is the module res_ael.so must be loaded by Asterisk. This will be done automatically if using 'autoload=yes' in `/etc/asterisk/modules.conf`. When the module is loaded, it will look for 'extensions.ael' in `/etc/asterisk/`. extensions.conf and extensions.ael can be used in conjunction with each other if that is what is desired. Some users may want to keep extensions.conf for the features that are configured in the 'general' section of extensions.conf.

To reload extensions.ael, the following command can be issued at the CLI:
*CLI¿ ael reload

4.4 Debugging

Right at this moment, the following commands are available, but do nothing:

Enable AEL contexts debug
*CLI> ael debug contexts
Enable AEL macros debug
*CLI> ael debug macros
Enable AEL read debug
*CLI> ael debug read
Enable AEL tokens debug
*CLI> ael debug tokens
Disable AEL debug messages
*CLI> ael no debug

If things are going wrong in your dialplan, you can use the following facilities to debug your file:

1. The messages log in `/var/log/asterisk`. (from the checks done at load time). 2. the "show dialplan" command in asterisk 3. the standalone executable, "aelparse" built in the utils/ dir in the source.

4.5 About "aelparse"

You can use the "aelparse" program to check your extensions.ael file before feeding it to asterisk. Wouldn't it be nice to eliminate most errors before

giving the file to asterisk?

aelparse is compiled in the utils directory of the asterisk release. It isn't installed anywhere (yet). You can copy it to your favorite spot in your PATH.

aelparse has two optional arguments:

- -d
  - Override the normal location of the config file dir, (usually `/etc/asterisk`), and use the current directory instead as the config file dir. Aelparse will then expect to find the file "./extensions.ael" in the current directory, and any included files in the current directory as well.
- -n
  - don't show all the function calls to set priorities and contexts within asterisk. It will just show the errors and warnings from the parsing and semantic checking phases.

4.6 General Notes about Syntax

Note that the syntax and style are now a little more free-form. The opening '' (curly-braces) do not have to be on the same line as the keyword that precedes them. Statements can be split across lines, as long as tokens are not broken by doing so. More than one statement can be included on a single line. Whatever you think is best!

You can just as easily say,

```
if(${x}=1) { NoOp(hello!); goto s,3; } else { NoOp(Goodbye!); goto s,12; }
```

as you can say:

```
if(${x}=1)
{
       NoOp(hello!);
   goto s,3;
}
else
{
       NoOp(Goodbye!);
       goto s,12;
}
```

or:

```
if(${x}=1) {
       NoOp(hello!);
   goto s,3;
} else {
       NoOp(Goodbye!);
       goto s,12;
}
```

or:

```
if (${x}=1) {
       NoOp(hello!); goto s,3;
} else {
       NoOp(Goodbye!); goto s,12;
}
```

4.7 Keywords

The AEL keywords are case-sensitive. If an application name and a keyword overlap, there is probably good reason, and you should consider replacing the application call with an AEL statement. If you do not wish to do so, you can still use the application, by using a capitalized letter somewhere in its name. In the Asterisk extension language, application names are NOT case-sensitive.

The following are keywords in the AEL language:

- abstract
- context
- macro
- globals
- ignorepat
- switch
- if
- ifTime
- else

- random
- goto
- jump
- local
- return
- break
- continue
- regexten
- hint
- for
- while
- case
- pattern
- default NOTE: the "default" keyword can be used as a context name, for those who would like to do so.
- catch
- switches
- eswitches
- includes

4.8 Procedural Interface and Internals

AEL first parses the extensions.ael file into a memory structure representing the file. The entire file is represented by a tree of "pval" structures linked together.

This tree is then handed to the semantic check routine.

Then the tree is handed to the compiler.

After that, it is freed from memory.

A program could be written that could build a tree of pval structures, and a pretty printing function is provided, that would dump the data to a file, or the tree could be handed to the compiler to merge the data into the asterisk dialplan. The modularity of the design offers several opportunities for developers to simplify apps to generate dialplan data.

4.8.1 AEL version 2 BNF

(hopefully, something close to bnf).

First, some basic objects

```
------------------------
<word>    a lexical token consisting of characters matching this pattern: [-a-zA-Z0-9"_/.\<\>\*\+!$#\[\]][-a-zA-Z0-9

<word3-list>  a concatenation of up to 3 <word>s.

<collected-word>  all characters encountered until the character that follows the <collected-word> in the grammar.
-------------------------

<file> :== <objects>

<objects> :== <object>
           | <objects> <object>


<object> :==  <context>
         | <macro>
         | <globals>
         | ';'


<context> :==  'context' <word> '{' <elements> '}'
            | 'context' <word> '{' '}'
            | 'context' 'default' '{' <elements> '}'
            | 'context' 'default' '{' '}'
            | 'abstract'  'context' <word> '{' <elements> '}'
            | 'abstract'  'context' <word> '{' '}'
            | 'abstract'  'context' 'default' '{' <elements> '}'
            | 'abstract'  'context' 'default' '{' '}'
```

```
<macro> :== 'macro' <word> '(' <arglist> ')' '{' <macro_statements> '}'
       | 'macro' <word> '(' <arglist> ')' '{'  '}'
       | 'macro' <word> '(' ')' '{' <macro_statements> '}'
       | 'macro' <word> '(' ')' '{'  '}'


<globals> :== 'globals' '{' <global_statements> '}'
         | 'globals' '{' '}'


<global_statements> :== <global_statement>
                  | <global_statements> <global_statement>


<global_statement> :== <word> '=' <collected-word> ';'


<arglist> :== <word>
         | <arglist> ',' <word>


<elements> :==  <element>
             | <elements> <element>


<element> :== <extension>
         | <includes>
         | <switches>
         | <eswitches>
         | <ignorepat>
         | <word> '='  <collected-word> ';'
         | 'local' <word> '='  <collected-word> ';'
         | ';'


<ignorepat> :== 'ignorepat' '=>' <word> ';'


<extension> :== <word> '=>' <statement>
           | 'regexten' <word> '=>' <statement>
           | 'hint' '(' <word3-list> ')' <word> '=>' <statement>
           | 'regexten' 'hint' '(' <word3-list> ')' <word> '=>' <statement>


<statements> :== <statement>
            | <statements> <statement>

<if_head> :== 'if' '('  <collected-word> ')'

<random_head> :== 'random' '(' <collected-word> ')'

<ifTime_head> :== 'ifTime' '(' <word3-list> ':' <word3-list> ':' <word3-list> '|' <word3-list> '|' <word3-list> '|'
                     | 'ifTime' '(' <word> '|' <word3-list> '|' <word3-list> '|' <word3-list> ')'


<word3-list> :== <word>
```

```
       | <word> <word>
       | <word> <word> <word>

<switch_head> :== 'switch' '(' <collected-word> ')'  '{'


<statement> :== '{' <statements> '}'
       | <word> '='  <collected-word> ';'
       | 'local' <word> '='  <collected-word> ';'
       | 'goto' <target> ';'
       | 'jump' <jumptarget> ';'
       | <word> ':'
       | 'for' '('  <collected-word> ';'  <collected-word> ';' <collected-word> ')' <statement>
       | 'while' '('  <collected-word> ')' <statement>
       | <switch_head> '}'
       | <switch_head> <case_statements> '}'
       | '&' macro_call ';'
       | <application_call> ';'
       | <application_call> '='  <collected-word> ';'
       | 'break' ';'
       | 'return' ';'
       | 'continue' ';'
       | <random_head> <statement>
       | <random_head> <statement> 'else' <statement>
       | <if_head> <statement>
       | <if_head> <statement> 'else' <statement>
       | <ifTime_head> <statement>
       | <ifTime_head> <statement> 'else' <statement>
       | ';'

<target> :== <word>
       | <word> '|' <word>
       | <word> '|' <word> '|' <word>
       | 'default' '|' <word> '|' <word>
       | <word> ',' <word>
       | <word> ',' <word> ',' <word>
       | 'default' ',' <word> ',' <word>

<jumptarget> :== <word>
               | <word> ',' <word>
               | <word> ',' <word> '@' <word>
               | <word> '@' <word>
               | <word> ',' <word> '@' 'default'
               | <word> '@' 'default'

<macro_call> :== <word> '(' <eval_arglist> ')'
       | <word> '(' ')'

<application_call_head> :== <word>  '('

<application_call> :== <application_call_head> <eval_arglist> ')'
       | <application_call_head> ')'

<eval_arglist> :==  <collected-word>
       | <eval_arglist> ','  <collected-word>
       |  /* nothing */
       | <eval_arglist> ','  /* nothing */
```

```
<case_statements> :== <case_statement>
       | <case_statements> <case_statement>


<case_statement> :== 'case' <word> ':' <statements>
       | 'default' ':' <statements>
       | 'pattern' <word> ':' <statements>
       | 'case' <word> ':'
       | 'default' ':'
       | 'pattern' <word> ':'

<macro_statements> :== <macro_statement>
       | <macro_statements> <macro_statement>

<macro_statement> :== <statement>
       | 'catch' <word> '{' <statements> '}'

<switches> :== 'switches' '{' <switchlist> '}'
       | 'switches' '{' '}'

<eswitches> :== 'eswitches' '{' <switchlist> '}'
       | 'eswitches' '{'  '}'

<switchlist> :== <word> ';'
       | <switchlist> <word> ';'

<includeslist> :== <includedname> ';'
       | <includedname> '|' <word3-list> ':' <word3-list> ':' <word3-list> '|' <word3-list> '|' <word3-list> '|' <wo
       | <includedname> '|' <word> '|' <word3-list> '|' <word3-list> '|' <word3-list> ';'
       | <includeslist> <includedname> ';'
       | <includeslist> <includedname> '|' <word3-list> ':' <word3-list> ':' <word3-list> '|' <word3-list> '|' <word
       | <includeslist> <includedname> '|' <word> '|' <word3-list> '|' <word3-list> '|' <word3-list> ';'

<includedname> :== <word>
        | 'default'

<includes> :== 'includes' '{' <includeslist> '}'
       | 'includes' '{' '}'
```

4.9 AEL Example USAGE

4.9.1 Comments

Comments begin with // and end with the end of the line.

Comments are removed by the lexical scanner, and will not be recognized in places where it is busy gathering expressions to wrap in $[] , or inside application call argument lists. The safest place to put comments is after terminating semicolons, or on otherwise empty lines.

4.9.2 Context

Contexts in AEL represent a set of extensions in the same way that they do in extensions.conf.

```
context default {

}
```

A context can be declared to be "abstract", in which case, this declaration expresses the intent of the writer, that this context will only be included by another context, and not "stand on its own". The current effect of this keyword is to prevent "goto " statements from being checked.

```
abstract context longdist {
     _1NXXNXXXXXX => NoOp(generic long distance dialing actions in the US);
}
```

4.9.3 Extensions

To specify an extension in a context, the following syntax is used. If more than one application is be called in an extension, they can be listed in order inside of a block.

```
context default {
    1234 => Playback(tt-monkeys);
    8000 => {
         NoOp(one);
         NoOp(two);
         NoOp(three);
    };
    _5XXX => NoOp(it's a pattern!);
}
```

Two optional items have been added to the AEL syntax, that allow the specification of hints, and a keyword, regexten, that will force the numbering of priorities to start at 2.

The ability to make extensions match by CID is preserved in AEL; just use '/' and the CID number in the specification. See below.

```
context default {

    regexten _5XXX => NoOp(it's a pattern!);
}

context default {

    hint(Sip/1) _5XXX => NoOp(it's a pattern!);
}
```

```
context default {

    regexten hint(Sip/1) _5XXX => NoOp(it's a pattern!);
}
```

The regexten must come before the hint if they are both present.

CID matching is done as with the extensions.conf file. Follow the extension name/number with a slash (/) and the number to match against the Caller ID:

```
context zoombo
{
  819/7079953345 => { NoOp(hello, 3345); }
}
```

In the above, the 819/7079953345 extension will only be matched if the CallerID is 7079953345, and the dialed number is 819. Hopefully you have another 819 extension defined for all those who wish 819, that are not so lucky as to have 7079953345 as their CallerID!

4.9.4 Includes

Contexts can be included in other contexts. All included contexts are listed within a single block.

```
context default {
    includes {
         local;
         longdistance;
         international;
    }
}
```

Time-limited inclusions can be specified, as in extensions.conf format, with the fields described in the wiki page Asterisk cmd GotoIfTime.

```
context default {
    includes {
         local;
         longdistance|16:00-23:59|mon-fri|*|*;
         international;
    }
}
```

4.9.5 #include

You can include other files with the #include "filepath" construct.

```
#include "/etc/asterisk/testfor.ael"
```

An interesting property of the #include, is that you can use it almost anywhere in the .ael file. It is possible to include the contents of a file in a macro, context, or even extension. The #include does not have to occur at the beginning of a line. Included files can include other files, up to 50 levels deep. If the path provided in quotes is a relative path, the parser looks in the config file directory for the file (usually `/etc/asterisk`).

4.9.6 Dialplan Switches

Switches are listed in their own block within a context. For clues as to what these are used for, see Asterisk - dual servers, and Asterisk config extensions.conf.

```
context default {
    switches {
         DUNDi/e164;
         IAX2/box5;
    };
    eswitches {
         IAX2/context@${CURSERVER};
    }
}
```

4.9.7 Ignorepat

ignorepat can be used to instruct channel drivers to not cancel dialtone upon receipt of a particular pattern. The most commonly used example is '9'.

```
context outgoing {
    ignorepat => 9;
}
```

4.9.8 Variables

Variables in Asterisk do not have a type, so to define a variable, it just has to be specified with a value.

Global variables are set in their own block.

```
globals {
    CONSOLE=Console/dsp;
    TRUNK=DAHDI/g2;
}
```

Variables can be set within extensions as well.

```
context foo {
    555 => {
         x=5;
         y=blah;
         divexample=10/2
         NoOp(x is ${x} and y is ${y} !);
    }
}
```

NOTE: AEL wraps the right hand side of an assignment with $[ ] to allow expressions to be used If this is unwanted, you can protect the right hand side from being wrapped by using the Set() application. Read the README.variables about the requirements and behavior of $[ ] expressions.

NOTE: These things are wrapped up in a $[ ] expression: The while() test; the if() test; the middle expression in the for(x; y; z) statement (the y expression); Assignments - the right hand side, so a = b -¿ Set(a=$[b])

Writing to a dialplan function is treated the same as writing to a variable.

```
context blah {
    s => {
         CALLERID(name)=ChickenMan;
         NoOp(My name is ${CALLERID(name)} !);
    }
}
```

You can declare variables in Macros, as so:

```
Macro myroutine(firstarg, secondarg)
{
  Myvar=1;
  NoOp(Myvar is set to ${myvar});
}
```

4.9.9 Local Variables

In 1.2, and 1.4, ALL VARIABLES are CHANNEL variables, including the function arguments and associated ARG1, ARG2, etc variables. Sorry.

In trunk (1.6 and higher), we have made all arguments local variables to a macro call. They will not affect channel variables of the same name. This includes the ARG1, ARG2, etc variables.

Users can declare their own local variables by using the keyword 'local' before setting them to a value;

```
Macro myroutine(firstarg, secondarg)
{
  local Myvar=1;
  NoOp(Myvar is set to ${Myvar}, and firstarg is ${firstarg}, and secondarg is ${secondarg});
}
```

In the above example, Myvar, firstarg, and secondarg are all local variables, and will not be visible to the calling code, be it an extension, or another Macro.

If you need to make a local variable within the Set() application, you can do it this way:

```
Macro myroutine(firstarg, secondarg)
{
  Set(LOCAL(Myvar)=1);
  NoOp(Myvar is set to ${Myvar}, and firstarg is ${firstarg}, and secondarg is ${secondarg});
}
```

4.9.10 Loops

AEL has implementations of 'for' and 'while' loops.

```
context loops {
    1 => {
         for (x=0; ${x} < 3; x=${x} + 1) {
              Verbose(x is ${x} !);
         }
    }
    2 => {
         y=10;
         while (${y} >= 0) {
              Verbose(y is ${y} !);
              y=${y}-1;
         }
    }
}
```

NOTE: The conditional expression (the "${y} >= 0" above) is wrapped in $[ ] so it can be evaluated. NOTE: The for loop test expression (the "$x < 3" above) is wrapped in $[ ] so it can be evaluated.

4.9.11 Conditionals

AEL supports if and switch statements, like AEL, but adds ifTime, and random. Unlike the original AEL, though, you do NOT need to put curly braces around a single statement in the "true" branch of an if(), the random(), or an ifTime() statement. The if(), ifTime(), and random() statements allow optional else clause.

```
context conditional {
    _8XXX => {
         Dial(SIP/${EXTEN});
         if ("${DIALSTATUS}" = "BUSY")
         {
              NoOp(yessir);
              Voicemail(${EXTEN},b);
         }
         else
              Voicemail(${EXTEN},u);
         ifTime (14:00-25:00,sat-sun,*,*)
              Voicemail(${EXTEN},b);
         else
         {
              Voicemail(${EXTEN},u);
              NoOp(hi, there!);
         }
         random(51) NoOp(This should appear 51% of the time);

         random( 60 )
         {
                      NoOp( This should appear 60% of the time );
         }
         else
         {
                      random(75)
                      {
                              NoOp( This should appear 30% of the time! );
                      }
                      else
                      {
                              NoOp( This should appear 10% of the time! );
                      }
          }
    }
    _777X => {
         switch (${EXTEN}) {
              case 7771:
                   NoOp(You called 7771!);
                   break;
              case 7772:
                   NoOp(You called 7772!);
                   break;
              case 7773:
                   NoOp(You called 7773!);
                   // fall thru-
              pattern 777[4-9]:
                    NoOp(You called 777 something!);
              default:
                   NoOp(In the default clause!);
         }
    }
}
```

NOTE: The conditional expression in if() statements (the "${DIALSTATUS}" = "BUSY" above) is wrapped by the compiler in $[] for evaluation.

NOTE: Neither the switch nor case values are wrapped in $[ ]; they can be constants, or ${var} type references only.

NOTE: AEL generates each case as a separate extension. case clauses with no terminating 'break', or 'goto', have a goto inserted, to the next clause, which creates a 'fall thru' effect.

NOTE: AEL introduces the ifTime keyword/statement, which works just like the if() statement, but the expression is a time value, exactly like that used by the application GotoIfTime(). See Asterisk cmd GotoIfTime

NOTE: The pattern statement makes sure the new extension that is created has an '_' preceding it to make sure asterisk recognizes the extension name as a pattern.

NOTE: Every character enclosed by the switch expression's parenthesis are included verbatim in the labels generated. So watch out for spaces!

NOTE: NEW: Previous to version 0.13, the random statement used the "Random()" application, which has been deprecated. It now uses the RAND() function instead, in the GotoIf application.

4.9.12 Break, Continue, and Return

Three keywords, break, continue, and return, are included in the syntax to provide flow of control to loops, and switches.

The break can be used in switches and loops, to jump to the end of the loop or switch.

The continue can be used in loops (while and for) to immediately jump to the end of the loop. In the case of a for loop, the increment and test will then be performed. In the case of the while loop, the continue will jump to the test at the top of the loop.

The return keyword will cause an immediate jump to the end of the context, or macro, and can be used anywhere.

4.9.13 goto, jump, and labels

This is an example of how to do a goto in AEL.

```
context gotoexample {
    s => {
begin:
        NoOp(Infinite Loop!  yay!);
        Wait(1);
        goto begin;    // go to label in same extension
```

```
    }
    3 => {
            goto s,begin;   // go to label in different extension
     }
     4 => {
            goto gotoexample,s,begin;  // overkill go to label in same context
     }
}

context gotoexample2 {
     s =>  {
   end:
           goto gotoexample,s,begin;   // go to label in different context
     }
}
```

You can use the special label of "1" in the goto and jump statements. It means the "first" statement in the extension. I would not advise trying to use numeric labels other than "1" in goto's or jumps, nor would I advise declaring a "1" label anywhere! As a matter of fact, it would be bad form to declare a numeric label, and it might conflict with the priority numbers used internally by asterisk.

The syntax of the jump statement is: jump extension[,priority][@context] If priority is absent, it defaults to "1". If context is not present, it is assumed to be the same as that which contains the "jump".

```
context gotoexample {
    s => {
begin:
         NoOp(Infinite Loop!  yay!);
         Wait(1);
         jump s;    // go to first extension in same extension
    }
    3 => {
            jump s,begin;   // go to label in different extension
     }
     4 => {
            jump s,begin@gotoexample;  // overkill go to label in same context
     }
}

context gotoexample2 {
     s =>  {
   end:
           jump s@gotoexample;   // go to label in different context
     }
}
```

NOTE: goto labels follow the same requirements as the Goto() application, except the last value has to be a label. If the label does not exist, you

will have run-time errors. If the label exists, but in a different extension, you have to specify both the extension name and label in the goto, as in: goto s,z; if the label is in a different context, you specify context,extension,label. There is a note about using goto's in a switch statement below...

NOTE AEL introduces the special label "1", which is the beginning context number for most extensions.

4.9.14 Macros

A macro is defined in its own block like this. The arguments to the macro are specified with the name of the macro. They are then referred to by that same name. A catch block can be specified to catch special extensions.

```
macro std-exten( ext , dev ) {
       Dial(${dev}/${ext},20);
       switch(${DIALSTATUS) {
       case BUSY:
               Voicemail(${ext},b);
               break;
       default:
               Voicemail(${ext},u);

       }
       catch a {
               VoiceMailMain(${ext});
               return;
       }
}
```

A macro is then called by preceding the macro name with an ampersand. Empty arguments can be passed simply with nothing between comments(0.11).

```
context example {
    _5XXX => &std-exten(${EXTEN}, "IAX2");
    _6XXX => &std-exten(, "IAX2");
    _7XXX => &std-exten(${EXTEN},);
    _8XXX => &std-exten(,);
}
```

4.10 Examples

```
context demo {
    s => {
         Wait(1);
         Answer();
         TIMEOUT(digit)=5;
```

```
        TIMEOUT(response)=10;
restart:
        Background(demo-congrats);
instructions:
        for (x=0; ${x} < 3; x=${x} + 1) {
             Background(demo-instruct);
             WaitExten();
        }
    }
    2 => {
         Background(demo-moreinfo);
         goto s,instructions;
    }
    3 => {
         LANGUAGE()=fr;
         goto s,restart;
    }

    500 => {
         Playback(demo-abouttotry);
         Dial(IAX2/guest@misery.digium.com);
         Playback(demo-nogo);
         goto s,instructions;
    }
    600 => {
         Playback(demo-echotest);
         Echo();
         Playback(demo-echodone);
         goto s,instructions;
    }
    # => {
hangup:
         Playback(demo-thanks);
         Hangup();
    }
    t => goto #,hangup;
    i => Playback(invalid);
}
```

4.11 Semantic Checks

AEL, after parsing, but before compiling, traverses the dialplan tree, and makes several checks:

- Macro calls to non-existent macros.
- Macro calls to contexts.
- Macro calls with argument count not matching the definition.
- application call to macro. (missing the '&')

- application calls to "GotoIf", "GotoIfTime", "while", "endwhile", "Random", and "execIf", will generate a message to consider converting the call to AEL goto, while, etc. constructs.
- goto a label in an empty extension.
- goto a non-existent label, either a within-extension, within-context, or in a different context, or in any included contexts. Will even check "sister" context references.
- All the checks done on the time values in the dial plan, are done on the time values in the ifTime() and includes times: o the time range has to have two times separated by a dash; o the times have to be in range of 0 to 24 hours. o The weekdays have to match the internal list, if they are provided; o the day of the month, if provided, must be in range of 1 to 31; o the month name or names have to match those in the internal list.
- (0.5) If an expression is wrapped in $[...], and the compiler will wrap it again, a warning is issued.
- (0.5) If an expression had operators (you know, +,-,*,/,issued. Maybe someone forgot to wrap a variable name?
- (0.12) check for duplicate context names.
- (0.12) check for abstract contexts that are not included by any context.
- (0.13) Issue a warning if a label is a numeric value.

There are a subset of checks that have been removed until the proposed AAL (Asterisk Argument Language) is developed and incorporated into Asterisk. These checks will be:

- (if the application argument analyzer is working: the presence of the 'j' option is reported as error.
- if options are specified, that are not available in an application.
- if you specify too many arguments to an application.
- a required argument is not present in an application call.

- Switch-case using "known" variables that applications set, that does not cover all the possible values. (a "default" case will solve this problem. Each "unhandled" value is listed.
- a Switch construct is used, which is uses a known variable, and the application that would set that variable is not called in the same extension. This is a warning only...
- Calls to applications not in the "applist" database (installed in `/var/lib/asterisk/applist` on most systems).
- In an assignment statement, if the assignment is to a function, the function name used is checked to see if it one of the currently known functions. A warning is issued if it is not.

4.12 Differences with the original version of AEL

1. The $[...] expressions have been enhanced to include the ==, ||, and && operators. These operators are exactly equivalent to the =, |, and & operators, respectively. Why? So the C, Java, C++ hackers feel at home here.
2. It is more free-form. The newline character means very little, and is pulled out of the white-space only for line numbers in error messages.
3. It generates more error messages – by this I mean that any difference between the input and the grammar are reported, by file, line number, and column.
4. It checks the contents of $[ ] expressions (or what will end up being $[ ] expressions!) for syntax errors. It also does matching paren/bracket counts.
5. It runs several semantic checks after the parsing is over, but before the compiling begins, see the list above.
6. It handles #include "filepath" directives. – ALMOST anywhere, in fact. You could easily include a file in a context, in an extension, or at the root level. Files can be included in files that are included in files, down to 50 levels of hierarchy...

7. Local Goto's inside Switch statements automatically have the extension of the location of the switch statement appended to them.

8. A pretty printer function is available within pbx_ael.so.

9. In the utils directory, two standalone programs are supplied for debugging AEL files. One is called "aelparse", and it reads in the `/etc/asterisk/extensions.ael` file, and shows the results of syntax and semantic checking on stdout, and also shows the results of compilation to stdout. The other is "aelparse1", which uses the original ael compiler to do the same work, reading in "`/etc/asterisk/extensions.ael`", using the original 'pbx_ael.so' instead.

10. AEL supports the "jump" statement, and the "pattern" statement in switch constructs. Hopefully these will be documented in the AEL README.

11. Added the "return" keyword, which will jump to the end of an extension/Macro.

12. Added the ifTime (<time range>|<days of week>|<days of month>|<months>) [else] construct, which executes much like an if () statement, but the decision is based on the current time, and the time spec provided in the ifTime. See the example above. (Note: all the other time-dependent Applications can be used via ifTime)

13. Added the optional time spec to the contexts in the includes construct. See examples above.

14. You don't have to wrap a single "true" statement in curly braces, as in the original AEL. An "else" is attached to the closest if. As usual, be careful about nested if statements! When in doubt, use curlies!

15. Added the syntax [regexten] [hint(channel)] to precede an extension declaration. See examples above, under "Extension". The regexten keyword will cause the priorities in the extension to begin with 2 instead of 1. The hint keyword will cause its arguments to be inserted in the extension under the hint priority. They are both optional, of course, but the order is fixed at the moment– the regexten must come before the hint, if they are both present.

16. Empty case/default/pattern statements will "fall thru" as expected. (0.6)

17. A trailing label in an extension, will automatically have a NoOp() added, to make sure the label exists in the extension on Asterisk. (0.6)

18. (0.9) the semicolon is no longer required after a closing brace! (i.e. "];" ===> "}". You can have them there if you like, but they are not necessary. Someday they may be rejected as a syntax error, maybe.

19. (0.9) the // comments are not recognized and removed in the spots where expressions are gathered, nor in application call arguments. You may have to move a comment if you get errors in existing files.

20. (0.10) the random statement has been added. Syntax: random (<expr>) <lucky-statement> [else <unlucky-statement>]. The probability of the lucky-statement getting executed is <expr>, which should evaluate to an integer between 0 and 100. If the <lucky-statement> isn't so lucky this time around, then the <unlucky-statement> gets executed, if it is present.

4.13 Hints and Bugs

The safest way to check for a null strings is to say $["${x}" = ""] The old way would do as shell scripts often do, and append something on both sides, like this: $[${x}foo = foo]. The trouble with the old way, is that, if x contains any spaces, then problems occur, usually syntax errors. It is better practice and safer wrap all such tests with double quotes! Also, there are now some functions that can be used in a variable reference, ISNULL(), and LEN(), that can be used to test for an empty string: ${ISNULL(${x})} or $[${LEN(${x})} = 0].

Assignment vs. Set(). Keep in mind that setting a variable to value can be done two different ways. If you choose say 'x=y;', keep in mind that AEL will wrap the right-hand-side with $[]. So, when compiled into extension language format, the end result will be 'Set(x=$[y])'. If you don't want this effect, then say "Set(x=y);" instead.

4.14 The Full Power of AEL

A newcomer to Asterisk will look at the above constructs and descriptions, and ask, "Where's the string manipulation functions?", "Where's all the cool operators that other languages have to offer?", etc.

The answer is that the rich capabilities of Asterisk are made available through AEL, via:

- Applications: See Asterisk - documentation of application commands
- Functions: Functions were implemented inside ${ .. } variable references, and supply many useful capabilities.
- Expressions: An expression evaluation engine handles items wrapped inside $[...]. This includes some string manipulation facilities, arithmetic expressions, etc.
- Application Gateway Interface: Asterisk can fork external processes that communicate via pipe. AGI applications can be written in any language. Very powerful applications can be added this way.
- Variables: Channels of communication have variables associated with them, and asterisk provides some global variables. These can be manipulated and/or consulted by the above mechanisms.

Chapter 5

SLA: Shared Line Appearances

5.1 Introduction

The "SLA" functionality in Asterisk is intended to allow a setup that emulates a simple key system. It uses the various abstraction layers already built into Asterisk to emulate key system functionality across various devices, including IP channels.

5.2 Configuration

5.2.1 Summary

An SLA system is built up of virtual trunks and stations mapped to real Asterisk devices. The configuration for all of this is done in three different files: extensions.conf, sla.conf, and the channel specific configuration file such as sip.conf or dahdi.conf.

5.2.2 Dialplan

The SLA implementation can automatically generate the dialplan necessary for basic operation if the "autocontext" option is set for trunks and stations in sla.conf. However, for reference, here is an automatically generated dialplan to help with custom building of the dialplan to include other features, such as voicemail (5.3.2).

However, note that there is a little bit of additional configuration needed if the trunk is an IP channel. This is discussed in the section on trunks (5.2.3).

There are extensions for incoming calls on a specific trunk, which execute the SLATrunk application, as well as incoming calls from a station, which execute SLAStation. Note that there are multiple extensions for incoming calls from a station. This is because the SLA system has to know whether the phone just went off hook, or if the user pressed a specific line button.

Also note that there is a hint for every line on every station. This lets the SLA system control each individual light on every phone to ensure that it shows the correct state of the line. The phones must subscribe to the state of each of their line appearances.

Please refer to the examples section for full dialplan samples for SLA.

5.2.3 Trunks

An SLA trunk is a mapping between a virtual trunk and a real Asterisk device. This device may be an analog FXO line, or something like a SIP trunk. A trunk must be configured in two places. First, configure the device itself in the channel specific configuration file such as dahdi.conf or sip.conf. Once the trunk is configured, then map it to an SLA trunk in sla.conf.

```
[line1]
type=trunk
device=DAHDI/1
```

Be sure to configure the trunk's context to be the same one that is set for the "autocontext" option in sla.conf if automatic dialplan configuration is used. This would be done in the regular device entry in dahdi.conf, sip.conf, etc. Note that the automatic dialplan generation creates the SLATrunk() extension at extension 's'. This is perfect for DAHDI channels that are FXO trunks, for example. However, it may not be good enough for an IP trunk, since the call coming in over the trunk may specify an actual number.

If the dialplan is being built manually, ensure that calls coming in on a trunk execute the SLATrunk() application with an argument of the trunk name, as shown in the dialplan example before.

IP trunks can be used, but they require some additional configuration to work.

For this example, let's say we have a SIP trunk called "mytrunk" that is going to be used as line4. Furthermore, when calls come in on this trunk,

they are going to say that they are calling the number "12564286000". Also, let's say that the numbers that are valid for calling out this trunk are NANP numbers, of the form _1NXXNXXXXXX.

In sip.conf, there would be an entry for [mytrunk]. For [mytrunk], set context=line4.

```
[line4]
type=trunk
device=Local/disa@line4_outbound

[line4]
exten => 12564286000,1,SLATrunk(line4)

[line4_outbound]
exten => disa,1,Disa(no-password,line4_outbound)
exten => _1NXXNXXXXX,1,Dial(SIP/${EXTEN}@mytrunk)
```

So, when a station picks up their phone and connects to line 4, they are connected to the local dialplan. The Disa application plays dialtone to the phone and collects digits until it matches an extension. In this case, once the phone dials a number like 12565551212, the call will proceed out the SIP trunk.

5.2.4 Stations

An SLA station is a mapping between a virtual station and a real Asterisk device. Currently, the only channel driver that has all of the features necessary to support an SLA environment is chan_sip. So, to configure a SIP phone to use as a station, you must configure sla.conf and sip.conf.

```
[station1]
type=station
device=SIP/station1
trunk=line1
trunk=line2
```

Here are some hints on configuring a SIP phone for use with SLA:

1. Add the SIP channel as a [station] in sla.conf.
2. Configure the phone in sip.conf. If automatic dialplan configuration was used by enabling the "autocontext" option in sla.conf, then this entry in sip.conf should have the same context setting.

3. On the phone itself, there are various things that must be configured to make everything work correctly:

   Let's say this phone is called "station1" in sla.conf, and it uses trunks named "line1" and line2".

   (a) Two line buttons must be configured to subscribe to the state of the following extensions: - station1_line1 - station1_line2

   (b) The line appearance buttons should be configured to dial the extensions that they are subscribed to when they are pressed.

   (c) If you would like the phone to automatically connect to a trunk when it is taken off hook, then the phone should be automatically configured to dial "station1" when it is taken off hook.

5.3 Configuration Examples

5.3.1 Basic SLA

This is an example of the most basic SLA setup. It uses the automatic dialplan generation so the configuration is minimal.

sla.conf:

```
[line1]
type=trunk
device=DAHDI/1
autocontext=line1

[line2]
type=trunk
device=DAHDI/2
autocontext=line2

[station](!)
type=station
trunk=line1
trunk=line2
autocontext=sla_stations

[station1](station)
device=SIP/station1

[station2](station)
device=SIP/station2

[station3](station)
device=SIP/station3
```

With this configuration, the dialplan is generated automatically. The first DAHDI channel should have its context set to "line1" and the second should be set to "line2" in dahdi.conf. In sip.conf, station1, station2, and station3 should all have their context set to "sla_stations".

For reference, here is the automatically generated dialplan for this situation:

```
[line1]
exten => s,1,SLATrunk(line1)

[line2]
exten => s,2,SLATrunk(line2)

[sla_stations]
exten => station1,1,SLAStation(station1)
exten => station1_line1,hint,SLA:station1_line1
exten => station1_line1,1,SLAStation(station1_line1)
exten => station1_line2,hint,SLA:station1_line2
exten => station1_line2,1,SLAStation(station1_line2)

exten => station2,1,SLAStation(station2)
exten => station2_line1,hint,SLA:station2_line1
exten => station2_line1,1,SLAStation(station2_line1)
exten => station2_line2,hint,SLA:station2_line2
exten => station2_line2,1,SLAStation(station2_line2)

exten => station3,1,SLAStation(station3)
exten => station3_line1,hint,SLA:station3_line1
exten => station3_line1,1,SLAStation(station3_line1)
exten => station3_line2,hint,SLA:station3_line2
exten => station3_line2,1,SLAStation(station3_line2)
```

5.3.2 SLA and Voicemail

This is an example of how you could set up a single voicemail box for the phone system. The voicemail box number used in this example is 1234, which would be configured in voicemail.conf.

For this example, assume that there are 2 trunks and 3 stations. The trunks are DAHDI/1 and DAHDI/2. The stations are SIP/station1, SIP/station2, and SIP/station3.

In dahdi.conf, channel 1 has context=line1 and channel 2 has context=line2.

In sip.conf, all three stations are configured with context=sla_stations.

When the stations pick up their phones to dial, they are allowed to dial NANP numbers for outbound calls, or 8500 for checking voicemail.

sla.conf:

```
[line1]
```

```
type=trunk
device=Local/disa@line1_outbound

[line2]
type=trunk
device=Local/disa@line2_outbound

[station](!)
type=station
trunk=line1
trunk=line2

[station1](station)
device=SIP/station1

[station2](station)
device=SIP/station2

[station3](station)
device=SIP/station3
```

extensions.conf:

```
[macro-slaline]
exten => s,1,SLATrunk(${ARG1})
exten => s,n,Goto(s-${SLATRUNK_STATUS},1)
exten => s-FAILURE,1,Voicemail(1234,u)
exten => s-UNANSWERED,1,Voicemail(1234,u)

[line1]
exten => s,1,Macro(slaline,line1)

[line2]
exten => s,2,Macro(slaline,line2)

[line1_outbound]
exten => disa,1,Disa(no-password,line1_outbound)
exten => _1NXXNXXXXXX,1,Dial(DAHDI/1/${EXTEN})
exten => 8500,1,VoicemailMain(1234)

[line2_outbound]
exten => disa,1,Disa(no-password|line2_outbound)
exten => _1NXXNXXXXXX,1,Dial(DAHDI/2/${EXTEN})
exten => 8500,1,VoicemailMain(1234)

[sla_stations]

exten => station1,1,SLAStation(station1)
exten => station1_line1,hint,SLA:station1_line1
exten => station1_line1,1,SLAStation(station1_line1)
exten => station1_line2,hint,SLA:station1_line2
exten => station1_line2,1,SLAStation(station1_line2)

exten => station2,1,SLAStation(station2)
exten => station2_line1,hint,SLA:station2_line1
exten => station2_line1,1,SLAStation(station2_line1)
```

```
exten => station2_line2,hint,SLA:station2_line2
exten => station2_line2,1,SLAStation(station2_line2)

exten => station3,1,SLAStation(station3)
exten => station3_line1,hint,SLA:station3_line1
exten => station3_line1,1,SLAStation(station3_line1)
exten => station3_line2,hint,SLA:station3_line2
exten => station3_line2,1,SLAStation(station3_line2)
```

5.4 Call Handling

5.4.1 Summary

This section is intended to describe how Asterisk handles calls inside of the SLA system so that it is clear what behavior is expected.

5.4.2 Station goes off hook (not ringing)

When a station goes off hook, it should initiate a call to Asterisk with the extension that indicates that the phone went off hook without specifying a specific line. In the examples in this document, for the station named "station1", this extension is simply named, "station1".

Asterisk will attempt to connect this station to the first available trunk that is not in use. Asterisk will check the trunks in the order that they were specified in the station entry in sla.conf. If all trunks are in use, the call will be denied.

If Asterisk is able to acquire an idle trunk for this station, then trunk is connected to the station and the station will hear dialtone. The station can then proceed to dial a number to call. As soon as a trunk is acquired, all appearances of this line on stations will show that the line is in use.

5.4.3 Station goes off hook (ringing)

When a station goes off hook while it is ringing, it should simply answer the call that had been initiated to it to make it ring. Once the station has answered, Asterisk will figure out which trunk to connect it to. It will connect it to the highest priority trunk that is currently ringing. Trunk priority is determined by the order that the trunks are listed in the station entry in sla.conf.

5.4.4 Line button on a station is pressed

When a line button is pressed on a station, the station should initiate a call to Asterisk with the extension that indicates which line button was pressed. In the examples given in this document, for a station named "station1" and a trunk named "line1", the extension would be "station1_line1".

If the specified trunk is not in use, then the station will be connected to it and will hear dialtone. All appearances of this trunk will then show that it is now in use.

If the specified trunk is on hold by this station, then this station will be reconnected to the trunk. The line appearance for this trunk on this station will now show in use. If this was the only station that had the call on hold, then all appearances of this trunk will now show that it is in use. Otherwise, all stations that are not currently connected to this trunk will show it on hold.

If the specified trunk is on hold by a different station, then this station will be connected to the trunk only if the trunk itself and the station(s) that have it on hold do not have private hold enabled. If connected, the appeareance of this trunk on this station will then show in use. All stations that are not currently connected to this trunk will show it on hold.

Chapter 6

Channel Drivers

6.1 IAX2

6.1.1 Introduction

This section is intended as an introduction to the Inter-Asterisk eXchange v2 (or simply IAX2) protocol. It provides both a theoretical background and practical information on its use.

6.1.2 Why IAX2?

The first question most people are thinking at this point is "Why do you need another VoIP protocol? Why didn't you just use SIP or H.323?"

Well, the answer is a fairly complicated one, but in a nutshell it's like this... Asterisk is intended as a very flexible and powerful communications tool. As such, the primary feature we need from a VoIP protocol is the ability to meet our own goals with Asterisk, and one with enough flexibility that we could use it as a kind of laboratory for inventing and implementing new concepts in the field. Neither H.323 or SIP fit the roles we needed, so we developed our own protocol, which, while not standards based, provides a number of advantages over both SIP and H.323, some of which are:

- Interoperability with NAT/PAT/Masquerade firewalls
  - IAX seamlessly interoperates through all sorts of NAT and PAT and other firewalls, including the ability to place and receive calls,

and transfer calls to other stations.

- High performance, low overhead protocol
  - When running on low-bandwidth connections, or when running large numbers of calls, optimized bandwidth utilization is imperative. IAX uses only 4 bytes of overhead
- Internationalization support
  - IAX transmits language information, so that remote PBX content can be delivered in the native language of the calling party.
- Remote dialplan polling
  - IAX allows a PBX or IP phone to poll the availability of a number from a remote server. This allows PBX dialplans to be centralized.
- Flexible authentication
  - IAX supports cleartext, md5, and RSA authentication, providing flexible security models for outgoing calls and registration services.
- Multimedia protocol
  - IAX supports the transmission of voice, video, images, text, HTML, DTMF, and URL's. Voice menus can be presented in both audibly and visually.
- Call statistic gathering
  - IAX gathers statistics about network performance (including latency and jitter, as well as providing end-to-end latency measurement.
- Call parameter communication
  - Caller*ID, requested extension, requested context, etc are all communicated through the call.
- Single socket design

– IAX's single socket design allows up to 32768 calls to be multiplexed.

While we value the importance of standards based (i.e. SIP) call handling, hopefully this will provide a reasonable explanation of why we developed IAX rather than starting with SIP.

6.1.3 Configuration

For examples of a configuration, please see the iax.conf.sample in your the /configs directory of you source code distribution.

6.1.4 IAX2 Jitterbuffer

The new jitterbuffer

You must add "jitterbuffer=yes" to either the [general] part of iax.conf, or to a peer or a user. (just like the old jitterbuffer). Also, you can set "maxjitterbuffer=n", which puts a hard-limit on the size of the jitterbuffer of "n milliseconds". It is not necessary to have the new jitterbuffer on both sides of a call; it works on the receive side only.

PLC

The new jitterbuffer detects packet loss. PLC is done to try to recreate these lost packets in the codec decoding stage, as the encoded audio is translated to slinear. PLC is also used to mask jitterbuffer growth.

This facility is enabled by default in iLBC and speex, as it has no additional cost. This facility can be enabled in adpcm, alaw, g726, gsm, lpc10, and ulaw by setting genericplc => true in the [plc] section of codecs.conf.

Trunktimestamps

To use this, both sides must be using Asterisk v1.2 or later. Setting "trunktimestamps=yes" in iax.conf will cause your box to send 16-bit timestamps for each trunked frame inside of a trunk frame. This will enable you to use jitterbuffer for an IAX2 trunk, something that was not possible in the old architecture.

The other side must also support this functionality, or else, well, bad things will happen. If you don't use trunktimestamps, there's lots of ways the jitterbuffer can get confused because timestamps aren't necessarily sent through the trunk correctly.

Communication with Asterisk v1.0.x systems

You can set up communication with v1.0.x systems with the new jitterbuffer, but you can't use trunks with trunktimestamps in this communication.

If you are connecting to an Asterisk server with earlier versions of the software (1.0.x), do not enable both jitterbuffer and trunking for the involved peers/users in order to be able to communicate. Earlier systems will not support trunktimestamps.

You may also compile chan_iax2.c without the new jitterbuffer, enabling the old backwards compatible architecture. Look in the source code for instructions.

Testing and monitoring

You can test the effectiveness of PLC and the new jitterbuffer's detection of loss by using the new CLI command "iax2 test losspct <n>". This will simulate n percent packet loss coming _in_ to chan_iax2. You should find that with PLC and the new JB, 10 percent packet loss should lead to just a tiny amount of distortion, while without PLC, it would lead to silent gaps in your audio.

"iax2 show netstats" shows you statistics for each iax2 call you have up. The columns are "RTT" which is the round-trip time for the last PING, and then a bunch of s tats for both the local side (what you're receiving), and the remote side (what the other end is telling us they are seeing). The remote stats may not be complete if the remote end isn't using the new jitterbuffer.

The stats shown are:

- Jit: The jitter we have measured (milliseconds)
- Del: The maximum delay imposed by the jitterbuffer (milliseconds)
- Lost: The number of packets we've detected as lost.
- %: The percentage of packets we've detected as lost recently.

- Drop: The number of packets we've purposely dropped (to lower latency).
- OOO: The number of packets we've received out-of-order
- Kpkts: The number of packets we've received / 1000.

Reporting problems

There's a couple of things that can make calls sound bad using the jitterbuffer:

1. The JB and PLC can make your calls sound better, but they can't fix everything. If you lost 10 frames in a row, it can't possibly fix that. It really can't help much more than one or two consecutive frames.

2. Bad timestamps: If whatever is generating timestamps to be sent to you generates nonsensical timestamps, it can confuse the jitterbuffer. In particular, discontinuities in timestamps will really upset it: Things like timestamps sequences which go 0, 20, 40, 60, 80, 34000, 34020, 34040, 34060... It's going to think you've got about 34 seconds of jitter in this case, etc.. The right solution to this is to find out what's causing the sender to send us such nonsense, and fix that. But we should also figure out how to make the receiver more robust in cases like this.

   chan_iax2 will actually help fix this a bit if it's more than 3 seconds or so, but at some point we should try to think of a better way to detect this kind of thing and resynchronize.

   Different clock rates are handled very gracefully though; it will actually deal with a sender sending 20% faster or slower than you expect just fine.

3. Really strange network delays: If your network "pauses" for like 5 seconds, and then when it restarts, you are sent some packets that are 5 seconds old, we are going to see that as a lot of jitter. We already throw away up to the worst 20 frames like this, though, and the "maxjitterbuffer" parameter should put a limit on what we do in this case.

6.2 mISDN

6.2.1 Introduction

This package contains the mISDN Channel Driver for the Asterisk PBX. It supports every mISDN Hardware and provides an interface for Asterisk.

6.2.2 Features

- NT and TE mode
- PP and PMP mode
- BRI and PRI (with BNE1 and BN2E1 Cards)
- Hardware bridging
- DTMF detection in HW+mISDNdsp
- Display messages on phones (on those that support it)
- app_SendText
- HOLD/RETRIEVE/TRANSFER on ISDN phones :)
- Allow/restrict user number presentation
- Volume control
- Crypting with mISDNdsp (Blowfish)
- Data (HDLC) callthrough
- Data calling (with app_ptyfork +pppd)
- Echo cancellation
- Call deflection
- Some others

6.2.3 Fast Installation Guide

It is easy to install mISDN and mISDNuser. This can be done by:

- You can download latest stable releases from `http://www.misdn.org/downloads/`
- Just fetch the newest head of the GIT (mISDN project moved from CVS) In details this process described here: `http://www.misdn.org/index.php/GIT`

then compile and install both with:

```
cd mISDN ;
make && make install
```

(you will need at least your kernel headers to compile mISDN).

```
cd mISDNuser ;
make && make install
```

Now you can compile chan_misdn, just by making Asterisk:

```
cd asterisk ;
./configure && make && make install
```

That's all!

Follow the instructions in the mISDN Package for how to load the Kernel Modules. Also install process described in `http://www.misdn.org/index.php/Installing_mISDN`

6.2.4 Pre-Requisites

To compile and install this driver, you'll need at least one mISDN Driver and the mISDNuser package. Chan_misdn works with both, the current release version and the development (svn trunk) version of Asterisk.

You should use Kernels >= 2.6.9

6.2.5 Configuration

First of all you must configure the mISDN drivers, please follow the instructions in the mISDN package to do that, the main config file and config script is:

```
/etc/init.d/misdn-init  and
/etc/misdn-init.conf
```

Now you will want to configure the misdn.conf file which resides in the Asterisk config directory (normally /etc/asterisk).

misdn.conf: [general]

The misdn.conf file contains a "general" subsection, and user subsections which contain misdn port settings and different Asterisk contexts.

In the general subsection you can set options that are not directly port related. There is for example the very important debug variable which you can set from the Asterisk cli (command line interface) or in this configuration file, bigger numbers will lead to more debug output. There's also a trace file option, which takes a path+filename where debug output is written to.

misdn.conf: [default] subsection

The default subsection is another special subsection which can contain all the options available in the user/port subsections. The user/port subsections inherit their parameters from the default subsection.

misdn.conf: user/port subsections

The user subsections have names which are unequal to "general". Those subsections contain the ports variable which mean the mISDN Ports. Here you can add multiple ports, comma separated.

Especially for TE-Mode Ports there is a msns option. This option tells the chan_misdn driver to listen for incoming calls with the given msns, you can insert a '*' as single msn, which leads to getting every incoming call. If you want to share on PMP TE S0 with Asterisk and a phone or ISDN card you should insert here the msns which you assign to Asterisk. Finally a context variable resides in the user subsections, which tells chan_misdn where to send incoming calls to in the Asterisk dial plan (extension.conf).

Dial and Options String

The dial string of chan_misdn got more complex, because we added more features, so the generic dial string looks like:

```
mISDN/<port>[:bchannel]|g:<group>/<extension>[/<OPTIONSSTRING>]

The Optionsstring looks Like:
:<optchar><optarg>:<optchar><optarg>...

the ":" character is the delimiter.

The available options are:
```

```
 a - Have Asterisk detect DTMF tones on called channel
 c - Make crypted outgoing call, optarg is keyindex
 d - Send display text to called phone, text is the optarg
 e - Perform echo cancelation on this channel,
     takes taps as optarg (32,64,128,256)
e! - Disable echo cancelation on this channel
 f - Enable fax detection
 h - Make digital outgoing call
h1 - Make HDLC mode digital outgoing call
 i - Ignore detected DTMF tones, don't signal them to Asterisk,
     they will be transported inband.
jb - Set jitter buffer length, optarg is length
jt - Set jitter buffer upper threshold, optarg is threshold
jn - Disable jitter buffer
 n - Disable mISDN DSP on channel.
     Disables: echo cancel, DTMF detection, and volume control.
 p - Caller ID presentation,
     optarg is either 'allowed' or 'restricted'
 s - Send Non-inband DTMF as inband
vr - Rx gain control, optarg is gain
vt - Tx gain control, optarg is gain
```

chan_misdn registers a new dial plan application "misdn_set_opt" when loaded. This application takes the Optionsstring as argument. The Syntax is:

```
misdn_set_opt(<OPTIONSSTRING>)
```

When you set options in the dialstring, the options are set in the external channel. When you set options with misdn_set_opt, they are set in the current incoming channel. So if you like to use static encryption, the scenario looks as follows:

```
Phone1 --> * Box 1 --> PSTN_TE
PSTN_TE --> * Box 2 --> Phone2
```

The encryption must be done on the PSTN sides, so the dialplan on the boxes are:

```
* Box 1:
exten => _${CRYPT_PREFIX}X.,1,Dial(mISDN/g:outbound/:c1)

* Box 2:
exten => ${CRYPT_MSN},1,misdn_set_opt(:c1)
exten => ${CRYPT_MSN},2,dial(${PHONE2})
```

6.2.6 mISDN CLI commands

At the Asterisk cli you can try to type in:

```
misdn <tab> <tab>
```

Now you should see the misdn cli commands:

```
- clean
  -> pid    (cleans a broken call, use with care, leads often
       to a segmentation fault)
- send
  -> display  (sends a Text Message to a Asterisk channel,
       this channel must be an misdn channel)
- set
  -> debug  (sets debug level)
- show
  -> config (shows the configuration options)
  -> channels (shows the current active misdn channels)
  -> channel  (shows details about the given misdn channels)
  -> stacks (shows the current ports, their protocols and states)
  -> fullstacks (shows the current active and inactive misdn channels)

- restart
  -> port   (restarts given port (L2 Restart) )

- reload    (reloads misdn.conf)
```

You can only use "misdn send display" when an Asterisk channel is created and isdn is in the correct state. "correct state" means that you have established a call to another phone (must not be isdn though).

Then you use it like this:

misdn send display mISDN/1/101 "Hello World!"

where 1 is the Port of the Card where the phone is plugged in, and 101 is the msn (callerid) of the Phone to send the text to.

6.2.7 mISDN Variables

mISDN Exports/Imports a few Variables:

```
- MISDN_ADDRESS_COMPLETE :  Is either set to 1 from the Provider, or you
       can set it to 1 to force a sending complete.
```

6.2.8 Debugging and sending bug reports

If you encounter problems, you should set up the debugging flag, usually debug=2 should be enough. The messages are divided into Asterisk and mISDN parts. mISDN Debug messages begin with an 'I', Asterisk messages begin with an '*', the rest is clear I think.

Please take a trace of the problem and open a report in the Asterisk issue tracker at `http://bugs.digium.com` in the "channel drivers" project, "chan_misdn" category. Read the bug guidelines to make sure you provide all the information needed.

6.2.9 Examples

Here are some examples of how to use chan_misdn in the dialplan (extensions.conf):

```
[globals]
OUT_PORT=1 ; The physical Port of the Card
OUT_GROUP=ExternE1 ; The Group of Ports defined in misdn.conf

[misdnIn]
exten => _X.,1,Dial(mISDN/${OUT_PORT}/${EXTEN})
exten => _0X.,1,Dial(mISDN/g:${OUT_GROUP}/${EXTEN:1})
exten => _1X.,1,Dial(mISDN/g:${OUT_GROUP}/${EXTEN:1}/:dHello)
exten => _1X.,1,Dial(mISDN/g:${OUT_GROUP}/${EXTEN:1}/:dHello Test:n)
```

On the last line, you will notice the last argument (Hello); this is sent as Display Message to the Phone.

6.2.10 Known Problems

Q: I cannot hear any tone after a successful CONNECT to the other end.

A: You forgot to load mISDNdsp, which is now needed by chan_misdn for switching and DTMF tone detection.

6.3 Local

6.3.1 Introduction

In Asterisk, Local channels are a method used to treat an extension in the dialplan as if it were an external device. In essense, Asterisk will send the

call back into the dialplan as the destination of the call, versus sending the call to a device.

Two of the most common areas where Local channels are used include members configured for queues, and in use with callfiles. There are also other uses where you want to ring two destinations, but with different information, such as different callerID for each outgoing request.

6.3.2 Examples

Local channels are best demonstrated through the use of an example. Our first example isn't terribly useful, but will demonstrate how Local channels can execute dialplan logic by dialing from the Dial() application.

6.3.3 Trivial Local channel example

In our dialplan (extensions.conf), we can Dial() another part of the dialplan through the use Local channels. To do this, we can use the following dialplan:

```
[devices]
exten => 201,1,Verbose(2,Dial another part of the dialplan via the Local chan)
exten => 201,n,Verbose(2,Outside channel:  ${CHANNEL})
exten => 201,n,Dial(Local/201@extensions)
exten => 201,n,Hangup()

[extensions]
exten => 201,1,Verbose(2,Made it to the Local channel)
exten => 201,n,Verbose(2,Inside channel:  ${CHANNEL})
exten => 201,n,Dial(SIP/some-named-extension,30)
exten => 201,n,Hangup()
```

The output of the dialplan would look something like the following. The output has been broken up with some commentary to explain what we're looking at.

```
-- Executing [201@devices:1] Verbose("SIP/my_desk_phone-00000014", "2,Dial another part of the dialplan via the
            Local chan") in new stack
== Dial another part of the dialplan via the Local chan
```

We dial extension 201 from SIP/my_desk_phone which has entered the [devices] context. The first line simply outputs some information via the Verbose() application.

```
-- Executing [201@devices:2] Verbose("SIP/my_desk_phone-00000014",
                  "2,Outside channel:  SIP/my_desk_phone-00000014") in new stack
== Outside channel:  SIP/my_desk_phone-00000014
```

The next line is another Verbose() application statement that tells us our current channel name. We can see that the channel executing the current dialplan is a desk phone (aptly named 'my_desk_phone').

```
-- Executing [201@devices:3] Dial("SIP/my_desk_phone-00000014", "Local/201@extensions") in new stack
-- Called 201@extensions
```

Now the third step in our dialplan executes the Dial() application which calls extension 201 in the [extensions] context of our dialplan. There is no requirement that we use the same extension number – we could have just as easily used a named extension, or some other number. Remember that we're dialing another channel, but instead of dialing a device, we're "dialing" another part of the dialplan.

```
-- Executing [201@extensions:1] Verbose("Local/201@extensions-7cf4;2", "2,Made it to the Local
            channel") in new stack
== Made it to the Local channel
```

Now we've verified we've dialed another part of the dialplan. We can see the channel executing the dialplan has changed to Local/201@extensions-7cf4;2. The part '-7cf4;2' is just the unique identifier, and will be different for you.

```
-- Executing [201@extensions:2] Verbose("Local/201@extensions-7cf4;2", "2,Inside channel:
            Local/201@extensions-7cf4;2") in new stack
== Inside channel:  Local/201@extensions-7cf4;2
```

Here we use the Verbose() application to see what our current channel name is. As you can see the current channel is a Local channel which we created from our SIP channel.

```
-- Executing [201@extensions:3] Dial("Local/201@extensions-7cf4;2", "SIP/some-named-extension,30") in new stack
```

And from here, we're using another Dial() application to call a SIP device configured in sip.conf as [some-named-extension].

Now that we understand a simple example of calling the Local channel, let's expand upon this example by using Local channels to call two devices at the same time, but delay calling one of the devices.

6.3.4 Delay dialing devices

Lets say when someone calls extension 201, we want to ring both the desk phone and their cellphone at the same time, but we want to wait about 6 seconds to start dialing the cellphone. This is useful in a situation when someone might be sitting at their desk, but don't want both devices ringing at the same time, but also doesn't want to wait for the full ring cycle to execute on their desk phone before rolling over to their cellphone.

The dialplan for this would look something like the following:

```
[devices]
exten => 201,1,Verbose(2,Call desk phone and cellphone but with delay)
exten => 201,n,Dial(Local/deskphone-201@extensions&Local/cellphone-201@extensions,30)
exten => 201,n,Voicemail(201@default,${IF($[${DIALSTATUS} = BUSY]?b:u)})
exten => 201,n,Hangup()

[extensions]
; Dial the desk phone
exten => deskphone-201,1,Verbose(2,Dialing desk phone of extension 201)
exten => deskphone-201,n,Dial(SIP/0004f2040001)  ; SIP device with MAC address
                                                 ; of 0004f2040001

; Dial the cellphone
exten => cellphone-201,1,Verbose(2,Dialing cellphone of extension 201)
exten => cellphone-201,n,Verbose(2,-- Waiting 6 seconds before dialing)
exten => cellphone-201,n,Wait(6)
exten => cellphone-201,n,Dial(DAHDI/g0/14165551212)
```

When someone dials extension 201 in the [devices] context, it will execute the Dial() application, and call two Local channels at the same time:

- Local/deskphone-201@extensions
- Local/cellphone-201@extensions

It will then ring both of those extensions for 30 seconds before rolling over to the Voicemail() application and playing the appropriate voicemail recording depending on whether the ${DIALSTATUS} variable returned BUSY or not.

When reaching the deskphone-201 extension, we execute the Dial() application which calls the SIP device configured as '0004f204001' (the MAC address of the device). When reaching the cellphone-201 extension, we dial the cellphone via the DAHDI channel using group zero (g0) and dialing phone number 1-416-555-1212.

6.3.5 Dialing destinations with different information

With Asterisk, we can place a call to multiple destinations by separating the technology/destination pair with an ampersand (&). For example, the following Dial() line would ring two separate destinations for 30 seconds:

```
exten => 201,1,Dial(SIP/0004f2040001&DAHDI/g0/14165551212,30)
```

That line would dial both the SIP/0004f2040001 device (likely a SIP device on the network) and dial the phone number 1-416-555-1212 via a DAHDI interface. In our example though, we would be sending the same callerID information to both end points, but perhaps we want to send a different callerID to one of the destinations?

We can send different callerIDs to each of the destinations if we want by using the Local channel. The following example shows how this is possible because we would Dial() two different Local channels from our top level Dial(), and that would then execute some dialplan before sending the call off to the final destinations.

```
[devices]
exten => 201,1,NoOp()
exten => 201,n,Dial(Local/201@internal&Local/201@external,30)
exten => 201,n,Voicemail(201@default,${IF($[${DIALSTATUS} = BUSY]?b:u)})
exten => 201,n,Hangup()

[internal]
exten => 201,1,Verbose(2,Placing internal call for extension 201)
exten => 201,n,Set(CALLERID(name)=From Sales)
exten => 201,n,Dial(SIP/0004f2040001,30)

[external]
exten => 201,1,Verbose(2,Placing external call for extension 201)
exten => 201,n,Set(CALLERID(name)=Acme Cleaning)
exten => 201,n,Dial(DAHDI/g0/14165551212)
```

With the dialplan above, we've sent two different callerIDs to the destinations:

- "From Sales" was sent to the local device SIP/0004f2040001
- "Acme Cleaning" was sent to the remote number 1-416-555-1212 via DAHDI

Because each of the channels is independent from the other, you could perform any other call manipulation you need. Perhaps the 1-416-555-1212

number is a cell phone and you know you can only ring that device for 18 seconds before the voicemail would pick up. You could then limit the length of time the external number is dialed, but still allow the internal device to be dialed for a longer period of time.

6.3.6 Using callfiles and Local channels

Another example is to use callfiles and Local channels so that you can execute some dialplan prior to performing a Dial(). We'll construct a callfile which will then utilize a Local channel to lookup a bit of information in the AstDB and then place a call via the channel configured in the AstDB.

First, lets construct our callfile that will use the Local channel to do some lookups prior to placing our call. More information on constructing callfiles is located in the doc/callfiles.txt file of your Asterisk source.

Our callfile will simply look like the following:

```
Channel: Local/201@devices
Application: Playback
Data: silence/1&tt-weasels
```

Add the callfile information to a file such as 'callfile.new' or some other appropriately named file.

Our dialplan will perform a lookup in the AstDB to determine which device to call, and will then call the device, and upon answer, Playback() the silence/1 (1 second of silence) and the tt-weasels sound files.

Before looking at our dialplan, lets put some data into AstDB that we can then lookup from the dialplan. From the Asterisk CLI, run the following command:

```
*CLI> database put phones 201/device SIP/0004f2040001
```

We've now put the device destination (SIP/0004f2040001) into the 201/device key within the phones family. This will allow us to lookup the device location for extension 201 from the database.

We can then verify our entry in the database using the 'database show' CLI command:

```
*CLI> database show
/phones/201/device                                : SIP/0004f2040001
```

Now lets create the dialplan that will allow us to call SIP/0004f2040001 when we request extension 201 from the [extensions] context via our Local channel.

```
[devices]
exten => 201,1,NoOp()
exten => 201,n,Set(DEVICE=${DB(phones/${EXTEN}/device)})
exten => 201,n,GotoIf($[${ISNULL(${DEVICE})}]?hangup) ; if nothing returned,
                                                      ; then hangup
exten => 201,n,Dial(${DEVICE},30)
exten => 201,n(hangup(),Hangup()
```

Then, we can perform a call to our device using the callfile by moving it into the /var/spool/asterisk/outgoing/ directory.

```
# mv callfile.new /var/spool/asterisks/outgoing
```

Then after a moment, you should see output on your console similar to the following, and your device ringing. Information about what is going on during the output has also been added throughout.

```
  -- Attempting call on Local/201@devices for application Playback(silence/1&tt-weasels) (Retry 1)
```

You'll see the line above as soon as Asterisk gets the request from the callfile.

```
-- Executing [201@devices:1] NoOp("Local/201@devices-ecf0;2", "") in new stack
-- Executing [201@devices:2] Set("Local/201@devices-ecf0;2", "DEVICE=SIP/0004f2040001") in new stack
```

This is where we performed our lookup in the AstDB. The value of SIP/0004f2040001 was then returned and saved to the DEVICE channel variable.

```
  -- Executing [201@devices:3] GotoIf("Local/201@devices-ecf0;2", "0?hangup") in new stack
```

We perform a check to make sure ${DEVICE} isn't NULL. If it is, we'll just hangup here.

```
  -- Executing [201@devices:4] Dial("Local/201@devices-ecf0;2", "SIP/0004f2040001,30") in new stack
  -- Called 000f2040001
  -- SIP/0004f2040001-00000022 is ringing
```

Now we call our device SIP/0004f2040001 from the Local channel.

```
 -- SIP/0004f2040001-00000022 answered Local/201@devices-ecf0;2
```

We answer the call.

```
> Channel Local/201@devices-ecf0;1 was answered.
> Launching Playback(silence/1&tt-weasels) on Local/201@devices-ecf0;1
```

We then start playing back the files.

```
  -- <Local/201@devices-ecf0;1> Playing 'silence/1.slin' (language 'en')
== Spawn extension (devices, 201, 4) exited non-zero on 'Local/201@devices-ecf0;2'
```

At this point we now see the Local channel has been optimized out of the call path. This is important as we'll see in examples later. By default, the Local channel will try to optimize itself out of the call path as soon as it can. Now that the call has been established and audio is flowing, it gets out of the way.

```
   -- <SIP/0004f2040001-00000022> Playing 'tt-weasels.ulaw' (language 'en')
[Mar  1 13:35:23] NOTICE[16814]: pbx_spool.c:349 attempt_thread: Call completed to Local/201@devices
```

We can now see the tt-weasels file is played directly to the destination (instead of through the Local channel which was optimized out of the call path) and then a NOTICE stating the call was completed.

6.3.7 Understanding When To Use /n

Lets take a look at an example that demonstrates when the use of the /n directive is necessary. If we spawn a Local channel which does a Dial() to a SIP channel, but we use the L() option (which is used to limit the amount of time a call can be active, along with warning tones when the time is nearly up), it will be associated with the Local channel, which is then optimized out of the call path, and thus won't perform as expected.

This following dialplan will not perform as expected.

[services] exten =¿ 2,1,Dial(SIP/PHONE_B,,L(60000:45000:15000))

[internal] exten =¿ 4,1,Dial(Local/2@services);

By default, the Local channel will try to optimize itself out of the call path. This means that once the Local channel has established the call between the destination and Asterisk, the Local channel will get out of the way and let Asterisk and the end point talk directly, instead of flowing through the Local channel.

This can have some adverse effects when you're expecting information to be available during the call that gets associated with the Local channel.

When the Local channel is optimized out of the call path, any Dial() flags, or channel variables associated with the Local channel are also destroyed and are no longer available to Asterisk.

We can force the Local channel to remain in the call path by utilizing the /n directive. By adding /n to the end of the channel definition, we can keep the Local channel in the call path, along with any channel variables, or other channel specific information.

In order to make this behave as we expect (limiting the call), we would change:

```
[internal]
exten => 4,1,Dial(Local/2@services)
```

...into the following:

```
[internal]
exten => 4,1,Dial(Local/2@services/n)
```

By adding /n to the end, our Local channel will now stay in the call path and not go away.

Why does adding the /n option all of a suddon make the 'L' option work? First we need to show an overview of the call flow that doesn't work properly, and discuss the information associated with the channels:

1. SIP device PHONE_A calls Asterisk via a SIP INVITE
2. Asterisk accepts the INVITE and then starts processing dialplan logic in the [internal] context
3. Our dialplan calls Dial(Local/2@services) – notice no /n
4. The Local channel then executes dialplan at extension 2 within the [services] context
5. Extension 2 within [services] then performs Dial() to PHONE_B with the line: Dial(SIP/PHONE_B,,L(60000:45000:15000))
6. SIP/PHONE_B then answers the call
7. Even though the L option was given when dialing the SIP device, the L information is stored in the channel that is doing the Dial() which is the Local channel, and not the endpoint SIP channel.

8. The Local channel in the middle, containing the information for tracking the time allowance of the call, is then optimized out of the call path, losing all information about when to terminate the call.

9. SIP/PHONE_A and SIP/PHONE_B then continue talking indefinitely.

Now, if we were to add /n to our dialplan at step three (3) then we would force the Local channel to stay in the call path, and the L() option associated with the Dial() from the Local channel would remain, and our warning sounds and timing would work as expected.

There are two workarounds for the above described scenario:

1. Use what we just described, Dial(Local/2@services/n) to cause the Local channel to remain in the call path so that the L() option used inside the Local channel is not discarded when optimization is performed.

2. Place the L() option at the outermost part of the path so that when the middle is optimized out of the call path, the information required to make L() work is associated with the outside channel. The L information will then be stored on the calling channel, which is PHONE_A. For example:

```
[services]
exten => 2,1,Dial(SIP/PHONE_B)

[internal]
exten => 4,1,Dial(Local/2@services,,L(60000:45000:15000));
```

6.3.8 Local channel modifiers

There are additional modifiers for the Local channel as well. They include:

- 'n' – Adding "/n" at the end of the string will make the Local channel not do a native transfer (the "n" stands for "n"o release) upon the remote end answering the line. This is an esoteric, but important feature if you expect the Local channel to handle calls exactly like a normal channel. If you do not have the "no release" feature set, then as soon as the destination (inside of the Local channel) answers the line and one audio frame passes, the variables and dial plan will revert

back to that of the original call, and the Local channel will become a zombie and be removed from the active channels list. This is desirable in some circumstances, but can result in unexpected dialplan behavior if you are doing fancy things with variables in your call handling.

- 'j' – Adding "/j" at the end of the string allows you to use the generic jitterbuffer on incoming calls going to Asterisk applications. For example, this would allow you to use a jitterbuffer for an incoming SIP call to Voicemail by putting a Local channel in the middle. The 'j' option must be used in conjunction with the 'n' option to make sure that the Local channel does not get optimized out of the call.

  This option is available starting in the Asterisk 1.6.0 branch.

- 'm' – Using the "/m" option will cause the Local channel to forward music on hold (MoH) start and stop requests. Normally the Local channel acts on them and it is started or stopped on the Local channel itself. This options allows those requests to be forwarded through the Local channel.

  This option is available starting in the Asterisk 1.4 branch.

- 'b' – The "/b" option causes the Local channel to return the actual channel that is behind it when queried. This is useful for transfer scenarios as the actual channel will be transferred, not the Local channel.

  This option is available starting in the Asterisk 1.6.0 branch.

Chapter 7

Distributed Universal Number Discovery (DUNDi)

7.1 Introduction

`http://www.dundi.com`

Mark Spencer, Digium, Inc.

DUNDi is essentially a trusted, peer-to-peer system for being able to call any phone number from the Internet. DUNDi works by creating a network of nodes called the "DUNDi E.164 Trust Group" which are bound by a common peering agreement known as the General Peering Agreement or GPA. The GPA legally binds the members of the Trust Group to provide good-faith accurate information to the other nodes on the network, and provides standards by which the community can insure the integrity of the information on the nodes themselves. Unlike ENUM or similar systems, DUNDi is explicitly designed to preclude any necessity for a single centralized system which could be a source of fees, regulation, etc.

Much less dramatically, DUNDi can also be used within a private enterprise to share a dialplan efficiently between multiple nodes, without incurring a risk of a single point of failure. In this way, administrators can locally add extensions which become immediately available to the other nodes in the system.

For more information visit `http://www.dundi.com`

7.2 DUNDIQUERY and DUNDIRESULT

The DUNDIQUERY and DUNDIRESULT dialplan functions will let you initiate a DUNDi query from the dialplan, see how many results there are, and access each one. Here is some example usage:

```
exten => 1,1,Set(ID=${DUNDIQUERY(1,dundi_test,b)})
exten => 1,n,Set(NUM=${DUNDIRESULT(${ID},getnum)})
exten => 1,n,NoOp(There are ${NUM} results)
exten => 1,n,Set(X=1)
exten => 1,n,While($[${X} <= ${NUM}])
exten => 1,n,NoOp(Result ${X} is ${DUNDIRESULT(${ID},${X})})
exten => 1,n,Set(X=$[${X} + 1])
exten => 1,n,EndWhile
```

7.3 Peering Agreement

DIGIUM GENERAL PEERING AGREEMENT (TM)
Version 1.0.0, September 2004

445 Jan Davis Drive, Huntsville, AL 35806 USA

DIGIUM GENERAL PEERING AGREEMENT

PREAMBLE

For most of the history of telecommunications, the power of being able to locate and communicate with another person in a system, be it across a hall or around the world, has always centered around a centralized authority -- from a local PBX administrator to regional and national RBOCs, generally requiring fees, taxes or regulation. By contrast, DUNDi is a technology developed to provide users the freedom to communicate with each other without the necessity of any centralized

authority. This General Peering Agreement ("GPA") is used by individual parties (each, a "Participant") to allow them to build the E164 trust group for the DUNDi protocol.

To protect the usefulness of the E164 trust group for those who use it, while keeping the system wholly decentralized, it is necessary to replace many of the responsibilities generally afforded to a company or government agency, with a set of responsibilities implemented by the parties who use the system, themselves. It is the goal of this document to provide all the protections necessary to keep the DUNDi E164 trust group useful and reliable.

The Participants wish to protect competition, promote innovation and value added services and make this service valuable both commercially and non-commercially. To that end, this GPA provides special terms and conditions outlining some permissible and non-permissible revenue sources.

This GPA is independent of any software license or other license agreement for a program or technology employing the DUNDi protocol. For example, the implementation of DUNDi used by Asterisk is covered under a separate license. Each Participant is responsible for compliance with any licenses or other agreements governing use of such program or technology that they use to peer.

You do not have to execute this GPA to use a program or technology employing the DUNDi protocol, however if you do not execute this GPA, you will not be able to peer using DUNDi and the E164 context with anyone who is a member of the trust group by virtue of their having executed this GPA with another member.

The parties to this GPA agree as follows:

0. DEFINITIONS. As used herein, certain terms shall be defined as follows:

(a) The term "DUNDi" means the DUNDi protocol as published by Digium, Inc. or its successor in interest with respect to the

DUNDi protocol specification.

(b) The terms "E.164" and "E164" mean ITU-T specification E.164 as published by the International Telecommunications Union (ITU) in May, 1997.

(c) The term "Service" refers to any communication facility (e.g., telephone, fax, modem, etc.), identified by an E.164-compatible number, and assigned by the appropriate authority in that jurisdiction.

(d) The term "Egress Gateway" refers an Internet facility that provides a communications path to a Service or Services that may not be directly addressable via the Internet.

(e) The term "Route" refers to an Internet address, policies, and other characteristics defined by the DUNDi protocol and associated with the Service, or the Egress Gateway which provides access to the specified Service.

(f) The term "Propagate" means to accept or transmit Service and/or Egress Gateway Routes only using the DUNDi protocol and the DUNDi context "e164" without regard to case, and does not apply to the exchange of information using any other protocol or context.

(g) The term "Peering System" means the network of systems that Propagate Routes.

(h) The term "Subscriber" means the owner of, or someone who contracts to receive, the services identified by an E.164 number.

(i) The term "Authorizing Individual" means the Subscriber to a number who has authorized a Participant to provide Routes regarding their services via this Peering System.

(j) The term "Route Authority" refers to a Participant that provides

an original source of said Route within the Peering System. Routes are propagated from the Route Authorities through the Peering System and may be cached at intermediate points. There may be multiple Route Authorities for any Service.

(k) The term "Participant" (introduced above) refers to any member of the Peering System.

(l) The term "Service Provider" refers to the carrier (e.g., exchange carrier, Internet Telephony Service Provider, or other reseller) that provides communication facilities for a particular Service to a Subscriber, Customer or other End User.

(m) The term "Weight" refers to a numeric quality assigned to a Route as per the DUNDi protocol specification. The current Weight definitions are shown in Exhibit A.

1. PEERING. The undersigned Participants agree to Propagate Routes with each other and any other member of the Peering System and further agree not to Propagate DUNDi Routes with a third party unless they have first have executed this GPA (in its unmodified form) with such third party. The Participants further agree only to Propagate Routes with Participants whom they reasonably believe to be honoring the terms of the GPA. Participants may not insert, remove, amend, or otherwise modify any of the terms of the GPA.

2. ACCEPTABLE USE POLICY. The DUNDi protocol contains information that reflect a Subscriber's or Egress Gateway's decisions to receive calls. In addition to the terms and conditions set forth in this GPA, the Participants agree to honor the intent of restrictions encoded in the DUNDi protocol. To that end, Participants agree to the following:

(a) A Participant may not utilize or permit the utilization of Routes for which the Subscriber or Egress Gateway provider has indicated that they do not wish to receive "Unsolicited Calls" for the purpose of making an unsolicited phone call on behalf of any party or organization.

(b) A Participant may not utilize or permit the utilization of Routes which have indicated that they do not wish to receive "Unsolicited Commercial Calls" for the purpose of making an unsolicited phone call on behalf of a commercial organization.

(c) A Participant may never utilize or permit the utilization of any DUNDi route for the purpose of making harassing phone calls.

(d) A Party may not utilize or permit the utilization of DUNDi provided Routes for any systematic or random calling of numbers (e.g., for the purpose of locating facsimile, modem services, or systematic telemarketing).

(e) Initial control signaling for all communication sessions that utilize Routes obtained from the Peering System must be sent from a member of the Peering System to the Service or Egress Gateway identified in the selected Route. For example, 'SIP INVITES' and IAX2 "NEW" commands must be sent from the requesting DUNDi node to the terminating Service.

(f) A Participant may not disclose any specific Route, Service or Participant contact information obtained from the Peering System to any party outside of the Peering System except as a by-product of facilitating communication in accordance with section 2e (e.g., phone books or other databases may not be published, but the Internet addresses of the Egress Gateway or Service does not need to be obfuscated.)

(g) The DUNDi Protocol requires that each Participant include valid contact information about itself (including information about nodes connected to each Participant). Participants may use or disclose the contact information only to ensure enforcement of legal furtherance of this Agreement.

3. ROUTES. The Participants shall only propagate valid Routes, as defined herein, through the Peering System, regardless of the original source. The Participants may only provide Routes as set forth below, and then only if such Participant has no good faith reason to believe

such Route to be invalid or unauthorized.

(a) A Participant may provide Routes if each Route has as its original source another member of the Peering System who has duly executed the GPA and such Routes are provided in accordance with this Agreement; provided that the Routes are not modified (e.g., with regards to existence, destination, technology or Weight); or

(b) A Participant may provide Routes for Services with any Weight for which it is the Subscriber; or

(c) A Participant may provide Routes for those Services whose Subscriber has authorized the Participant to do so, provided that the Participant is able to confirm that the Authorizing Individual is the Subscriber through:

i. a written statement of ownership from the Authorizing Individual, which the Participant believes in good faith to be accurate (e.g., a phone bill with the name of the Authorizing Individual and the number in question); or

ii. the Participant's own direct personal knowledge that the Authorizing Individual is the Subscriber.

(d) A Participant may provide Routes for Services, with Weight in accordance with the Current DUNDi Specification, if it can in good faith provide an Egress Gateway to that Service on the traditional telephone network without cost to the calling party.

4. REVOCATION. A Participant must provide a free, easily accessible mechanism by which a Subscriber may revoke permission to act as a Route Authority for his Service. A Participant must stop acting as a Route Authority for that Service within 7 days after:

(a) receipt of a revocation request;

(b) receiving other notice that the Service is no longer valid; or

(c) determination that the Subscriber's information is no longer accurate (including that the Subscriber is no longer the service owner or the service owner's authorized delegate).

5. SERVICE FEES. A Participant may charge a fee to act as a Route Authority for a Service, with any Weight, provided that no Participant may charge a fee to propagate the Route received through the Peering System.

6. TOLL SERVICES. No Participant may provide Routes for any Services that require payment from the calling party or their customer for communication with the Service. Nothing in this section shall prohibit a Participant from providing routes for Services where the calling party may later enter into a financial transaction with the called party (e.g., a Participant may provide Routes for calling cards services).

7. QUALITY. A Participant may not intentionally impair communication using a Route provided to the Peering System (e.g. by adding delay, advertisements, reduced quality). If for any reason a Participant is unable to deliver a call via a Route provided to the Peering System, that Participant shall return out-of-band Network Congestion notification (e.g. "503 Service Unavailable" with SIP protocol or "CONGESTION" with IAX protocol).

8. PROTOCOL COMPLIANCE. Participants agree to Propagate Routes in strict compliance with current DUNDi protocol specifications.

9. ADMINISTRATIVE FEES. A Participant may charge (but is not required to charge) another Participant a reasonable fee to cover administrative expenses incurred in the execution of this Agreement. A Participant may not charge any fee to continue the relationship or to provide Routes to another Participant in the Peering System.

10. CALLER IDENTIFICATION. A Participant will make a good faith effort to ensure the accuracy and appropriate nature of any caller identification that it transmits via any Route obtained from the Peering System. Caller identification shall at least be provided as a valid

E.164 number.

11. COMPLIANCE WITH LAWS. The Participants are solely responsible for determining to what extent, if any, the obligations set forth in this GPA conflict with any laws or regulations their region. A Participant may not provide any service or otherwise use DUNDi under this GPA if doing so is prohibited by law or regulation, or if any law or regulation imposes requirements on the Participant that are inconsistent with the terms of this GPA or the Acceptable Use Policy.

12. WARRANTY. EACH PARTICIPANT WARRANTS TO THE OTHER PARTICIPANTS THAT IT MADE, AND WILL CONTINUE TO MAKE, A GOOD FAITH EFFORT TO AUTHENTICATE OTHERS IN THE PEERING SYSTEM AND TO PROVIDE ACCURATE INFORMATION IN ACCORDANCE WITH THE TERMS OF THIS GPA. THIS WARRANTY IS MADE BETWEEN THE PARTICIPANTS, AND THE PARTICIPANTS MAY NOT EXTEND THIS WARRANTY TO ANY NON-PARTICIPANT INCLUDING END-USERS.

13. DISCLAIMER OF WARRANTIES. THE PARTICIPANTS UNDERSTAND AND AGREE THAT ANY SERVICE PROVIDED AS A RESULT OF THIS GPA IS "AS IS." EXCEPT FOR THOSE WARRANTIES OTHERWISE EXPRESSLY SET FORTH HEREIN, THE PARTICIPANTS DISCLAIM ANY REPRESENTATIONS OR WARRANTIES OF ANY KIND OR NATURE, EXPRESS OR IMPLIED, AS TO THE CONDITION, VALUE OR QUALITIES OF THE SERVICES PROVIDED HEREUNDER, AND SPECIFICALLY DISCLAIM ANY REPRESENTATION OR WARRANTY OF MERCHANTABILITY, SUITABILITY OR FITNESS FOR A PARTICULAR PURPOSE OR AS TO THE CONDITION OR WORKMANSHIP THEREOF, OR THE ABSENCE OF ANY DEFECTS THEREIN, WHETHER LATENT OR PATENT, INCLUDING ANY WARRANTIES ARISING FROM A COURSE OF DEALING, USAGE OR TRADE PRACTICE. EXCEPT AS EXPRESSLY PROVIDED HEREIN, THE PARTICIPANTS EXPRESSLY DISCLAIM ANY REPRESENTATIONS OR WARRANTIES THAT THE PEERING SERVICE WILL BE CONTINUOUS, UNINTERRUPTED OR ERROR-FREE, THAT ANY DATA SHARED OR OTHERWISE MADE AVAILABLE WILL BE ACCURATE OR COMPLETE OR OTHERWISE COMPLETELY SECURE FROM UNAUTHORIZED ACCESS.

14. LIMITATION OF LIABILITIES. NO PARTICIPANT SHALL BE LIABLE TO ANY OTHER PARTICIPANT FOR INCIDENTAL, INDIRECT, CONSEQUENTIAL, SPECIAL, PUNITIVE OR EXEMPLARY DAMAGES OF ANY KIND (INCLUDING LOST REVENUES OR PROFITS, LOSS OF BUSINESS OR LOSS OF DATA) IN ANY WAY RELATED TO THIS GPA, WHETHER IN CONTRACT OR IN TORT, REGARDLESS OF WHETHER SUCH

PARTICIPANT WAS ADVISED OF THE POSSIBILITY THEREOF.

15. END-USER AGREEMENTS. The Participants may independently enter into agreements with end-users to provide certain services (e.g., fees to a Subscriber to originate Routes for that Service). To the extent that provision of these services employs the Peering System, the Parties will include in their agreements with their end-users terms and conditions consistent with the terms of this GPA with respect to the exclusion of warranties, limitation of liability and Acceptable Use Policy. In no event may a Participant extend the warranty described in Section 12 in this GPA to any end-users.

16. INDEMNIFICATION. Each Participant agrees to defend, indemnify and hold harmless the other Participant or third-party beneficiaries to this GPA (including their affiliates, successors, assigns, agents and representatives and their respective officers, directors and employees) from and against any and all actions, suits, proceedings, investigations, demands, claims, judgments, liabilities, obligations, liens, losses, damages, expenses (including, without limitation, attorneys' fees) and any other fees arising out of or relating to (i) personal injury or property damage caused by that Participant, its employees, agents, servants, or other representatives; (ii) any act or omission by the Participant, its employees, agents, servants or other representatives, including, but not limited to, unauthorized representations or warranties made by the Participant; or (iii) any breach by the Participant of any of the terms or conditions of this GPA.

17. THIRD PARTY BENEFICIARIES. This GPA is intended to benefit those Participants who have executed the GPA and who are in the Peering System. It is the intent of the Parties to this GPA to give to those Participants who are in the Peering System standing to bring any necessary legal action to enforce the terms of this GPA.

18. TERMINATION. Any Participant may terminate this GPA at any time, with or without cause. A Participant that terminates must immediately cease to Propagate.

19. CHOICE OF LAW. This GPA and the rights and duties of the Parties

hereto shall be construed and determined in accordance with the internal laws of the State of New York, United States of America, without regard to its conflict of laws principles and without application of the United Nations Convention on Contracts for the International Sale of Goods.

20. DISPUTE RESOLUTION. Unless otherwise agreed in writing, the exclusive procedure for handling disputes shall be as set forth herein. Notwithstanding such procedures, any Participant may, at any time, seek injunctive relief in addition to the process described below.

(a) Prior to mediation or arbitration the disputing Participants shall seek informal resolution of disputes. The process shall be initiated with written notice of one Participant to the other describing the dispute with reasonable particularity followed with a written response within ten (10) days of receipt of notice. Each Participant shall promptly designate an executive with requisite authority to resolve the dispute. The informal procedure shall commence within ten (10) days of the date of response. All reasonable requests for non-privileged information reasonably related to the dispute shall be honored. If the dispute is not resolved within thirty (30) days of commencement of the procedure either Participant may proceed to mediation or arbitration pursuant to the rules set forth in (b) or (c) below.

(b) If the dispute has not been resolved pursuant to (a) above or, if the disputing Participants fail to commence informal dispute resolution pursuant to (a) above, either Participant may, in writing and within twenty (20) days of the response date noted in (a) above, ask the other Participant to participate in a one (1) day mediation with an impartial mediator, and the other Participant shall do so. Each Participant will bear its own expenses and an equal share of the fees of the mediator. If the mediation is not successful the Participants may proceed with arbitration pursuant to (c) below.

(c) If the dispute has not been resolved pursuant to (a) or (b) above, the dispute shall be promptly referred, no later than one (1) year from the date of original notice and subject to

applicable statute of limitations, to binding arbitration in accordance with the UNCITRAL Arbitration Rules in effect on the date of this contract. The appointing authority shall be the International Centre for Dispute Resolution. The case shall be administered by the International Centre for Dispute Resolution under its Procedures for Cases under the UNCITRAL Arbitration Rules. Each Participant shall bear its own expenses and shall share equally in fees of the arbitrator. All arbitrators shall have substantial experience in information technology and/or in the telecommunications business and shall be selected by the disputing participants in accordance with UNCITRAL Arbitration Rules. If any arbitrator, once selected is unable or unwilling to continue for any reason, replacement shall be filled via the process described above and a re-hearing shall be conducted. The disputing Participants will provide each other with all requested documents and records reasonably related to the dispute in a manner that will minimize the expense and inconvenience of both parties. Discovery will not include depositions or interrogatories except as the arbitrators expressly allow upon a showing of need. If disputes arise concerning discovery requests, the arbitrators shall have sole and complete discretion to resolve the disputes. The parties and arbitrator shall be guided in resolving discovery disputes by the Federal Rules of Civil Procedure. The Participants agree that time of the essence principles shall guide the hearing and that the arbitrator shall have the right and authority to issue monetary sanctions in the event of unreasonable delay. The arbitrator shall deliver a written opinion setting forth findings of fact and the rationale for the award within thirty (30) days following conclusion of the hearing. The award of the arbitrator, which may include legal and equitable relief, but which may not include punitive damages, will be final and binding upon the disputing Participants, and judgment may be entered upon it in accordance with applicable law in any court having jurisdiction thereof. In addition to award the arbitrator shall have the discretion to award the prevailing Participant all or part of its attorneys' fees and costs, including fees associated with arbitrator, if the arbitrator

determines that the positions taken by the other Participant on material issues of the dispute were without substantial foundation. Any conflict between the UNCITRAL Arbitration Rules and the provisions of this GPA shall be controlled by this GPA.

21. INTEGRATED AGREEMENT. This GPA, constitutes the complete integrated agreement between the parties concerning the subject matter hereof. All prior and contemporaneous agreements, understandings, negotiations or representations, whether oral or in writing, relating to the subject matter of this GPA are superseded and canceled in their entirety.

22. WAIVER. No waiver of any of the provisions of this GPA shall be deemed or shall constitute a waiver of any other provision of this GPA, whether or not similar, nor shall such waiver constitute a continuing waiver unless otherwise expressly so provided in writing. The failure of either party to enforce at any time any of the provisions of this GPA, or the failure to require at any time performance by either party of any of the provisions of this GPA, shall in no way be construed to be a present or future waiver of such provisions, nor in any way affect the ability of a Participant to enforce each and every such provision thereafter.

23. INDEPENDENT CONTRACTORS. Nothing in this GPA shall make the Parties partners, joint venturers, or otherwise associated in or with the business of the other. Parties are, and shall always remain, independent contractors. No Participant shall be liable for any debts, accounts, obligations, or other liabilities of the other Participant, its agents or employees. No party is authorized to incur debts or other obligations of any kind on the part of or as agent for the other. This GPA is not a franchise agreement and does not create a franchise relationship between the parties, and if any provision of this GPA is deemed to create a franchise between the parties, then this GPA shall automatically terminate.

24. CAPTIONS AND HEADINGS. The captions and headings used in this GPA are used for convenience only and are not to be given any legal effect.

25. EXECUTION. This GPA may be executed in counterparts, each of which so executed will be deemed to be an original and such counterparts together will constitute one and the same Agreement. The Parties shall transmit to each other a signed copy of the GPA by any means that faithfully reproduces the GPA along with the Signature. For purposes of this GPA, the term "signature" shall include digital signatures as defined by the jurisdiction of the Participant signing the GPA.

Exhibit A

Weight Range	Requirements
0-99	May only be used under authorization of Owner
100-199	May only be used by the Owner's service provider, regardless of authorization.
200-299	Reserved -- do not use for e164 context.
300-399	May only be used by the owner of the code under which the Owner's number is a part of.
400-499	May be used by any entity providing access via direct connectivity to the Public Switched Telephone Network.
500-599	May be used by any entity providing access via indirect connectivity to the Public Switched Telephone Network (e.g. Via another VoIP provider)
600-	Reserved-- do not use for e164 context.

Participant Participant

Company:

Address:

Email:

_____	_____
Authorized Signature	Authorized Signature

Name:

END OF GENERAL PEERING AGREEMENT

How to Peer using this GPA If you wish to exchange routing information with parties using the e164 DUNDi context, all you must do is execute this GPA with any member of the Peering System and you will become a member of the Peering System and be able to make Routes available in accordance with this GPA.

DUNDi, IAX, Asterisk and GPA are trademarks of Digium, Inc.

Chapter 8

ENUM

8.1 The ENUMLOOKUP dialplan function

The ENUMLOOKUP function is more complex than it first may appear, and this guide is to give a general overview and set of examples that may be well-suited for the advanced user to evaluate in their consideration of ENUM or ENUM-like lookup strategies. This document assumes a familiarity with ENUM (RFC3761) or ENUM-like methods, as well as familiarity with NAPTR DNS records (RFC2915, RFC3401-3404). For an overview of NAPTR records, and the use of NAPTRs in the ENUM global phone-number-to-DNS mapping scheme, please see `http://www.voip-info.org/tiki-index.php?page=ENUM` for more detail.

Using ENUM within Asterisk can be simple or complex, depending on how many failover methods and redundancy procedures you wish to utilize. Implementation of ENUM paths is supposedly defined by the person creating the NAPTR records, but the local administrator may choose to ignore certain NAPTR response methods (URI types) or prefer some over others, which is in contradiction to the RFC. The ENUMLOOKUP method simply provides administrators a method for determining NAPTR results in either the globally unique ENUM (e164.arpa) DNS tree, or in other ENUM-like DNS trees which are not globally unique. The methods to actually create channels ("dial") results given by the ENUMLOOKUP function is then up to the administrator to implement in a way that best suits their environment.

```
Function: ENUMLOOKUP(number[,Method-type[,options[,record#[,zone-suffix]]]])
```

Performs an ENUM tree lookup on the specified number, method type, and ordinal record offset, and returns one of four different values:

1. post-parsed NAPTR of one method (URI) type
2. count of elements of one method (URI) type
3. count of all method types
4. full URI of method at a particular point in the list of all possible methods

8.1.1 Arguments

- number
  - telephone number or search string. Only numeric values within this string are parsed; all other digits are ignored for search, but are re-written during NAPTR regexp expansion.
- service_type
  - tel, sip, h323, iax2, mailto, ...[any other string], ALL. Default type is "sip". Special name of "ALL" will create a list of method types across all NAPTR records for the search number, and then put the results in an ordinal list starting with 1. The position ¡number¿ specified will then be returned, starting with 1 as the first record (lowest value) in the list. The service types are not hardcoded in Asterisk except for the default (sip) if no other service type specified; any method type string (IANA-approved or not) may be used except for the string "ALL".
- options
  - c
    - count. Returns the number of records of this type are returned (regardless of order or priority.) If "ALL" is the specified service_type, then a count of all methods will be returned for the DNS record.
- record#

- which record to present if multiple answers are returned ¡integer¿ = The record in priority/order sequence based on the total count of records passed back by the query. If a service_type is specified, all entries of that type will be sorted into an ordinal list starting with 1 (by order first, then priority). The default of ¡options¿ is "1"

- zone_suffix
  - allows customization of the ENUM zone. Default is e164.arpa.

8.1.2 Examples

Let's use this ENUM list as an example (note that these examples exist in the DNS, and will hopefully remain in place as example destinations, but they may change or become invalid over time. The end result URIs are not guaranteed to actually work, since some of these hostnames or SIP proxies are imaginary. Of course, the tel: replies go to directory assistance for New York City and San Francisco...) Also note that the complex SIP NAPTR at weight 30 will strip off the leading "+" from the dialed string if it exists. This is probably a better NAPTR than hard-coding the number into the NAPTR, and it is included as a more complex regexp example, though other simpler NAPTRs will work just as well.

```
0.2.0.1.1.6.5.1.0.3.1.loligo.com. 3600 IN NAPTR 10 100 "u"
     "E2U+tel" "!^\\+13015611020$!tel:+12125551212!" .
0.2.0.1.1.6.5.1.0.3.1.loligo.com. 3600 IN NAPTR 21 100 "u"
     "E2U+tel" "!^\\+13015611020$!tel:+14155551212!" .
0.2.0.1.1.6.5.1.0.3.1.loligo.com. 3600 IN NAPTR 25 100 "u"
     "E2U+sip" "!^\\+13015611020$!sip:2203@sip.fox-den.com!" .
0.2.0.1.1.6.5.1.0.3.1.loligo.com. 3600 IN NAPTR 26 100 "u"
     "E2U+sip" "!^\\+13015611020$!sip:1234@sip-2.fox-den.com!" .
0.2.0.1.1.6.5.1.0.3.1.loligo.com. 3600 IN NAPTR 30 100 "u"
     "E2U+sip" "!^\\+*([^\\*]*)!sip:\\1@sip-3.fox-den.com!" .
0.2.0.1.1.6.5.1.0.3.1.loligo.com. 3600 IN NAPTR 55 100 "u"
     "E2U+mailto" "!^\\+13015611020$!mailto:jtodd@fox-den.com!" .
```

Example 1: Simplest case, using first SIP return (use all defaults except for domain name)

```
exten => 100,1,Set(foo=${ENUMLOOKUP(+13015611020,,,,loligo.com)})
  returns: ${foo}="2203@sip.fox-den.com"
```

Example 2: What is the first "tel" pointer type for this number? (after sorting by order/preference; default of "1" is assumed in options field)

```
exten => 100,1,Set(foo=${ENUMLOOKUP(+13015611020,tel,,,loligo.com)})
  returns: ${foo}="+12125551212"
```

Example 3: How many "sip" pointer type entries are there for this number?

```
exten => 100,1,Set(foo=${ENUMLOOKUP(+13015611020,sip,c,,loligo.com)})
  returns: ${foo}=3
```

Example 4: For all the "tel" pointer type entries, what is the second one in the list? (after sorting by preference)

```
exten => 100,1,Set(foo=${ENUMLOOKUP(+13015611020,tel,,2,loligo.com)})
  returns: ${foo}="+14155551212"
```

Example 5: How many NAPTRs (tel, sip, mailto, etc.) are in the list for this number?

```
exten => 100,1,Set(foo=${ENUMLOOKUP(+13015611020,ALL,c,,loligo.com)})
  returns: ${foo}=6
```

Example 6: Give back the second full URI in the sorted list of all NAPTR URIs:

```
exten => 100,1,Set(foo=${ENUMLOOKUP(+13015611020,ALL,,2,loligo.com)})
  returns: ${foo}="tel:+14155551212"  [note the "tel:" prefix in the string]
```

Example 7: Look up first SIP entry for the number in the e164.arpa zone (all defaults)

```
exten => 100,1,Set(foo=${ENUMLOOKUP(+437203001721)})
  returns: ${foo}="enum-test@sip.nemox.net"  [note: this result is
  subject to change as it is "live" DNS and not under my control]
```

Example 8: Look up the ISN mapping in freenum.org alpha test zone

```
exten => 100,1,Set(foo=${ENUMLOOKUP(1234*256,,,,freenum.org)})
  returns: ${foo}="1234@204.91.156.10"  [note: this result is subject
  to change as it is "live" DNS]
```

Example 9: Give back the first SIP pointer for a number in the

```
enum.yoydynelabs.com zone (invalid lookup)
exten => 100,1,Set(foo=${ENUMLOOKUP(1234567890,sip,,1,enum.yoyodynelabs.com)})
  returns: ${foo}=""
```

8.1.3 Usage notes and subtle features

- The use of "+" in lookups is confusing, and warrants further explanation. All E.164 numbers ("global phone numbers") by definition need a leading "+" during ENUM lookup. If you neglect to add a leading "+", you may discover that numbers that seem to exist in the DNS aren't getting matched by the system or are returned with a null string result. This is due to the NAPTR reply requiring a "+" in the regular expression matching sequence. Older versions of Asterisk add a "+" from within the code, which may confuse administrators converting to the new function. Please ensure that all ENUM (e164.arpa) lookups contain a leading "+" before lookup, so ensure your lookup includes the leading plus sign. Other DNS trees may or may not require a leading "+" - check before using those trees, as it is possible the parsed NAPTRs will not provide correct results unless you have the correct dialed string. If you get console messages like "WARNING[24907]: enum.c:222 parse_naptr: NAPTR Regex match failed." then it is very possible that the returned NAPTR expects a leading "+" in the search string (or the returned NAPTR is mis-formed.)

- If a query is performed of type "c" ("count") and let's say you get back 5 records and then some seconds later a query is made against record 5 in the list, it may not be the case that the DNS resolver has the same answers as it did a second or two ago - maybe there are only 4 records in the list in the newest query. The resolver should be the canonical storage location for DNS records, since that is the intent of ENUM. However, some obscure future cases may have wildly changing NAPTR records within several seconds. This is a corner case, and probably only worth noting as a very rare circumstance. (note: I do

not object to Asterisk's dnsmgr method of locally caching DNS replies, but this method needs to honor the TTL given by the remote zone master. Currently, the ENUMLOOKUP function does not use the dnsmgr method of caching local DNS replies.)

- If you want strict NAPTR value ordering, then it will be necessary to use the "ALL" method to incrementally step through the different returned NAPTR pointers. You will need to use string manipulation to strip off the returned method types, since the results will look like "sip:12125551212" in the returned value. This is a non-trivial task, though it is required in order to have strict RFC compliance and to comply with the desires of the remote party who is presenting NAPTRs in a particular order for a reason.

- Default behavior for the function (even in event of an error) is to move to the next priority, and the result is a null value. Most ENUM lookups are going to be failures, and it is the responsibility of the dialplan administrator to manage error conditions within their dialplan. This is a change from the old app_enumlookup method and it's arbitrary priority jumping based on result type or failure.

- Anything other than digits will be ignored in lookup strings. Example: a search string of "+4372030blah01721" will turn into 1.2.7.1.0.0.3.0.2.7.3.4.e164.arpa. for the lookup. The NAPTR parsing may cause unexpected results if there are strings inside your NAPTR lookups.

- If there exist multiple records with the same weight and order as a result of your query, the function will RANDOMLY select a single NAPTR from those equal results.

- Currently, the function ignores the settings in enum.conf as the search zone name is now specified within the function, and the H323 driver can be chosen by the user via the dialplan. There were no other values in this file, and so it becomes deprecated.

- The function will digest and return NAPTRs which use older (deprecated) style, reversed method strings such as "sip+E2U" instead of the more modern "E2U+sip"

- There is no provision for multi-part methods at this time. If there are multiple NAPTRs with (as an example) a method of "E2U+voice:sip" and then another NAPTR in the same DNS record with a method of ""E2U+sip", the system will treat these both as method "sip" and they will be separate records from the perspective of the function. Of course, if both records point to the same URI and have equal priority/weight (as is often the case) then this will cause no serious difficulty, but it bears mentioning.

- ISN (ITAD Subscriber Number) usage: If the search number is of the form ABC*DEF (where ABC and DEF are at least one numeric digit) then perform an ISN-style lookup where the lookup is manipulated to C.B.A.DEF.domain.tld (all other settings and options apply.) See `http://www.freenum.org/` for more details on ISN lookups. In the unlikely event you wish to avoid ISN re-writes, put an "n" as the first digit of the search string - the "n" will be ignored for the search.

8.1.4 Some more Examples

All examples below except where noted use "e164.arpa" as the referenced domain, which is the default domain name for ENUMLOOKUP. All numbers are assumed to not have a leading "+" as dialed by the inbound channel, so that character is added where necessary during ENUMLOOKUP function calls.

```
; example 1
;
; Assumes North American international dialing (011) prefix.
; Look up the first SIP result and send the call there, otherwise
;  send the call out a PRI.  This is the most simple possible
;  ENUM example, but only uses the first SIP reply in the list of
;  NAPTR(s).
;
exten => _011.,1,Set(enumresult=${ENUMLOOKUP(+${EXTEN:3})})
exten => _011.,n,Dial(SIP/${enumresult})
exten => _011.,n,Dial(DAHDI/g1/${EXTEN})
;
; end example 1

; example 2
;
; Assumes North American international dialing (011) prefix.
; Check to see if there are multiple SIP NAPTRs returned by
;  the lookup, and dial each in order.  If none work (or none
;  exist) then send the call out a PRI, group 1.
```

```
;
exten => _011.,1,Set(sipcount=${ENUMLOOKUP(${EXTEN:3},sip,c)}|counter=0)
exten => _011.,n,While($["${counter}"<"${sipcount}"])
exten => _011.,n,Set(counter=$[${counter}+1])
exten => _011.,n,Dial(SIP/${ENUMLOOKUP(+${EXTEN:3},sip,,${counter})})
exten => _011.,n,EndWhile
exten => _011.,n,Dial(DAHDI/g1/${EXTEN})
;
; end example 2

; example 3
;
; This example expects an ${EXTEN} that is an e.164 number (like
;  14102241145 or 437203001721)
; Search through e164.arpa and then also search through e164.org
;  to see if there are any valid SIP or IAX termination capabilities.
;  If none, send call out via DAHDI channel 1.
;
; Start first with e164.arpa zone...
;
exten => _X.,1,Set(sipcount=${ENUMLOOKUP(+${EXTEN},sip,c)}|counter=0)
exten => _X.,2,GotoIf($["${counter}"<"${sipcount}"]?3:6)
exten => _X.,3,Set(counter=$[${counter}+1])
exten => _X.,4,Dial(SIP/${ENUMLOOKUP(+${EXTEN},sip,,${counter})})
exten => _X.,5,GotoIf($["${counter}"<"${sipcount}"]?3:6)
;
exten => _X.,6,Set(iaxcount=${ENUMLOOKUP(+${EXTEN},iax2,c)}|counter=0)
exten => _X.,7,GotoIf($["${counter}"<"${iaxcount}"]?8:11)
exten => _X.,8,Set(counter=$[${counter}+1])
exten => _X.,9,Dial(IAX2/${ENUMLOOKUP(+${EXTEN},iax2,,${counter})})
exten => _X.,10,GotoIf($["${counter}"<"${iaxcount}"]?8:11)
;
exten => _X.,11,NoOp("No valid entries in e164.arpa for ${EXTEN} - checking in e164.org")
;
; ...then also try e164.org, and look for SIP and IAX NAPTRs...
;
exten => _X.,12,Set(sipcount=${ENUMLOOKUP(+${EXTEN},sip,c,,e164.org)}|counter=0)
exten => _X.,13,GotoIf($["${counter}"<"${sipcount}"]?14:17)
exten => _X.,14,Set(counter=$[${counter}+1])
exten => _X.,15,Dial(SIP/${ENUMLOOKUP(+${EXTEN},sip,,${counter},e164.org)})
exten => _X.,16,GotoIf($["${counter}"<"${sipcount}"]?14:17)
;
exten => _X.,17,Set(iaxcount=${ENUMLOOKUP(+${EXTEN},iax2,c,,e164.org)}|counter=0)
exten => _X.,18,GotoIf($["${counter}"<"${iaxcount}"]?19:22)
exten => _X.,19,Set(counter=$[${counter}+1])
exten => _X.,20,Dial(IAX2/${ENUMLOOKUP(+${EXTEN},iax2,,${counter},e164.org)})
exten => _X.,21,GotoIf($["${counter}"<"${iaxcount}"]?19:22)
;
; ...then send out PRI.
;
exten => _X.,22,NoOp("No valid entries in e164.org for ${EXTEN} - sending out via DAHDI")
exten => _X.,23,Dial(DAHDI/g1/${EXTEN})
;
; end example 3
```

Chapter 9

AMI: Asterisk Manager Interface

9.1 The Asterisk Manager TCP/IP API

The manager is a client/server model over TCP. With the manager interface, you'll be able to control the PBX, originate calls, check mailbox status, monitor channels and queues as well as execute Asterisk commands.

AMI is the standard management interface into your Asterisk server. You configure AMI in manager.conf. By default, AMI is available on TCP port 5038 if you enable it in manager.conf.

AMI receive commands, called "actions". These generate a "response" from Asterisk. Asterisk will also send "Events" containing various information messages about changes within Asterisk. Some actions generate an initial response and data in the form list of events. This format is created to make sure that extensive reports do not block the manager interface fully.

Management users are configured in the configuration file manager.conf and are given permissions for read and write, where write represents their ability to perform this class of "action", and read represents their ability to receive this class of "event".

If you develop AMI applications, treat the headers in Actions, Events and Responses as local to that particular message. There is no cross-message standardization of headers.

If you develop applications, please try to reuse existing manager headers and their interpretation. If you are unsure, discuss on the asterisk-dev mailing list.

9.2 Device status reports

Manager subscribes to extension status reports from all channels, to be able to generate events when an extension or device changes state. The level of details in these events may depend on the channel and device configuration. Please check each channel configuration file for more information. (in sip.conf, check the section on subscriptions and call limits)

9.3 Command Syntax

Management communication consists of tags of the form "header: value", terminated with an empty newline (\r\n) in the style of SMTP, HTTP, and other headers.

The first tag MUST be one of the following:

- Action: An action requested by the CLIENT to the Asterisk SERVER. Only one "Action" may be outstanding at any time.
- Response: A response to an action from the Asterisk SERVER to the CLIENT.
- Event: An event reported by the Asterisk SERVER to the CLIENT

9.4 Manager commands

To see all of the available manager commands, use the "manager show commands" CLI command.

You can get more information about a manager command with the "manager show command <command>" CLI command in Asterisk.

9.5 Examples

Login - Log a user into the manager interface.

```
Action: Login
Username: testuser
Secret: testsecret
```

Originate - Originate a call from a channel to an extension.

```
Action: Originate
Channel: sip/12345
Exten: 1234
Context: default
```

Originate - Originate a call from a channel to an extension without waiting for call to complete.

```
Action: Originate
Channel: sip/12345
Exten: 1234
Context: default
Async: yes
```

Redirect with ExtraChannel:

Attempted goal: Have a 'robot' program Redirect both ends of an already-connected call to a meetme room using the ExtraChannel feature through the management interface.

```
Action: Redirect
Channel: DAHDI/1-1
ExtraChannel: SIP/3064-7e00 (varies)
Exten: 680
Priority: 1
```

Where 680 is an extension that sends you to a MeetMe room.

There are a number of GUI tools that use the manager interface, please search the mailing list archives and the documentation page on the `http://www.asterisk.org` web site for more information.

9.6 Some standard AMI headers

```
Account:                -- Account Code (Status)
AccountCode:            -- Account Code (cdr_manager)
ACL: <Y | N>            -- Does ACL exist for object ?
Action: <action>        -- Request or notification of a particular action
```

```
Address-IP:              -- IPaddress
Address-Port:            -- IP port number
Agent: <string>          -- Agent name
AMAflags:                -- AMA flag (cdr_manager, sippeers)
AnswerTime:              -- Time of answer (cdr_manager)
Append: <bool>           -- CDR userfield Append flag
Application:             -- Application to use
Async:                   -- Whether or not to use fast setup
AuthType:                -- Authentication type (for login or challenge)
        "md5"
BillableSeconds:         -- Billable seconds for call (cdr_manager)
CallerID:                -- Caller id (name and number in Originate & cdr_manager
CallerID:                -- CallerID number
                            Number or "<unknown>" or "unknown"
                            (should change to "<unknown>" in app_queue)
CallerID1:               -- Channel 1 CallerID (Link event)
CallerID2:               -- Channel 2 CallerID (Link event)
CallerIDName:            -- CallerID name
                            Name or "<unknown>" or "unknown"
                            (should change to "<unknown>" in app_queue)
Callgroup:               -- Call group for peer/user
CallsTaken: <num>        -- Queue status variable
Cause: <value>           -- Event change cause - "Expired"
Cause: <value>           -- Hangupcause (channel.c)
CID-CallingPres:         -- Caller ID calling presentation
Channel: <channel>       -- Channel specifier
Channel: <dialstring>    -- Dialstring in Originate
Channel: <tech/[peer/username]> -- Channel in Registry events (SIP, IAX2)
Channel: <tech>          -- Technology (SIP/IAX2 etc) in Registry events
ChannelType:             -- Tech: SIP, IAX2, DAHDI, MGCP etc
Channel1:                -- Link channel 1
Channel2:                -- Link channel 2
ChanObjectType:          -- "peer", "user"
Codecs:                  -- Codec list
CodecOrder:              -- Codec order, separated with comma ","
Command:                 -- Cli command to run
Context:                 -- Context
Count: <num>             -- Number of callers in queue
```

```
Data:                     -- Application data
Default-addr-IP:          -- IP address to use before registration
Default-Username:         -- Username part of URI to use before registration
Destination:              -- Destination for call (Dialstring ) (dial, cdr_manager
DestinationContext:       -- Destination context (cdr_manager)
DestinationChannel:       -- Destination channel (cdr_manager)
DestUniqueID:             -- UniqueID of destination (dial event)
Disposition:              -- Call disposition (CDR manager)
Domain: <domain>          -- DNS domain
Duration: <secs>          -- Duration of call (cdr_manager)
Dynamic: <Y |  N>         -- Device registration supported?
Endtime:                  -- End time stamp of call (cdr_manager)
EventList: <flag>         -- Flag being "Start", "End", "Cancelled" or "ListObject
Events: <eventmask>       -- Eventmask filter ("on", "off", "system", "call", "log
Exten:                    -- Extension (Redirect command)
Extension:                -- Extension (Status)
Family: <string>          -- ASTdb key family
File: <filename>          -- Filename (monitor)
Format: <format>          -- Format of sound file (monitor)
From: <time>              --  Parking time (ParkedCall event)
Hint:                     -- Extension hint
Incominglimit:            -- SIP Peer incoming limit
Key:
Key:                      -- ASTdb Database key
LastApplication:          -- Last application executed (cdr_manager)
LastCall: <num>           -- Last call in queue
LastData:                 -- Data for last application (cdr_manager)
Link:                     -- (Status)
ListItems: <number>       -- Number of items in Eventlist (Optionally sent in "end
Location:                 -- Interface (whatever that is -maybe tech/name in app_c
Loginchan:                -- Login channel for agent
Logintime: <number>       -- Login time for agent
Mailbox:                  -- VM Mailbox (id@vmcontext) (mailboxstatus, mailboxcoun
MD5SecretExist: <Y | N>   -- Whether secret exists in MD5 format
Membership: <string>      -- "Dynamic" or "static" member in queue
Message: <text>           -- Text message in ACKs, errors (explanation)
Mix: <bool>               -- Boolean parameter (monitor)
NewMessages: <count>      -- Count of new Mailbox messages (mailboxcount)
```

```
Newname:
ObjectName:              -- Name of object in list
OldName:                 -- Something in Rename (channel.c)
OldMessages: <count>     -- Count of old mailbox messages (mailboxcount)
Outgoinglimit:           -- SIP Peer outgoing limit
Paused: <num>            -- Queue member paused status
Peer: <tech/name>        -- "channel" specifier :-)
PeerStatus: <tech/name>  -- Peer status code
                          "Unregistered", "Registered", "Lagged", "Reachable"
Penalty: <num>           -- Queue penalty
Priority:                -- Extension priority
Privilege: <privilege>   -- AMI authorization class (system, call, log, verbose,
Pickupgroup:             -- Pickup group for peer
Position: <num>          -- Position in Queue
Queue:                   -- Queue name
Reason:                  -- "Autologoff"
Reason:                  -- "Chanunavail"
Response: <response>     -- response code, like "200 OK"
                          "Success", "Error", "Follows"
Restart:                 -- "True", "False"
RegExpire:               -- SIP registry expire
RegExpiry:               -- SIP registry expiry
Reason:                  -- Originate reason code
Seconds:                 -- Seconds (Status)
Secret: <password>       -- Authentication secret (for login)
SecretExist: <Y | N>     -- Whether secret exists
Shutdown:                -- "Uncleanly", "Cleanly"
SIP-AuthInsecure:
SIP-FromDomain:          -- Peer FromDomain
SIP-FromUser:            -- Peer FromUser
SIP-NatSupport:
SIPLastMsg:
Source:                  -- Source of call (dial event, cdr_manager)
SrcUniqueID:             -- UniqueID of source (dial event)
StartTime:               -- Start time of call (cdr_manager)
State:                   -- Channel state
Status:                  -- Registration status (Registry events SIP)
Status:                  -- Extension status (Extensionstate)
```

```
Status:                  -- Peer status (if monitored)  ** Will change name **
                            "unknown", "lagged", "ok"
Status: <num>            -- Queue Status
Status:                  -- DND status (DNDState)
Time: <sec>              -- Roundtrip time (latency)
Timeout:                 -- Parking timeout time
Timeout:                 -- Timeout for call setup (Originate)
Timeout: <seconds>       -- Timeout for call
Uniqueid:                -- Channel Unique ID
Uniqueid1:               -- Channel 1 Unique ID (Link event)
Uniqueid2:               -- Channel 2 Unique ID (Link event)
User:                    -- Username (SIP registry)
UserField:               -- CDR userfield (cdr_manager)
Val:                     -- Value to set/read in ASTdb
Variable:                -- Variable AND value to set (multiple separated with |
Variable: <name>         -- For channel variables
Value: <value>           -- Value to set
VoiceMailbox:            -- VM Mailbox in SIPpeers
Waiting:                 -- Count of mailbox messages (mailboxstatus)
```

** Please try to re-use existing headers to simplify manager message parsing in clients.

Read the CODING-GUIDELINES if you develop new manager commands or events.

9.7 Asynchronous Javascript Asterisk Manger (AJAM)

AJAM is a new technology which allows web browsers or other HTTP enabled applications and web pages to directly access the Asterisk Manger Interface (AMI) via HTTP. Setting up your server to process AJAM involves a few steps:

9.7.1 Setup the Asterisk HTTP server

1. Uncomment the line "enabled=yes" in `/etc/asterisk/http.conf` to enable Asterisk's builtin micro HTTP server.

2. If you want Asterisk to actually deliver simple HTML pages, CSS, javascript, etc. you should uncomment "enablestatic=yes"

3. Adjust your "bindaddr" and "bindport" settings as appropriate for your desired accessibility

4. Adjust your "prefix" if appropriate, which must be the beginning of any URI on the server to match. The default is "asterisk" and the rest of these instructions assume that value.

9.7.2 Allow Manager Access via HTTP

1. Make sure you have both "enabled = yes" and "webenabled = yes" setup in `/etc/asterisk/manager.conf`

2. You may also use "httptimeout" to set a default timeout for HTTP connections.

3. Make sure you have a manager username/secret

Once those configurations are complete you can reload or restart Asterisk and you should be able to point your web browser to specific URI's which will allow you to access various web functions. A complete list can be found by typing "http show status" at the Asterisk CLI.

examples:

```
http://localhost:8088/asterisk/manager?action=login&username=foo&secret=bar
```

This logs you into the manager interface's "HTML" view. Once you're logged in, Asterisk stores a cookie on your browser (valid for the length of httptimeout) which is used to connect to the same session.

```
http://localhost:8088/asterisk/rawman?action=status
```

Assuming you've already logged into manager, this URI will give you a "raw" manager output for the "status" command.

```
http://localhost:8088/asterisk/mxml?action=status
```

This will give you the same status view but represented as AJAX data, theoretically compatible with RICO (`http://www.openrico.org`).

```
http://localhost:8088/asterisk/static/ajamdemo.html
```

If you have enabled static content support and have done a make install, Asterisk will serve up a demo page which presents a live, but very basic, "astman" like interface. You can login with your username/secret for manager and have a basic view of channels as well as transfer and hangup calls. It's only tested in Firefox, but could probably be made to run in other browsers as well.

A sample library (astman.js) is included to help ease the creation of manager HTML interfaces.

Note that for the demo, there is no need for *any* external web server.

9.7.3 Integration with other web servers

Asterisk's micro HTTP server is *not* designed to replace a general purpose web server and it is intentionally created to provide only the minimal interfaces required. Even without the addition of an external web server, one can use Asterisk's interfaces to implement screen pops and similar tools pulling data from other web servers using iframes, div's etc. If you want to integrate CGI's, databases, PHP, etc. you will likely need to use a more traditional web server like Apache and link in your Asterisk micro HTTP server with something like this:

```
ProxyPass /asterisk http://localhost:8088/asterisk
```

Chapter 10

CDR: Call Detail Records

10.1 Applications

- SetAccount - Set account code for billing
- SetAMAFlags - Sets AMA flags
- NoCDR - Make sure no CDR is saved for a specific call
- ResetCDR - Reset CDR
- ForkCDR - Save current CDR and start a new CDR for this call
- Authenticate - Authenticates and sets the account code
- SetCDRUserField - Set CDR user field
- AppendCDRUserField - Append data to CDR User field

For more information, use the "core show application <application>" command. You can set default account codes and AMA flags for devices in channel configuration files, like sip.conf, iax.conf etc.

10.2 Fields of the CDR in Asterisk

- accountcode: What account number to use, (string, 20 characters)

- src: Caller*ID number (string, 80 characters)
- dst: Destination extension (string, 80 characters)
- dcontext: Destination context (string, 80 characters)
- clid: Caller*ID with text (80 characters)
- channel: Channel used (80 characters)
- dstchannel: Destination channel if appropriate (80 characters)
- lastapp: Last application if appropriate (80 characters)
- lastdata: Last application data (arguments) (80 characters)
- start: Start of call (date/time)
- answer: Answer of call (date/time)
- end: End of call (date/time)
- duration: Total time in system, in seconds (integer), from dial to hangup
- billsec: Total time call is up, in seconds (integer), from answer to hangup
- disposition: What happened to the call: ANSWERED, NO ANSWER, BUSY
- amaflags: What flags to use: DOCUMENTATION, BILL, IGNORE etc, specified on a per channel basis like accountcode.
- user field: A user-defined field, maximum 255 characters

In some cases, uniqueid is appended:

- uniqueid: Unique Channel Identifier (32 characters) This needs to be enabled in the source code at compile time

NOTE: If you use IAX2 channels for your calls, and allow 'full' transfers (not media-only transfers), then when the calls is transferred the server in the middle will no longer be involved in the signaling path, and thus will not generate accurate CDRs for that call. If you can, use media-only transfers with IAX2 to avoid this problem, or turn off transfers completely (although this can result in a media latency increase since the media packets have to traverse the middle server(s) in the call).

10.3 CDR Variables

If the channel has a cdr, that cdr record has its own set of variables which can be accessed just like channel variables. The following builtin variables are available.

```
${CDR(clid)}         Caller ID
${CDR(src)}          Source
${CDR(dst)}          Destination
${CDR(dcontext)}     Destination context
${CDR(channel)}      Channel name
${CDR(dstchannel)}   Destination channel
${CDR(lastapp)}      Last app executed
${CDR(lastdata)}     Last app's arguments
${CDR(start)}        Time the call started.
${CDR(answer)}       Time the call was answered.
${CDR(end)}          Time the call ended.
${CDR(duration)}     Duration of the call.
${CDR(billsec)}      Duration of the call once it was answered.
${CDR(disposition)}  ANSWERED, NO ANSWER, BUSY
${CDR(amaflags)}     DOCUMENTATION, BILL, IGNORE etc
${CDR(accountcode)}  The channel's account code.
${CDR(uniqueid)}     The channel's unique id.
${CDR(userfield)}    The channels uses specified field.
```

In addition, you can set your own extra variables by using Set(CDR(name)=value). These variables can be output into a text-format CDR by using the cdr_custom CDR driver; see the cdr_custom.conf.sample file in the configs directory for an example of how to do this. Call data records can be stored in many different databases or even CSV text.

10.4 MSSQL

Asterisk can currently store CDRs into an MSSQL database in two different ways: cdr_odbc or cdr_tds

Call Data Records can be stored using unixODBC (which requires the FreeTDS package) [cdr_odbc] or directly by using just the FreeTDS package [cdr_tds] The following provide some examples known to get asterisk working with mssql.

NOTE: Only choose one db connector.

10.4.1 ODBC using cdr_odbc

Compile, configure, and install the latest unixODBC package:

```
tar -zxvf unixODBC-2.2.9.tar.gz &&
cd unixODBC-2.2.9 &&
./configure --sysconfdir=/etc --prefix=/usr --disable-gui &&
make &&
make install
```

Compile, configure, and install the latest FreeTDS package:

```
tar -zxvf freetds-0.62.4.tar.gz &&
cd freetds-0.62.4 &&
./configure --prefix=/usr --with-tdsver=7.0 \
                --with-unixodbc=/usr/lib &&
make && make install
```

Compile, or recompile, asterisk so that it will now add support for cdr_odbc.

```
make clean && ./configure --with-odbc &&
make update &&
make &&
make install
```

Setup odbc configuration files. These are working examples from my system. You will need to modify for your setup. You are not required to store usernames or passwords here.

```
/etc/odbcinst.ini
   [FreeTDS]
   Description    = FreeTDS ODBC driver for MSSQL
   Driver         = /usr/lib/libtdsodbc.so
   Setup          = /usr/lib/libtdsS.so
   FileUsage      = 1

/etc/odbc.ini
   [MSSQL-asterisk]
   description         = Asterisk ODBC for MSSQL
```

```
driver              = FreeTDS
server              = 192.168.1.25
port                = 1433
database            = voipdb
tds_version         = 7.0
language            = us_english
```

Only install one database connector. Do not confuse asterisk by using both ODBC (cdr_odbc) and FreeTDS (cdr_tds). This command will erase the contents of cdr_tds.conf

```
[ -f /etc/asterisk/cdr_tds.conf ] > /etc/asterisk/cdr_tds.conf
```

NOTE: unixODBC requires the freeTDS package, but asterisk does not call freeTDS directly.

Now set up cdr_odbc configuration files. These are working samples from my system. You will need to modify for your setup. Define your usernames and passwords here, secure file as well.

```
/etc/asterisk/cdr_odbc.conf
   [global]
   dsn=MSSQL-asterisk
   username=voipdbuser
   password=voipdbpass
   loguniqueid=yes
```

And finally, create the 'cdr' table in your mssql database.

```
CREATE TABLE cdr (
        [calldate]      [datetime]              NOT NULL ,
        [clid]          [varchar] (80)          NOT NULL ,
        [src]           [varchar] (80)          NOT NULL ,
        [dst]           [varchar] (80)          NOT NULL ,
        [dcontext]      [varchar] (80)          NOT NULL ,
        [channel]       [varchar] (80)          NOT NULL ,
        [dstchannel]    [varchar] (80)          NOT NULL ,
        [lastapp]       [varchar] (80)          NOT NULL ,
        [lastdata]      [varchar] (80)          NOT NULL ,
        [duration]      [int]                   NOT NULL ,
        [billsec]       [int]                   NOT NULL ,
        [disposition]   [varchar] (45)          NOT NULL ,
        [amaflags]      [int]                   NOT NULL ,
        [accountcode]   [varchar] (20)          NOT NULL ,
        [uniqueid]      [varchar] (150)         NOT NULL ,
        [userfield]     [varchar] (255)         NOT NULL
)
```

Start asterisk in verbose mode, you should see that asterisk logs a connection to the database and will now record every call to the database when it's complete.

10.4.2 TDS, using cdr_tds

Compile, configure, and install the latest FreeTDS package:

```
tar -zxvf freetds-0.62.4.tar.gz &&
cd freetds-0.62.4 &&
./configure --prefix=/usr --with-tdsver=7.0
make &&
make install
```

Compile, or recompile, asterisk so that it will now add support for cdr_tds.

```
make clean && ./configure --with-tds &&
make update &&
make &&
make install
```

Only install one database connector. Do not confuse asterisk by using both ODBC (cdr_odbc) and FreeTDS (cdr_tds). This command will erase the contents of cdr_odbc.conf

```
[ -f /etc/asterisk/cdr_odbc.conf ] > /etc/asterisk/cdr_odbc.conf
```

Setup cdr_tds configuration files. These are working samples from my system. You will need to modify for your setup. Define your usernames and passwords here, secure file as well.

```
            /etc/asterisk/cdr_tds.conf
[global]
hostname=192.168.1.25
port=1433
dbname=voipdb
user=voipdbuser
password=voipdpass
charset=BIG5
```

And finally, create the 'cdr' table in your mssql database.

```
CREATE TABLE cdr (
        [accountcode]   [varchar] (20)          NULL ,
        [src]           [varchar] (80)          NULL ,
        [dst]           [varchar] (80)          NULL ,
        [dcontext]      [varchar] (80)          NULL ,
        [clid]          [varchar] (80)          NULL ,
        [channel]       [varchar] (80)          NULL ,
        [dstchannel]    [varchar] (80)          NULL ,
        [lastapp]       [varchar] (80)          NULL ,
        [lastdata]      [varchar] (80)          NULL ,
        [start]         [datetime]              NULL ,
        [answer]        [datetime]              NULL ,
        [end]           [datetime]              NULL ,
        [duration]      [int]                   NULL ,
        [billsec]       [int]                   NULL ,
```

```
        [disposition]   [varchar] (20)          NULL ,
        [amaflags]      [varchar] (16)          NULL ,
        [uniqueid]      [varchar] (150)         NULL ,
        [userfield]     [varchar] (256)         NULL
)
```

Start asterisk in verbose mode, you should see that asterisk logs a connection to the database and will now record every call to the database when it's complete.

10.5 MYSQL

Using MySQL for CDR records is supported by using ODBC and the cdr_odbc module.

10.6 PGSQL

If you want to go directly to postgresql database, and have the cdr_pgsql.so compiled you can use the following sample setup. On Debian, before compiling asterisk, just install libpqxx-dev. Other distros will likely have a similiar package.

Once you have the compile done, copy the sample cdr_pgsql.conf file or create your own.

Here is a sample:

```
/etc/asterisk/cdr_pgsql.conf
  ; Sample Asterisk config file for CDR logging to PostgresSQL
  [global]
  hostname=localhost
  port=5432
  dbname=asterisk
  password=password
  user=postgres
  table=cdr
```

Now create a table in postgresql for your cdrs

```
CREATE TABLE cdr (
        calldate      time               NOT NULL ,
        clid          varchar (80)          NOT NULL ,
        src           varchar (80)          NOT NULL ,
        dst           varchar (80)          NOT NULL ,
        dcontext      varchar (80)          NOT NULL ,
        channel       varchar (80)          NOT NULL ,
        dstchannel    varchar (80)          NOT NULL ,
```

```
        lastapp       varchar (80)          NOT NULL ,
        lastdata      varchar (80)          NOT NULL ,
        duration      int                   NOT NULL ,
        billsec       int                   NOT NULL ,
        disposition   varchar (45)          NOT NULL ,
        amaflags      int                   NOT NULL ,
        accountcode   varchar (20)          NOT NULL ,
        uniqueid      varchar (150)         NOT NULL ,
        userfield     varchar (255)         NOT NULL
);
```

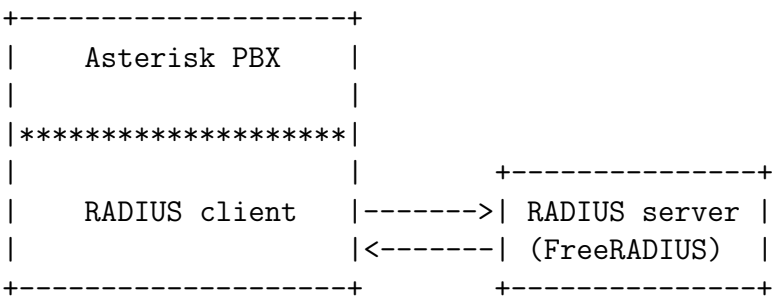
10.7 SQLLITE

SQLite version 2 is supported in cdr_sqlite.

10.8 RADIUS

10.8.1 What is needed

- FreeRADIUS server
- Radiusclient-ng library
- Asterisk PBX



10.8.2 Steps to follow in order to have RADIUS support

Installation of the Radiusclient library

Download the sources from `http://developer.berlios.de/projects/radiusclient-ng/`
Untar the source tarball:

```
root@localhost:/usr/local/src# tar xvfz radiusclient-ng-0.5.2.tar.gz
```

Compile and install the library:

```
root@localhost:/usr/local/src# cd radiusclient-ng-0.5.2
root@localhost:/usr/local/src/radiusclient-ng-0.5.2# ./configure
root@localhost:/usr/local/src/radiusclient-ng-0.5.2# make
root@localhost:/usr/local/src/radiusclient-ng-0.5.2# make install
```

Configuration of the Radiusclient library

By default all the configuration files of the radiusclient library will be in `/usr/local/etc/radiusclient-ng` directory.

File "radiusclient.conf" Open the file and find lines containing the following:

authserver localhost

This is the hostname or IP address of the RADIUS server used for authentication. You will have to change this unless the server is running on the same host as your Asterisk PBX.

acctserver localhost

This is the hostname or IP address of the RADIUS server used for accounting. You will have to change this unless the server is running on the same host as your Asterisk PBX.

File "servers"

RADIUS protocol uses simple access control mechanism based on shared secrets that allows RADIUS servers to limit access from RADIUS clients.

A RADIUS server is configured with a secret string and only RADIUS clients that have the same secret will be accepted.

You need to configure a shared secret for each server you have configured in radiusclient.conf file in the previous step. The shared secrets are stored in `/usr/local/etc/radiusclient-ng/servers` file.

Each line contains hostname of a RADIUS server and shared secret used in communication with that server. The two values are separated by white spaces. Configure shared secrets for every RADIUS server you are going to use.

File "dictionary"

Asterisk uses some attributes that are not included in the dictionary of radiusclient library, therefore it is necessary to add them. A file called dictionary.digium (kept in the contrib dir) was created to list all new attributes

used by Asterisk. Add to the end of the main dictionary file `/usr/local/etc/radiusclient-ng/dictionary` the line:

$INCLUDE /path/to/dictionary.digium

Install FreeRADIUS Server (Version 1.1.1)

Download sources tarball from:

`http://freeradius.org/`

Untar, configure, build, and install the server:

```
root@localhost:/usr/local/src# tar xvfz freeradius-1.1.1.tar.gz
root@localhost:/usr/local/src# cd freeradius-1.1.1
root@localhost"/usr/local/src/freeradius-1.1.1# ./configure
root@localhost"/usr/local/src/freeradius-1.1.1# make
root@localhost"/usr/local/src/freeradius-1.1.1# make install
```

All the configuration files of FreeRADIUS server will be in /usr/local/etc/raddb directory.

Configuration of the FreeRADIUS Server

There are several files that have to be modified to configure the RADIUS server. These are presented next.

File "clients.conf"

File `/usr/local/etc/raddb/clients.conf` contains description of RADIUS clients that are allowed to use the server. For each of the clients you need to specify its hostname or IP address and also a shared secret. The shared secret must be the same string you configured in radiusclient library.

Example:

```
client myhost {
    secret = mysecret
    shortname = foo
}
```

This fragment allows access from RADIUS clients on "myhost" if they use "mysecret" as the shared secret. The file already contains an entry for localhost (127.0.0.1), so if you are running the RADIUS server on the same host as your Asterisk server, then modify the existing entry instead, replacing the default password.

File "dictionary"

Note: as of version 1.1.2, the dictionary.digium file ships with FreeRADIUS. The following procedure brings the dictionary.digium file to previous versions of FreeRADIUS.

File `/usr/local/etc/raddb/dictionary` contains the dictionary of FreeRADIUS server. You have to add the same dictionary file (dictionary.digium), which you added to the dictionary of radiusclient-ng library. You can include it into the main file, adding the following line at the end of file `/usr/local/etc/raddb/dictionary`:

$INCLUDE /path/to/dictionary.digium

That will include the same new attribute definitions that are used in radiusclient-ng library so the client and server will understand each other.

Asterisk Accounting Configuration

Compilation and installation:

The module will be compiled as long as the radiusclient-ng library has been detected on your system.

By default FreeRADIUS server will log all accounting requests into `/usr/local/var/log/radius/radacct` directory in form of plain text files. The server will create one file for each hostname in the directory. The following example shows how the log files look like.

Asterisk now generates Call Detail Records. See `/include/asterisk/cdr.h` for all the fields which are recorded. By default, records in comma separated values will be created in `/var/log/asterisk/cdr-csv`.

The configuration file for cdr_radius.so module is `/etc/asterisk/cdr.conf`

This is where you can set CDR related parameters as well as the path to the radiusclient-ng library configuration file.

10.9 Logged Values

```
"Asterisk-Acc-Code",          The account name of detail records
"Asterisk-Src",
"Asterisk-Dst",
"Asterisk-Dst-Ctx",           The destination context
"Asterisk-Clid",
```

```
"Asterisk-Chan",              The channel
"Asterisk-Dst-Chan",         (if applicable)
"Asterisk-Last-App",         Last application run on the channel
"Asterisk-Last-Data",          Argument to the last channel
"Asterisk-Start-Time",
"Asterisk-Answer-Time",
"Asterisk-End-Time",
"Asterisk-Duration",           Duration is the whole length that the entire
                               call lasted. ie. call rx'd to hangup
                               "end time" minus "start time"
"Asterisk-Bill-Sec",           The duration that a call was up after other
                               end answered which will be <= to duration
                               "end time" minus "answer time"
"Asterisk-Disposition",        ANSWERED, NO ANSWER, BUSY
"Asterisk-AMA-Flags",         DOCUMENTATION, BILL, IGNORE etc, specified on
                               a per channel basis like accountcode.
"Asterisk-Unique-ID",          Unique call identifier
"Asterisk-User-Field"          User field set via SetCDRUserField
```

Chapter 11

Voicemail

11.1 ODBC Storage

ODBC Storage allows you to store voicemail messages within a database instead of using a file. This is **not** a full realtime engine and **only** supports ODBC. The table description for the "voicemessages" table is as follows:

```
+---------------+------------+------+-----+---------+-------+
| Field         | Type       | Null | Key | Default | Extra |
+---------------+------------+------+-----+---------+-------+
| msgnum        | int(11)    | YES  |     | NULL    |       |
| dir           | varchar(80)| YES  | MUL | NULL    |       |
| context       | varchar(80)| YES  |     | NULL    |       |
| macrocontext  | varchar(80)| YES  |     | NULL    |       |
| callerid      | varchar(40)| YES  |     | NULL    |       |
| origtime      | varchar(40)| YES  |     | NULL    |       |
| duration      | varchar(20)| YES  |     | NULL    |       |
| flag          | varchar(8) | YES  |     | NULL    |       |
| mailboxuser   | varchar(80)| YES  |     | NULL    |       |
| mailboxcontext| varchar(80)| YES  |     | NULL    |       |
| recording     | longblob   | YES  |     | NULL    |       |
+---------------+------------+------+-----+---------+-------+
```

The database name (from `/etc/asterisk/res_odbc.conf`) is in the "odbc-storage" variable in the general section of voicemail.conf.

You may modify the voicemessages table name by using odbctable=??? in voicemail.conf.

11.2 IMAP Storage

By enabling IMAP Storage, Asterisk will use native IMAP as the storage mechanism for voicemail messages instead of using the standard file structure.

Tighter integration of Asterisk voicemail and IMAP email services allows additional voicemail functionality, including:

- Listening to a voicemail on the phone will set its state to "read" in a user's mailbox automatically.
- Deleting a voicemail on the phone will delete it from the user's mailbox automatically.
- Accessing a voicemail recording email message will turn off the message waiting indicator (MWI) on the user's phone.
- Deleting a voicemail recording email will also turn off the message waiting indicator, and delete the message from the voicemail system.

11.2.1 Installation Notes

University of Washington IMAP C-Client

If you do not have the University of Washington's IMAP c-client installed on your system, you will need to download the c-client source distribution (`http://www.washington.edu/imap/`) and compile it. Asterisk supports the 2007 version of c-client as there appears to be issues with older versions which cause Asterisk to crash in certain scenarios. It is highly recommended that you utilize a current version of the c-client libraries. Additionally, mail_expunge_full is enabled in the 2006 and later versions.

Note that Asterisk only uses the 'c-client' portion of the UW IMAP toolkit, but building it also builds an IMAP server and various other utilities. Because of this, the build instructions for the IMAP toolkit are somewhat complicated and can lead to confusion about what is needed.

If you are going to be connecting Asterisk to an existing IMAP server, then you don't need to care about the server or utilities in the IMAP toolkit

at all. If you want to also install the UW IMAPD server, that is outside the scope of this document.

Building the c-client library is fairly straightforward; for example, on a Debian system there are two possibilities:

1. If you will not be using SSL to connect to the IMAP server:

```
$ make slx SSLTYPE=none
```

2. If you will be using SSL to connect to the IMAP server:

```
$ make slx EXTRACFLAGS="-I/usr/include/openssl"
```

Additionally, you may wish to build on a 64-bit machine, in which case you need to add -fPIC to EXTRACFLAGS. So, building on a 64-bit machine with SSL support would look something like:

```
$ make slx EXTRACFLAGS="-fPIC -I/usr/include/openssl"
```

Or without SSL support:

```
$ make slx SSLTYPE=none EXTRACFLAGS=-fPIC
```

Once this completes you can proceed with the Asterisk build; there is no need to run 'make install'.

Compiling Asterisk

Configure with ./configure –with-imap=/usr/src/imap or wherever you built the UWashington IMAP Toolkit. This directory will be searched for a source installation. If no source installation is found there, then a package installation of the IMAP c-client will be searched for in this directory. If one is not found, then configure will fail.

A second configure option is to not specify a directory (i.e. ./configure –with-imap). This will assume that you have the imap-2007e source installed in the ../imap directory relative to the Asterisk source. If you do not have

this source, then configure will default to the "system" option defined in the next paragraph

A third option is ./configure –with-imap=system. This will assume that you have installed a dynamically linked version of the c-client library (most likely via a package provided by your distro). This will attempt to link agains -lc-client and will search for c-client headers in your include path starting with the imap directory, and upon failure, in the c-client directory.

When you run 'make menuselect', choose 'Voicemail Build Options' and the IMAP_STORAGE option should be available for selection.

After selecting the IMAP_STORAGE option, use the 'x' key to exit menuselect and save your changes, and the build/install Asterisk normally.

11.2.2 Modify voicemail.conf

The following directives have been added to voicemail.conf:

```
imapserver=<name or IP address of IMAP mail server>
imapport=<IMAP port, defaults to 143>
imapflags=<IMAP flags, "novalidate-cert" for example>
imapfolder=<IMAP folder to store messages to>
imapgreetings=<yes or no>
greetingsfolder=<IMAP folder to store greetings in if imapgreetings is enabled>
expungeonhangup=<yes or no>
authuser=<username>
authpassword=<password>
opentimeout=<TCP open timeout in seconds>
closetimeout=<TCP close timeout in seconds>
readtimeout=<TCP read timeout in seconds>
writetimeout=<TCP write timeout in seconds>
```

The "imapfolder" can be used to specify an alternative folder on your IMAP server to store voicemails in. If not specified, the default folder 'INBOX' will be used.

The "imapgreetings" parameter can be enabled in order to store voicemail greetings on the IMAP server. If disabled, then they will be stored on the local file system as normal.

The "greetingsfolder" can be set to store greetings on the IMAP server when "imapgreetings" is enabled in an alternative folder than that set by "imapfolder" or the default folder for voicemails.

The "expungeonhangup" flag is used to determine if the voicemail system should expunge all messages marked for deletion when the user hangs up the phone.

Each mailbox definition should also have imapuser=<imap username>. For example:

```
4123=>4123,James Rothenberger,jar@onebiztone.com,,attach=yes|imapuser=jar
```

The directives "authuser" and "authpassword" are not needed when using Kerberos. They are defined to allow Asterisk to authenticate as a single user that has access to all mailboxes as an alternative to Kerberos.

11.2.3 IMAP Folders

Besides INBOX, users should create "Old", "Work", "Family" and "Friends" IMAP folders at the same level of hierarchy as the INBOX. These will be used as alternate folders for storing voicemail messages to mimic the behavior of the current (file-based) voicemail system.

Please note that it is not recommended to store your voicemails in the top level folder where your users will keep their emails, especially if there are a large number of emails. A large number of emails in the same folder(s) that you're storing your voicemails could cause a large delay as Asterisk must parse through all the emails. For example a mailbox with 100 emails in it could take up to 60 seconds to receive a response.

11.2.4 Separate vs. Shared Email Accounts

As administrator you will have to decide if you want to send the voicemail messages to a separate IMAP account or use each user's existing IMAP mailbox for voicemail storage. The IMAP storage mechanism will work either way.

By implementing a single IMAP mailbox, the user will see voicemail messages appear in the same INBOX as other messages. The disadvantage of this method is that if the IMAP server does NOT support UIDPLUS, Asterisk voicemail will expunge ALL messages marked for deletion when the user exits the voicemail system, not just the VOICEMAIL messages marked for deletion.

By implementing separate IMAP mailboxes for voicemail and email, voicemail expunges will not remove regular email flagged for deletion.

11.2.5 IMAP Server Implementations

There are various IMAP server implementations, each supports a potentially different set of features.

UW IMAP-2005 or earlier

UIDPLUS is currently NOT supported on these versions of UW-IMAP. Please note that without UID_EXPUNGE, Asterisk voicemail will expunge ALL messages marked for deletion when a user exits the voicemail system (hangs up the phone).

This version is *not* recommended for Asterisk.

UW IMAP-2006

This version supports UIDPLUS, which allows UID_EXPUNGE capabilities. This feature allow the system to expunge ONLY pertinent messages, instead of the default behavior, which is to expunge ALL messages marked for deletion when EXPUNGE is called. The IMAP storage mechanism is this version of Asterisk will check if the UID_EXPUNGE feature is supported by the server, and use it if possible.

This version is *not* recommended for Asterisk.

UW IMAP-2007

This is the currently recommended version for use with Asterisk.

Cyrus IMAP

Cyrus IMAP server v2.3.3 has been tested using a hierarchy delimiter of '/'.

11.2.6 Quota Support

If the IMAP server supports quotas, Asterisk will check the quota when accessing voicemail. Currently only a warning is given to the user that their quota is exceeded.

11.2.7 Application Notes

Since the primary storage mechanism is IMAP, all message information that was previously stored in an associated text file, AND the recording itself, is now stored in a single email message. This means that the .gsm recording will ALWAYS be attached to the message (along with the user's preference of

recording format if different - ie. .WAV). The voicemail message information is stored in the email message headers. These headers include:

```
X-Asterisk-VM-Message-Num
X-Asterisk-VM-Server-Name
X-Asterisk-VM-Context
X-Asterisk-VM-Extension
X-Asterisk-VM-Priority
X-Asterisk-VM-Caller-channel
X-Asterisk-VM-Caller-ID-Num
X-Asterisk-VM-Caller-ID-Name
X-Asterisk-VM-Duration
X-Asterisk-VM-Category
X-Asterisk-VM-Orig-date
X-Asterisk-VM-Orig-time
```

Chapter 12

SMS

12.1 Introduction

The SMS module for Asterisk was developed by Adrian Kennard, and is an implementation of the ETSI specification for landline SMS, ETSI ES 201 912, which is available from `www.etsi.org`. Landline SMS is starting to be available in various parts of Europe, and is available from BT in the UK. However, Asterisk would allow gateways to be created in other locations such as the US, and use of SMS capable phones such as the Magic Messenger. SMS works using analogue or ISDN lines.

12.2 Background

Short Message Service (SMS), or texting is very popular between mobile phones. A message can be sent between two phones, and normally contains 160 characters. There are ways in which various types of data can be encoded in a text message such as ring tones, and small graphic, etc. Text messaging is being used for voting and competitions, and also SPAM...

Sending a message involves the mobile phone contacting a message centre (SMSC) and passing the message to it. The message centre then contacts the destination mobile to deliver the message. The SMSC is responsible for storing the message and trying to send it until the destination mobile is available, or a timeout.

Landline SMS works in basically the same way. You would normally have

a suitable text capable landline phone, or a separate texting box such as a Magic Messenger on your phone line. This sends a message to a message centre your telco provides by making a normal call and sending the data using 1200 Baud FSK signaling according to the ETSI spec. To receive a message the message centre calls the line with a specific calling number, and the text capable phone answers the call and receives the data using 1200 Baud FSK signaling. This works particularly well in the UK as the calling line identity is sent before the first ring, so no phones in the house would ring when a message arrives.

12.3 Typical use with Asterisk

Sending messages from an Asterisk box can be used for a variety of reasons, including notification from any monitoring systems, email subject lines, etc.

Receiving messages to an Asterisk box is typically used just to email the messages to someone appropriate - we email and texts that are received to our direct numbers to the appropriate person. Received messages could also be used to control applications, manage competitions, votes, post items to IRC, anything.

Using a terminal such as a magic messenger, an Asterisk box could ask as a message centre sending messages to the terminal, which will beep and pop up the message (and remember 100 or so messages in its memory).

12.4 Terminology

- SMS - Short Message Service i.e. text messages
- SMSC - Short Message Service Centre The system responsible for storing and forwarding messages
- MO - Mobile Originated A message on its way from a mobile or landline device to the SMSC
- MT - Mobile Terminated A message on its way from the SMSC to the mobile or landline device
- RX - Receive A message coming in to the Asterisk box
- TX - Transmit A message going out of the Asterisk box

12.5 Sub address

When sending a message to a landline, you simply send to the landline number. In the UK, all of the mobile operators (bar one) understand sending messages to landlines and pass the messages to the BTText system for delivery to the landline.

The specification for landline SMS allows for the possibility of more than one device on a single landline. These can be configured with Sub addresses which are a single digit. To send a message to a specific device the message is sent to the landline number with an extra digit appended to the end. The telco can define a default sub address (9 in the UK) which is used when the extra digit is not appended to the end. When the call comes in, part of the calling line ID is the sub address, so that only one device on the line answers the call and receives the message.

Sub addresses also work for outgoing messages. Part of the number called by the device to send a message is its sub address. Sending from the default sub address (9 in the UK) means the message is delivered with the sender being the normal landline number. Sending from any other sub address makes the sender the landline number with an extra digit on the end.

Using Asterisk, you can make use of the sub addresses for sending and receiving messages. Using DDI (DID, i.e. multiple numbers on the line on ISDN) you can also make use of many different numbers for SMS.

12.6 extensions.conf

The following contexts are recommended.

```
; Mobile Terminated, RX. This is used when an incoming call from the SMS arrive
s, with the queue (called number and sub address) in ${EXTEN}
; Running an app after receipt of the text allows the app to find all messages
in the queue and handle them, e.g. email them.
; The app may be something like   smsq --process=somecommand --queue=${EXTEN}
to run a command for each received message
; See below for usage
[smsmtrx]
exten = _X.,1, SMS(${EXTEN},a)
exten = _X.,2,System("someapptohandleincomingsms ${EXTEN}")
exten = _X.,3,Hangup
; Mobile originated, RX. This is receiving a message from a device, e.g.
; a Magic Messenger on a sip extension
; Running an app after receipt of the text allows the app to find all messages
; in the queue and handle then, e.g. sending them to the public SMSC
; The app may be something like   smsq --process=somecommand --queue=${EXTEN}
```

```
; to run a command for each received message
; See below for example usage
[smsmorx]
exten = _X.,1, SMS(${EXTEN},sa)
exten = _X.,2,System("someapptohandlelocalsms ${EXTEN}")
exten = _X.,3,Hangup
```

smsmtrx is normally accessed by an incoming call from the SMSC. In the UK this call is from a CLI of 080058752X0 where X is the sub address. As such a typical usage in the extensions.conf at the point of handling an incoming call is:

```
exten = _X./8005875290,1,Goto(smsmtrx,${EXTEN},1)
exten = _X./_80058752[0-8]0,1,Goto(smsmtrx,${EXTEN}-${CALLERID(num):8:1},1)
```

Alternatively, if you have the correct national prefix on incoming CLI, e.g. using dahdi_hfc, you might use:

```
exten = _X./08005875290,1,Goto(smsmtrx,${EXTEN},1)
exten = _X./_080058752[0-8]0,1,Goto(smsmtrx,${EXTEN}-${CALLERID(num):9:1},1)
```

smsmorx is normally accessed by a call from a local sip device connected to a Magic Messenger. It could however by that you are operating Asterisk as a message centre for calls from outside. Either way, you look at the called number and goto smsmorx. In the UK, the SMSC number that would be dialed is 1709400X where X is the caller sub address. As such typical usage in extension.config at the point of handling a call from a sip phone is:

```
exten = 17094009,1,Goto(smsmorx,${CALLERID(num)},1)
exten = _1709400[0-8],1,Goto(smsmorx,${CALLERID(num)}-{EXTEN:7:1},1)
```

12.7 Using smsq

smsq is a simple helper application designed to make it easy to send messages from a command line. it is intended to run on the Asterisk box and have direct access to the queue directories for SMS and for Asterisk.

In its simplest form you can send an SMS by a command such as smsq 0123456789 This is a test to 0123456789 This would create a queue file for a mobile originated TX message in queue 0 to send the text "This is a test to 0123456789" to 0123456789. It would then place a file in the `/var/spool/asterisk/outgoing` directory to initiate a call to 17094009 (the default message centre in smsq) attached to application SMS with argument of the queue name (0).

Normally smsq will queue a message ready to send, and will then create a file in the Asterisk outgoing directory causing Asterisk to actually connect to the message centre or device and actually send the pending message(s).

Using `--process`, smsq can however be used on received queues to run a command for each file (matching the queue if specified) with various environment variables set based on the message (see below); smsq options:

```
--help
Show help text
--usage
Show usage
--queue
-q
Specify a specific queue
In no specified, messages are queued under queue "0"
--da
-d
Specify destination address
--oa
-o
Specify originating address
This also implies that we are generating a mobile terminated message
--ud
-m
Specify the actual message
--ud-file
-f
Specify a file to be read for the context of the message
A blank filename (e.g. --ud-file= on its own) means read stdin. Very
useful when using via ssh where command line parsing could mess up the
message.
--mt
-t
Mobile terminated message to be generated
--mo
Mobile originated message to be generated
Default
--tx
```

```
Transmit message
Default
--rx
-r
Generate a message in the receive queue
--UTF-8
Treat the file as UTF-8 encoded (default)
--UCS-1
Treat the file as raw 8 bit UCS-1 data, not UTF-8 encoded
--UCS-2
Treat the file as raw 16 bit bigendian USC-2 data
--process
Specific a command to process for each file in the queue
Implies --rx and --mt if not otherwise specified.
Sets environment variables for every possible variable, and also ud,
ud8 (USC-1 hex), and ud16 (USC-2 hex) for each call. Removes files.
--motx-channel
Specify the channel for motx calls
May contain X to use sub address based on queue name or may be full
number
Default is Local/1709400X
--motx-callerid
Specify the caller ID for motx calls
The default is the queue name without -X suffix
--motx-wait
Wait time for motx call
Default 10
--motx-delay
Retry time for motx call
Default 1
--motx-retries
Retries for motx call
Default 10
--mttx-channel
Specify the channel for mttx calls
Default is Local/ and the queue name without -X suffix
--mtttx-callerid
Specify the callerid for mttx calls
```

```
May include X to use sub address based on queue name or may be full
number
Default is 080058752X0
--mttx-wait
Wait time for mttx call
Default 10
--mttx-delay
Retry time for mttx call
Default 30
--mttx-retries
Retries for mttx call
Default 100
--default-sub-address
The default sub address assumed (e.g. for X in CLI and dialled numbers
as above) when none added (-X) to queue
Default 9
--no-dial
-x
Create queue, but do not dial to send message
--no-wait
Do not wait if a call appears to be in progress
This could have a small window where a message is queued but not
sent, so regular calls to smsq should be done to pick up any missed
messages
--concurrent
How many concurrent calls to allow (per queue), default 1
--mr
-n
Message reference
--pid
-p
Protocol ID
--dcs
Data coding scheme
--udh
Specific hex string of user data header specified (not including the
initial length byte)
May be a blank string to indicate header is included in the user data
```

```
already but user data header indication to be set.
--srr
Status report requested
--rp
Return path requested
--vp
Specify validity period (seconds)
--scts
Specify timestamp (YYYY-MM-DDTHH:MM:SS)
--spool-dir
Spool dir (in which sms and outgoing are found)
Default /var/spool/asterisk
```

Other arguments starting '-' or '--' are invalid and will cause an error. Any trailing arguments are processed as follows:-

- If the message is mobile originating and no destination address has been specified, then the first argument is assumed to be a destination address
- If the message is mobile terminating and no destination address has been specified, then the first argument is assumed to be the queue name
- If there is no user data, or user data file specified, then any following arguments are assumed to be the message, which are concatenated.
- If no user data is specified, then no message is sent. However, unless `--no-dial` is specified, smsq checks for pending messages and generates an outgoing anyway

Note that when smsq attempts to make a file in `/var/spool/asterisk/outgoing`, it checks if there is already a call queued for that queue. It will try several filenames, up to the `--concurrent` setting. If these files exist, then this means Asterisk is already queued to send all messages for that queue, and so Asterisk should pick up the message just queued. However, this alone could create a race condition, so if the files exist then smsq will wait up to 3 seconds to confirm it still exists or if the queued messages have been sent already. The `--no-wait` turns off this behaviour. Basically, this

means that if you have a lot of messages to send all at once, Asterisk will not make unlimited concurrent calls to the same message centre or device for the same queue. This is because it is generally more efficient to make one call and send all of the messages one after the other.

smsq can be used with no arguments, or with a queue name only, and it will check for any pending messages and cause an outgoing if there are any. It only sets up one outgoing call at a time based on the first queued message it finds. A outgoing call will normally send all queued messages for that queue. One way to use smsq would be to run with no queue name (so any queue) every minute or every few seconds to send pending message. This is not normally necessary unless `--no-dial` is selected. Note that smsq does only check motx or mttx depending on the options selected, so it would need to be called twice as a general check.

UTF-8 is used to parse command line arguments for user data, and is the default when reading a file. If an invalid UTF-8 sequence is found, it is treated as UCS-1 data (i.e, as is). The `--process` option causes smsq to scan the specified queue (default is mtrx) for messages (matching the queue specified, or any if queue not specified) and run a command and delete the file. The command is run with a number of environment variables set as follows. Note that these are unset if not needed and not just taken from the calling environment. This allows simple processing of incoming messages

```
$queue
Set if a queue specified
$?srr
srr is set (to blank) if srr defined and has value 1.
$?rp
rp is set (to blank) if rp defined and has value 1.
$ud
User data, UTF-8 encoding, including any control characters, but with
nulls stripped out
Useful for the content of emails, for example, as it includes any
newlines, etc.
$ude
User data, escaped UTF-8, including all characters, but control
characters \n, \r, \t, \f, \xxx and \ is escaped as \\
Useful guaranteed one line printable text, so useful in Subject lines
of emails, etc
```

```
$ud8
Hex UCS-1 coding of user data (2 hex digits per character)
Present only if all user data is in range U+0000 to U+00FF
$ud16
Hex UCS-2 coding of user data (4 hex digits per character)
other
Other fields set using their field name, e.g. mr, pid, dcs, etc. udh
is a hex byte string
```

12.8 File formats

By default all queues are held in a director `/var/spool/asterisk/sms`. Within this directory are sub directories mtrx, mttx, morx, motx which hold the received messages and the messages ready to send. Also, `/var/log/asterisk/sms` is a log file of all messages handled.

The file name in each queue directory starts with the queue parameter to SMS which is normally the CLI used for an outgoing message or the called number on an incoming message, and may have -X (X being sub address) appended. If no queue ID is known, then 0 is used by smsq by default. After this is a dot, and then any text. Files are scanned for matching queue ID and a dot at the start. This means temporary files being created can be given a different name not starting with a queue (we recommend a . on the start of the file name for temp files). Files in these queues are in the form of a simple text file where each line starts with a keyword and an = and then data. udh and ud have options for hex encoding, see below.

UTF-8. The user data (ud) field is treated as being UTF-8 encoded unless the DCS is specified indicating 8 bit format. If 8 bit format is specified then the user data is sent as is. The keywords are as follows:

```
oa Originating address
The phone number from which the message came
Present on mobile terminated messages and is the CLI for morx messages
da
Destination Address
The phone number to which the message is sent
Present on mobile originated messages
scts
The service centre time stamp
```

```
Format YYYY-MM-DDTHH:MM:SS
Present on mobile terminated messages
pid
One byte decimal protocol ID
See GSM specs for more details
Normally 0 or absent
dcs
One byte decimal data coding scheme
If omitted, a sensible default is used (see below)
See GSM specs for more details
mr
One byte decimal message reference
Present on mobile originated messages, added by default if absent
srr
0 or 1 for status report request
Does not work in UK yet, not implemented in app_sms yet
rp
0 or 1 return path
See GSM specs for details
vp
Validity period in seconds
Does not work in UK yet
udh
Hex string of user data header prepended to the SMS contents,
excluding initial length byte.
Consistent with ud, this is specified as udh# rather than udh=
If blank, this means that the udhi flag will be set but any user data
header must be in the ud field
ud
User data, may be text, or hex, see below
```

udh is specified as as udh# followed by hex (2 hex digits per byte). If present, then the user data header indicator bit is set, and the length plus the user data header is added to the start of the user data, with padding if necessary (to septet boundary in 7 bit format). User data can hold an USC character codes U+0000 to U+FFFF. Any other characters are coded as U+FEFF

ud can be specified as ud= followed by UTF-8 encoded text if it contains

no control characters, i.e. only (U+0020 to U+FFFF). Any invalid UTF-8 sequences are treated as is (U+0080-U+00FF).

ud can also be specified as ud# followed by hex (2 hex digits per byte) containing characters U+0000 to U+00FF only.

ud can also be specified as ud## followed by hex (4 hex digits per byte) containing UCS-2 characters.

When written by app_sms (e.g. incoming messages), the file is written with ud= if it can be (no control characters). If it cannot, the a comment line ;ud= is used to show the user data for human readability and ud# or ud## is used.

12.9 Delivery reports

The SMS specification allows for delivery reports. These are requested using the srr bit. However, as these do not work in the UK yet they are not fully implemented in this application. If anyone has a telco that does implement these, please let me know. BT in the UK have a non standard way to do this by starting the message with *0#, and so this application may have a UK specific bodge in the near future to handle these.

The main changes that are proposed for delivery report handling are :

- New queues for sent messages, one file for each destination address and message reference.
- New field in message format, user reference, allowing applications to tie up their original message with a report.
- Handling of the delivery confirmation/rejection and connecting to the outgoing message - the received message file would then have fields for the original outgoing message and user reference allowing applications to handle confirmations better.

Chapter 13

Queues

13.1 Introduction

Pardon, but the dialplan in this tutorial will be expressed in AEL, the new Asterisk Extension Language. If you are not used to its syntax, we hope you will find it to some degree intuitive. If not, there are documents explaining its syntax and constructs.

13.2 Configuring Call Queues

13.2.1 queues.conf

First of all, set up call queues in queue.conf

Here is an example:

```
========== queues.conf ==========
| ; Cool Digium Queues           |
| [general]                      |
| persistentmembers = yes        |
|                                |
| ; General sales queue          |
| [sales-general]                |
| music=default                  |
| context=sales                  |
| strategy=ringall               |
| joinempty=strict               |
| leavewhenempty=strict          |
|                                |
| ; Customer service queue       |
| [customerservice]              |
```

```
| music=default                  |
| context=customerservice        |
| strategy=ringall               |
| joinempty=strict               |
| leavewhenempty=strict          |
|                                |
| ; Support dispatch queue       |
| [dispatch]                     |
| music=default                  |
| context=dispatch               |
| strategy=ringall               |
| joinempty=strict               |
| leavewhenempty=strict          |
==================================
```

In the above, we have defined 3 separate calling queues: sales-general, customerservice, and dispatch.

Please note that the sales-general queue specifies a context of "sales", and that customerservice specifies the context of "customerservice", and the dispatch queue specifies the context "dispatch". These three contexts must be defined somewhere in your dialplan. We will show them after the main menu below.

In the [general] section, specifying the persistentmembers=yes, will cause the agent lists to be stored in astdb, and recalled on startup.

The strategy=ringall will cause all agents to be dialed together, the first to answer is then assigned the incoming call.

"joinempty" set to "strict" will keep incoming callers from being placed in queues where there are no agents to take calls. The Queue() application will return, and the dial plan can determine what to do next.

If there are calls queued, and the last agent logs out, the remaining incoming callers will immediately be removed from the queue, and the Queue() call will return, IF the "leavewhenempty" is set to "strict".

13.2.2 Routing incoming Calls to Queues

Then in extensions.ael, you can do these things:

The Main Menu

At Digium, incoming callers are sent to the "mainmenu" context, where they are greeted, and directed to the numbers they choose...

```
context mainmenu {
```

```
includes {
digium;
queues-loginout;
}

        0 => goto dispatch,s,1;
        2 => goto sales,s,1;
        3 => goto customerservice,s,1;
        4 => goto dispatch,s,1;

        s => {
                Ringing();
                Wait(1);
                Set(attempts=0);
                Answer();
                Wait(1);
                Background(digium/ThankYouForCallingDigium);
                Background(digium/YourOpenSourceTelecommunicationsSupplier);
                WaitExten(0.3);
        repeat:
                Set(attempts=$[${attempts} + 1]);
                Background(digium/IfYouKnowYourPartysExtensionYouMayDialItAtAnyTime);
                WaitExten(0.1);
                Background(digium/Otherwise);
                WaitExten(0.1);
                Background(digium/ForSalesPleasePress2);
                WaitExten(0.2);
                Background(digium/ForCustomerServicePleasePress3);
                WaitExten(0.2);
                Background(digium/ForAllOtherDepartmentsPleasePress4);
                WaitExten(0.2);
                Background(digium/ToSpeakWithAnOperatorPleasePress0AtAnyTime);
                if( ${attempts} < 2 ) {
                        WaitExten(0.3);
                        Background(digium/ToHearTheseOptionsRepeatedPleaseHold);
                }
                WaitExten(5);
                if( ${attempts} < 2 ) goto repeat;
                Background(digium/YouHaveMadeNoSelection);
                Background(digium/ThisCallWillBeEnded);
                Background(goodbye);
                Hangup();
        }
}
```

The Contexts referenced from the queues.conf file

```
context sales {

        0 => goto dispatch,s,1;
        8 => Voicemail(${SALESVM});

        s => {
                Ringing();
                Wait(2);
```

```
                Background(digium/ThankYouForContactingTheDigiumSalesDepartment);
                WaitExten(0.3);
                Background(digium/PleaseHoldAndYourCallWillBeAnsweredByOurNextAvailableSalesRepresentative);
                WaitExten(0.3);
                Background(digium/AtAnyTimeYouMayPress0ToSpeakWithAnOperatorOr8ToLeaveAMessage);
                Set(CALLERID(name)=Sales);
                Queue(sales-general,t);
                Set(CALLERID(name)=EmptySalQ);
                goto dispatch,s,1;
                Playback(goodbye);
                Hangup();
        }
}
```

Please note that there is only one attempt to queue a call in the sales queue. All sales agents that are logged in will be rung.

```
context customerservice {

        0 => {
                SetCIDName(CSVTrans);
                goto dispatch|s|1;
        }
        8 => Voicemail(${CUSTSERVVM});

        s => {
                Ringing();
                Wait(2);
                Background(digium/ThankYouForCallingDigiumCustomerService);
                WaitExten(0.3);
        notracking:
                Background(digium/PleaseWaitForTheNextAvailableCustomerServiceRepresentative);
                WaitExten(0.3);
                Background(digium/AtAnyTimeYouMayPress0ToSpeakWithAnOperatorOr8ToLeaveAMessage);
                Set(CALLERID(name)=Cust Svc);
                Set(QUEUE_MAX_PENALTY=10);
                Queue(customerservice,t);
                Set(QUEUE_MAX_PENALTY=0);
                Queue(customerservice,t);
                Set(CALLERID(name)=EmptyCSVQ);
                goto dispatch,s,1;
                Background(digium/NoCustomerServiceRepresentativesAreAvailableAtThisTime);
                Background(digium/PleaseLeaveAMessageInTheCustomerServiceVoiceMailBox);
                Voicemail(${CUSTSERVVM});
                Playback(goodbye);
                Hangup();
        }
}
```

Note that calls coming into customerservice will first be try to queue calls to those agents with a QUEUE_MAX_PENALTY of 10, and if none are available, then all agents are rung.

```
context dispatch
{

        s => {
                Ringing();
                Wait(2);
                Background(digium/ThankYouForCallingDigium);
                WaitExten(0.3);
                Background(digium/YourCallWillBeAnsweredByOurNextAvailableOperator);
                Background(digium/PleaseHold);
                Set(QUEUE_MAX_PENALTY=10);
                Queue(dispatch|t);
                Set(QUEUE_MAX_PENALTY=20);
                Queue(dispatch|t);
                Set(QUEUE_MAX_PENALTY=0);
                Queue(dispatch|t);
                Background(digium/NoOneIsAvailableToTakeYourCall);
                Background(digium/PleaseLeaveAMessageInOurGeneralVoiceMailBox);
                Voicemail(${DISPATCHVM});
                Playback(goodbye);
                Hangup();
        }
}
```

And in the dispatch context, first agents of priority 10 are tried, then 20, and if none are available, all agents are tried.

Notice that a common pattern is followed in each of the three queue contexts:

First, you set QUEUE_MAX_PENALTY to a value, then you call Queue(<queue-name>,option,...) (see the Queue application documetation for details)

In the above, note that the "t" option is specified, and this allows the agent picking up the incoming call the luxury of transferring the call to other parties.

The purpose of specifying the QUEUE_MAX_PENALTY is to develop a set of priorities amongst agents. By the above usage, agents with lower number priorities will be given the calls first, and then, if no-one picks up the call, the QUEUE_MAX_PENALTY will be incremented, and the queue tried again. Hopefully, along the line, someone will pick up the call, and the Queue application will end with a hangup.

The final attempt to queue in most of our examples sets the QUEUE_MAX_PENALTY to zero, which means to try all available agents.

13.2.3 Assigning agents to Queues

In this example dialplan, we want to be able to add and remove agents to handle incoming calls, as they feel they are available. As they log in,

they are added to the queue's agent list, and as they log out, they are removed. If no agents are available, the queue command will terminate, and it is the duty of the dialplan to do something appropriate, be it sending the incoming caller to voicemail, or trying the queue again with a higher QUEUE_MAX_PENALTY.

Because a single agent can make themselves available to more than one queue, the process of joining multiple queues can be handled automatically by the dialplan.

Agents Log In and Out

```
context queues-loginout
{
        6092 => {
                        Answer();
                        Read(AGENT_NUMBER,agent-enternum);
                        VMAuthenticate(${AGENT_NUMBER}@default,s);
                        Set(queue-announce-success=1);
                        goto queues-manip,I${AGENT_NUMBER},1;
                }

        6093 => {
                        Answer();
                        Read(AGENT_NUMBER,agent-enternum);
                        Set(queue-announce-success=1);
                        goto queues-manip,O${AGENT_NUMBER},1;
                }
}
```

In the above contexts, the agents dial 6092 to log into their queues, and they dial 6093 to log out of their queues. The agent is prompted for their agent number, and if they are logging in, their passcode, and then they are transferred to the proper extension in the queues-manip context. The queues-manip context does all the actual work:

```
context queues-manip {

        // Raquel Squelch
        _[IO]6121 => {
                &queue-addremove(dispatch,10,${EXTEN});
                &queue-success(${EXTEN});
        }

        // Brittanica Spears
        _[IO]6165 => {
                &queue-addremove(dispatch,20,${EXTEN});
                &queue-success(${EXTEN});
        }
```

```
        // Rock Hudson
        _[IO]6170 => {
                &queue-addremove(sales-general,10,${EXTEN});
                &queue-addremove(customerservice,20,${EXTEN});
                &queue-addremove(dispatch,30,${EXTEN});
                &queue-success(${EXTEN});
        }

        // Saline Dye-on
        _[IO]6070 => {
                &queue-addremove(sales-general,20,${EXTEN});
                &queue-addremove(customerservice,30,${EXTEN});
                &queue-addremove(dispatch,30,${EXTEN});
                &queue-success(${EXTEN});
        }
}
```

In the above extensions, note that the queue-addremove macro is used to actually add or remove the agent from the applicable queue, with the applicable priority level. Note that agents with a priority level of 10 will be called before agents with levels of 20 or 30.

In the above example, Raquel will be dialed first in the dispatch queue, if she has logged in. If she is not, then the second call of Queue() with priority of 20 will dial Brittanica if she is present, otherwise the third call of Queue() with MAX_PENALTY of 0 will dial Rock and Saline simultaneously.

Also note that Rock will be among the first to be called in the sales-general queue, and among the last in the dispatch queue. As you can see in main menu, the callerID is set in the main menu so they can tell which queue incoming calls are coming from.

The call to queue-success() gives some feedback to the agent as they log in and out, that the process has completed.

```
macro queue-success(exten)
{
        if( ${queue-announce-success} > 0 )
        {
                switch(${exten:0:1})
                {
                case I:
                        Playback(agent-loginok);
                        Hangup();
                        break;
                case O:
                        Playback(agent-loggedoff);
                        Hangup();
                        break;
                }
        }
}
```

The queue-addremove macro is defined in this manner:

```
macro queue-addremove(queuename,penalty,exten)
{
        switch(${exten:0:1})
        {
        case I:  // Login
                 AddQueueMember(${queuename},Local/${exten:1}@agents,${penalty});
 break;
        case O:  // Logout
                 RemoveQueueMember(${queuename},Local/${exten:1}@agents);
 break;
        case P:  // Pause
                 PauseQueueMember(${queuename},Local/${exten:1}@agents);
 break;
        case U:  // Unpause
                 UnpauseQueueMember(${queuename},Local/${exten:1}@agents);
 break;
        default: // Invalid
                 Playback(invalid);
                 break;
        }
}
```

Basically, it uses the first character of the exten variable, to determine the proper actions to take. In the above dial plan code, only the cases I or O are used, which correspond to the Login and Logout actions.

13.2.4 Controlling The Way Queues Call the Agents

Notice in the above, that the commands to manipulate agents in queues have "@agents" in their arguments. This is a reference to the agents context:

```
context agents
{
// General sales queue
8010 =>
{
Set(QUEUE_MAX_PENALTY=10);
Queue(sales-general,t);
Set(QUEUE_MAX_PENALTY=0);
Queue(sales-general,t);
Set(CALLERID(name)=EmptySalQ);
goto dispatch,s,1;
}
// Customer Service queue
8011 =>
{
Set(QUEUE_MAX_PENALTY=10);
Queue(customerservice,t);
Set(QUEUE_MAX_PENALTY=0);
Queue(customerservice,t);
```

```
Set(CALLERID(name)=EMptyCSVQ);
goto dispatch,s,1;
}
8013 =>
{
Dial(iax2/sweatshop/9456@from-ecstacy);

Set(CALLERID(name)=EmptySupQ);
Set(QUEUE_MAX_PENALTY=10);
Queue(support-dispatch,t);
Set(QUEUE_MAX_PENALTY=20);
Queue(support-dispatch,t);
Set(QUEUE_MAX_PENALTY=0); // means no max
Queue(support-dispatch,t);
goto dispatch,s,1;
}
6121 => &callagent(${RAQUEL},${EXTEN});
6165 => &callagent(${SPEARS},${EXTEN});
6170 => &callagent(${ROCK},${EXTEN});
6070 => &callagent(${SALINE},${EXTEN});
}
```

In the above, the variables ${RAQUEL}, etc stand for actual devices to ring that person's phone (like DAHDI/37).

The 8010, 8011, and 8013 extensions are purely for transferring incoming callers to queues. For instance, a customer service agent might want to transfer the caller to talk to sales. The agent only has to transfer to extension 8010, in this case.

Here is the callagent macro, note that if a person in the queue is called, but does not answer, then they are automatically removed from the queue.

```
macro callagent(device,exten)
{
if( ${GROUP_COUNT(${exten}@agents)}=0 )
{
Set(OUTBOUND_GROUP=${exten}@agents);
Dial(${device},300,t);
switch(${DIALSTATUS})
{
case BUSY:
Busy();
break;
case NOANSWER:
Set(queue-announce-success=0);
goto queues-manip,O${exten},1;
default:
Hangup();
break;
}
}
else
{
Busy();
```

```
}
}
```

In the callagent macro above, the ${exten} will be 6121, or 6165, etc, which is the extension of the agent.

The use of the GROUP_COUNT, and OUTBOUND_GROUP follow this line of thinking. Incoming calls can be queued to ring all agents in the current priority. If some of those agents are already talking, they would get bothersome call-waiting tones. To avoid this inconvenience, when an agent gets a call, the OUTBOUND_GROUP assigns that conversation to the group specified, for instance 6171@agents. The ${GROUP_COUNT()} variable on a subsequent call should return "1" for that group. If GROUP_COUNT returns 1, then the busy() is returned without actually trying to dial the agent.

13.2.5 Pre Acknowledgement Message

If you would like to have a pre acknowledge message with option to reject the message you can use the following dialplan Macro as a base with the 'M' dial argument.

```
[macro-screen]
exten=>s,1,Wait(.25)
exten=>s,2,Read(ACCEPT,screen-callee-options,1)
exten=>s,3,Gotoif($[${ACCEPT} = 1] ?50)
exten=>s,4,Gotoif($[${ACCEPT} = 2] ?30)
exten=>s,5,Gotoif($[${ACCEPT} = 3] ?40)
exten=>s,6,Gotoif($[${ACCEPT} = 4] ?30:30)
exten=>s,30,Set(MACRO_RESULT=CONTINUE)
exten=>s,40,Read(TEXTEN,custom/screen-exten,)
exten=>s,41,Gotoif($[${LEN(${TEXTEN})} = 3]?42:45)
exten=>s,42,Set(MACRO_RESULT=GOTO:from-internal^${TEXTEN}^1)
exten=>s,45,Gotoif($[${TEXTEN} = 0] ?46:4)
exten=>s,46,Set(MACRO_RESULT=CONTINUE)
exten=>s,50,Playback(after-the-tone)
exten=>s,51,Playback(connected)
exten=>s,52,Playback(beep)
```

13.2.6 Caveats

In the above examples, some of the possible error checking has been omitted, to reduce clutter and make the examples clearer.

13.3 Queue Logs

In order to properly manage ACD queues, it is important to be able to keep track of details of call setups and teardowns in much greater detail than traditional call detail records provide. In order to support this, extensive and detailed tracing of every queued call is stored in the queue log, located (by default) in `/var/log/asterisk/queue_log`.

These are the events (and associated information) in the queue log:

ABANDON(position|origposition|waittime)

The caller abandoned their position in the queue. The position is the caller's position in the queue when they hungup, the origposition is the original position the caller was when they first entered the queue, and the waittime is how long the call had been waiting in the queue at the time of disconnect.

AGENTDUMP

The agent dumped the caller while listening to the queue announcement.

AGENTLOGIN(channel)

The agent logged in. The channel is recorded.

AGENTCALLBACKLOGIN(exten@context)

The callback agent logged in. The login extension and context is recorded.

AGENTLOGOFF(channel|logintime)

The agent logged off. The channel is recorded, along with the total time the agent was logged in.

AGENTCALLBACKLOGOFF(exten@context|logintime|reason)

The callback agent logged off. The last login extension and context is recorded, along with the total time the agent was logged in, and the reason for the logoff if it was not a normal logoff (e.g., Autologoff, Chanunavail)

COMPLETEAGENT(holdtime|calltime|origposition)

The caller was connected to an agent, and the call was terminated normally by the *agent*. The caller's hold time and the length of the call are both recorded. The caller's original position in the queue is recorded in origposition.

COMPLETECALLER(holdtime|calltime|origposition)

The caller was connected to an agent, and the call was terminated normally by the *caller*. The caller's hold time and the length of the call are both recorded. The caller's original position in the queue is recorded in origposition.

CONFIGRELOAD

The configuration has been reloaded (e.g. with asterisk -rx reload)

CONNECT(holdtime|bridgedchanneluniqueid|ringtime)

The caller was connected to an agent. Hold time represents the amount of time the caller was on hold. The bridged channel unique ID contains the unique ID of the queue member channel that is taking the call. This is useful when trying to link recording filenames to a particular call in the queue. Ringtime represents the time the queue members phone was ringing prior to being answered.

ENTERQUEUE(url|callerid)

A call has entered the queue. URL (if specified) and Caller*ID are placed in the log.

EXITEMPTY(position|origposition|waittime)

The caller was exited from the queue forcefully because the queue had no reachable members and it's configured to do that to callers when there are no reachable members. The position is the caller's position in the queue when they hungup, the origposition is the original position the caller was when they first entered the queue, and the waittime is how long the call had been waiting in the queue at the time of disconnect.

EXITWITHKEY(key|position|origposition|waittime)

The caller elected to use a menu key to exit the queue. The key and the caller's position in the queue are recorded. The caller's entry position and amoutn of time waited is also recorded.

EXITWITHTIMEOUT(position|origposition|waittime)

The caller was on hold too long and the timeout expired. The position in the queue when the timeout occurred, the entry position, and the amount of time waited are logged.

QUEUESTART

The queueing system has been started for the first time this session.

RINGNOANSWER(ringtime)

After trying for ringtime ms to connect to the available queue member, the attempt ended without the member picking up the call. Bad queue member!

SYSCOMPAT

A call was answered by an agent, but the call was dropped because the channels were not compatible.

TRANSFER(extension|context|holdtime|calltime|origposition)

Caller was transferred to a different extension. Context and extension are recorded. The caller's hold time and the length of the call are both recorded, as is the caller's entry position at the time of the transfer. PLEASE remember

that transfers performed by SIP UA's by way of a reinvite may not always be caught by Asterisk and trigger off this event. The only way to be 100% sure that you will get this event when a transfer is performed by a queue member is to use the built-in transfer functionality of Asterisk.

Chapter 14

Phone Provisioning

14.1 Introduction

Asterisk includes basic phone provisioning support through the res_phoneprov module. The current implementation is based on a templating system using Asterisk dialplan function and variable substitution and obtains information to substitute into those templates from `phoneprov.conf` and `users.conf`. A profile and set of templates is provided for provisioning Polycom phones. Note that res_phoneprov is currently limited to provisioning a single user per device.

14.2 Configuration of phoneprov.conf

The configuration file, `phoneprov.conf`, is used to set up the built-in variables SEVER and SERVER_PORT, to define a default phone profile to use, and to define different phone profiles available for provisioning.

14.2.1 The [general] section

Below is a sample of the general section of `phoneprov.conf`:

```
[general]
;serveriface=eth0
;serveraddr=192.168.1.1
;serverport=5060
default_profile=polycom
```

By default, res_phoneprov will set the SERVER variable to the IP address on the server that the requesting phone uses to contact the asterisk HTTP server. The SERVER_PORT variable will default to the **bindport** setting in sip.conf.

Should the defaults be insufficient, there are two choices for overriding the default setting of the SERVER variable. If the IP address of the server is known, or the hostname resolvable by the phones, the appropriate **serveraddr** value should be set. Alternatively, the network interface that the server listens on can be set by specifying a **serveriface** and SERVER will be set to the IP address of that interface. Only one of these options should be set.

The default SERVER_PORT variable can be overridden by setting the **serverport**. If **bindport** is not set in `sip.conf` and serverport is not specified, it is set to a default value of 5060.

Any user set for auto-provisioning in users.conf without a specified profile will be assumed to belong to the profile set with **default_profile**.

14.2.2 Creating phone profiles

A phone profile is basically a list of files that a particular group of phones needs to function. For most phone types there are files that are identical for all phones (firmware, for instance) as well as a configuration file that is specific to individual phones. res_phoneprov breaks these two groups of files into static files and dynamic files, respectively. A sample profile:

```
[polycom]
staticdir => configs/
mime_type => text/xml
setvar => CUSTOM_CONFIG=/var/lib/asterisk/phoneprov/configs/custom.cfg
static_file => bootrom.ld,application/octet-stream
static_file => bootrom.ver,plain/text
static_file => sip.ld,application/octet-stream
static_file => sip.ver,plain/text
static_file => sip.cfg
static_file => custom.cfg
${TOLOWER(${MAC})}.cfg => 000000000000.cfg
${TOLOWER(${MAC})}-phone.cfg => 000000000000-phone.cfg
config/${TOLOWER(${MAC})} => polycom.xml
${TOLOWER(${MAC})}-directory.xml => 000000000000-directory.xml
```

A **static_file** is set by specifying the file name, relative to `AST\_DATA\ _DIR/phoneprov`. The mime-type of the file can optionally be specified after a comma. If **staticdir** is set, all static files will be relative to the subdirectory of AST_DATA_DIR/phoneprov specified.

Since phone-specific config files generally have file names based on phone-specifc data, dynamic filenames in res_phoneprov can be defined with Asterisk dialplan function and variable substitution. In the above example, ${TOLOWER(${MAC})}.cfg ⇒ 000000000000.cfg would define a relative URI to be served that matches the format of MACADDRESS.cfg, all lower case. A request for that file would then point to the template found at AST_DATA_DIR/phoneprov/000000000000.cfg. The template can be followed by a comma and mime-type. Notice that the dynamic filename (URI) can contain contain directories. Since these files are dynamically generated, the config file itself does not reside on the filesystem–only the template. To view the generated config file, open it in a web browser. If the config file is XML, Firefox should display it. Some browsers will require viewing the source of the page requested.

A default mime-type for the profile can be defined by setting **mimetype**. If a custom variable is required for a template, it can be specified with **setvar**. Variable substitution on this value is done while building the route list, so ${USERNAME} would expand to the username of the users.conf user that registers the dynamic filename.

NOTE: Any dialplan function that is used for generation of dynamic file names MUST be loaded before res_phoneprov. Add "preload ⇒ modulename.so" to `modules.conf` for required functions. In the example above, "preload ⇒ func_strings.so" would be required.

14.3 Configuration of users.conf

The asterisk-gui sets up extensions, SIP/IAX2 peers, and a host of other settings. User-specific settings are stored in users.conf. If the asterisk-gui is not being used, manual entries to users.conf can be made.

14.3.1 The [general] section

There are only two settings in the general section of `users.conf` that apply to phone provisioning: localextenlength which maps to template variable EXTENSION_LENGTH and **vmexten** which maps to the VOICEMAIL_EXTEN variable.

14.3.2 Invdividual Users

To enable auto-provisioning of a phone, the user in `users.conf` needs to have:

```
...
autoprov=yes
macaddress=deadbeef4dad
profile=polycom
```

The profile is optional if a **default_profile** is set in `phoneprov.conf`. The following is a sample users.conf entry, with the template variables commented next to the settings:

```
[6001]
callwaiting = yes
context = numberplan-custom-1
hasagent = no
hasdirectory = yes
hasiax = no
hasmanager = no
hassip = yes
hasvoicemail = yes
host = dynamic
mailbox = 6001
threewaycalling = yes
deletevoicemail = no
autoprov = yes
profile = polycom
canreinvite = no
nat = no
fullname = User Two ; ${DISPLAY_NAME}
secret = test ; ${SECRET}
username = 6001 ; ${USERNAME}
macaddress = deadbeef4dad ; ${MAC}
label = 6001 ; ${LABEL}
cid_number = 6001 ; ${CALLERID}
```

The variables above, are the user-specfic variables that can be substituted into dynamic filenames and config templates.

14.4 Templates

Configuration templates are a generic way to configure phones with text-based configuration files. Templates can use any loaded dialplan function and all of the variables created by `phoneprov.conf` and `users.conf`. A short example is the included 000000000000.cfg Polycom template:

```
<?xml version="1.0" standalone="yes"?>
  <APPLICATION
    APP_FILE_PATH="sip.ld"
    CONFIG_FILES="${IF($[${STAT(e|${CUSTOM_CONFIG})}] ? "custom.cfg,
")}config/${TOLOWER(${MAC})}, sip.cfg"
    MISC_FILES="" LOG_FILE_DIRECTORY=""
  />
```

This template uses dialplan functions, expressions, and a couple of variables to generate a config file to instruct the Polycom where to pull other needed config files. If a phone with MAC address 0xDEADBEEF4DAD requests this config file, and the filename that is stored in variable CUSTOM_CONFIG does not exist, then the generated output would be:

```
<?xml version="1.0" standalone="yes"?>
  <APPLICATION
    APP_FILE_PATH="sip.ld"
    CONFIG_FILES="config/deadbeef4dad, sip.cfg"
    MISC_FILES="" LOG_FILE_DIRECTORY=""
  />
```

The Polycom phone would then download both sip.cfg (which would be registered in `phoneprov.conf` as a static file) and config/deadbeef4dad (which would be registered as a dynamic file pointing to another template, polycom.xml).

res_phoneprov also registers its own dialplan function: PP_EACH_USER. This function was designed to be able to print out a particular string for each user that res_phoneprov knows about. An example use of this function is the template for a Polycom contact directory:

```
<?xml version="1.0" standalone="yes"?>
<directory>
  <item_list>
    ${PP_EACH_USER(<item><fn>%{DISPLAY_NAME}</fn><ct>%{CALLERID}</ct><bw>1</bw></item>|${MAC})}
  </item_list>
</directory>
```

PP_EACH_USER takes two arguments. The first is the string to be printed for each user. Any variables that are to be substituted need to be in the format %{VARNAME} so that Asterisk doesn't try to substitute the variable immediately before it is passed to PP_EACH_USER. The second, optional, argument is a MAC address to exclude from the list iterated over (so, in this case, a phone won't be listed in its own contact directory).

14.5 Putting it all together

Make sure that `manager.conf` has:

```
[general]
enabled = yes
webenabled = yes
```

and that `http.conf` has:

```
[general]
enabled = yes
bindaddr = 192.168.1.1 ; Your IP here ;-)
bindport = 8088 ; Or port 80 if it is the only http server running on the machine
```

With `phoneprov.conf` and `users.conf` in place, start Astersik. From the CLI, type "http show status". An example output:

```
HTTP Server Status:
Prefix: /asterisk
Server Enabled and Bound to 192.168.1.1:8088

Enabled URI's:
/asterisk/httpstatus => Asterisk HTTP General Status
/asterisk/phoneprov/... => Asterisk HTTP Phone Provisioning Tool
/asterisk/manager => HTML Manager Event Interface
/asterisk/rawman => Raw HTTP Manager Event Interface
/asterisk/static/... => Asterisk HTTP Static Delivery
/asterisk/mxml => XML Manager Event Interface

Enabled Redirects:
  None.

POST mappings:
None.
```

There should be a phoneprov URI listed. Next, from the CLI, type "phoneprov show routes" and verify that the information there is correct. An example output for Polycom phones woud look like:

```
Static routes

Relative URI                            Physical location
sip.ver                                 configs/sip.ver
sip.ld                                  configs/sip.ld
bootrom.ver                             configs/bootrom.ver
sip.cfg                                 configs/sip.cfg
bootrom.ld                              configs/bootrom.ld
custom.cfg                              configs/custom.cfg

Dynamic routes
```

Relative URI	Template
deadbeef4dad.cfg	000000000000.cfg
deadbeef4dad-directory.xml	000000000000-directory.xml
deadbeef4dad-phone.cfg	000000000000-phone.cfg
config/deadbeef4dad	polycom.xml

With the above examples, the phones would be pointed to `http://192.168.1.1:8080/asterisk/phoneprov` for pulling config files. Templates would all be placed in AST_DATA_DIR/phoneprov and static files would be placed in AST_DATA_DIR/phoneprov/configs. Examples of valid URIs would be:

- http://192.168.1.1:8080/asterisk/phoneprov/sip.cfg
- http://192.168.1.1:8080/asterisk/phoneprov/deadbeef4dad.cfg
- http://192.168.1.1:8080/asterisk/phoneprov/config/deadbeef4dad

Chapter 15

Development

15.1 Backtrace

This document is intended to provide information on how to obtain the backtraces required on the asterisk bug tracker, available at `http://bugs.digium.com`. The information is required by developers to help fix problem with bugs of any kind. Backtraces provide information about what was wrong when a program crashed; in our case, Asterisk. There are two kind of backtraces (aka 'bt') which are useful: bt and bt full.

First of all, when you start Asterisk, you MUST start it with option -g. This tells Asterisk to produce a core file if it crashes.

If you start Asterisk with the safe_asterisk script, it automatically starts using the option -g.

If you're not sure if Asterisk is running with the -g option, type the following command in your shell:

```
debian:/tmp# ps aux | grep asterisk
root     17832  0.0  1.2   2348   788 pts/1    S    Aug12   0:00 /bin/sh /usr/sbin/safe_asterisk
root     26686  0.0  2.8  15544  1744 pts/1    S    Aug13   0:02 asterisk -vvvg -c
[...]
```

The interesting information is located in the last column.

Second, your copy of Asterisk must have been built without optimization or the backtrace will be (nearly) unusable. This can be done by selecting the 'DONT_OPTIMIZE' option in the Compiler Flags submenu in the 'make menuselect' tree before building Asterisk.

After Asterisk crashes, a core file will be "dumped" in your `/tmp/` directory. To make sure it's really there, you can just type the following command in your shell:

```
debian:/tmp# ls -l /tmp/core.*
-rw-------  1 root root 10592256 Aug 12 19:40 /tmp/core.26252
-rw-------  1 root root  9924608 Aug 12 20:12 /tmp/core.26340
-rw-------  1 root root 10862592 Aug 12 20:14 /tmp/core.26374
-rw-------  1 root root  9105408 Aug 12 20:19 /tmp/core.26426
-rw-------  1 root root  9441280 Aug 12 20:20 /tmp/core.26462
-rw-------  1 root root  8331264 Aug 13 00:32 /tmp/core.26647
debian:/tmp#
```

In the event that there are multiple core files present (as in the above example), it is important to look at the file timestamps in order to determine which one you really intend to look at.

Now that we've verified the core file has been written to disk, the final part is to extract 'bt' from the core file. Core files are pretty big, don't be scared, it's normal.

NOTE: Don't attach core files on the bug tracker, we only need the bt and bt full.

For extraction, we use a really nice tool, called gdb. To verify that you have gdb installed on your system:

```
debian:/tmp# gdb -v
GNU gdb 6.3-debian
Copyright 2004 Free Software Foundation, Inc.
GDB is free software, covered by the GNU General Public License, and you are
welcome to change it and/or distribute copies of it under certain conditions.
Type "show copying" to see the conditions.
There is absolutely no warranty for GDB.  Type "show warranty" for details.
This GDB was configured as "i386-linux".
debian:/tmp#
```

Which is great, we can continue. If you don't have gdb installed, go install gdb.

Now load the core file in gdb, as follows:

```
debian:/tmp# gdb asterisk /tmp/core.26252
[...]
(You would see a lot of output here.)
[...]
Reading symbols from /usr/lib/asterisk/modules/app_externalivr.so...done.
Loaded symbols for /usr/lib/asterisk/modules/app_externalivr.so
#0  0x29b45d7e in ?? ()
(gdb)
```

Now at the gdb prompt, type: bt You would see output similar to:

```
(gdb) bt
#0  0x29b45d7e in ?? ()
#1  0x08180bf8 in ?? ()
#2  0xbcdffa58 in ?? ()
#3  0x08180bf8 in ?? ()
#4  0xbcdffa60 in ?? ()
#5  0x08180bf8 in ?? ()
#6  0x180bf894 in ?? ()
#7  0x0bf80008 in ?? ()
#8  0x180b0818 in ?? ()
#9  0x08068008 in ast_stopstream (tmp=0x40758d38) at file.c:180
#10 0x000000a0 in ?? ()
#11 0x000000a0 in ?? ()
#12 0x00000000 in ?? ()
#13 0x407513c3 in confcall_careful_stream (conf=0x8180bf8, filename=0x8181de8 "DAHDI/pseudo-1324221520") at app_meet
#14 0x40751332 in streamconfthread (args=0x8180bf8) at app_meetme.c:1965
#15 0xbcdffbe0 in ?? ()
#16 0x40028e51 in pthread_start_thread () from /lib/libpthread.so.0
#17 0x401ec92a in clone () from /lib/libc.so.6
(gdb)
```

The bt's output is the information that we need on the bug tracker.

```
Now do a bt full as follows:
(gdb) bt full
#0  0x29b45d7e in ?? ()
No symbol table info available.
#1  0x08180bf8 in ?? ()
No symbol table info available.
#2  0xbcdffa58 in ?? ()
No symbol table info available.
#3  0x08180bf8 in ?? ()
No symbol table info available.
#4  0xbcdffa60 in ?? ()
No symbol table info available.
#5  0x08180bf8 in ?? ()
No symbol table info available.
#6  0x180bf894 in ?? ()
No symbol table info available.
#7  0x0bf80008 in ?? ()
No symbol table info available.
#8  0x180b0818 in ?? ()
No symbol table info available.
#9  0x08068008 in ast_stopstream (tmp=0x40758d38) at file.c:180
No locals.
#10 0x000000a0 in ?? ()
No symbol table info available.
#11 0x000000a0 in ?? ()
No symbol table info available.
#12 0x00000000 in ?? ()
No symbol table info available.
#13 0x407513c3 in confcall_careful_stream (conf=0x8180bf8, filename=0x8181de8 "DAHDI/pseudo-1324221520") at app_meet
        f = (struct ast_frame *) 0x8180bf8
        trans = (struct ast_trans_pvt *) 0x0
#14 0x40751332 in streamconfthread (args=0x8180bf8) at app_meetme.c:1965
No locals.
```

```
#15 0xbcdffbe0 in ?? ()
No symbol table info available.
#16 0x40028e51 in pthread_start_thread () from /lib/libpthread.so.0
No symbol table info available.
#17 0x401ec92a in clone () from /lib/libc.so.6
No symbol table info available.
(gdb)
```

We also need gdb's output. That output gives more details compared to the simple "bt". So we recommend that you use bt full instead of bt. But, if you could include both, we appreciate that.

The final "extraction" would be to know all traces by all threads. Even if asterisk runs on the same thread for each call, it could have created some new threads.

To make sure we have the correct information, just do: (gdb) thread apply all bt

```
Thread 1 (process 26252):
#0  0x29b45d7e in ?? ()
#1  0x08180bf8 in ?? ()
#2  0xbcdffa58 in ?? ()
#3  0x08180bf8 in ?? ()
#4  0xbcdffa60 in ?? ()
#5  0x08180bf8 in ?? ()
#6  0x180bf894 in ?? ()
#7  0x0bf80008 in ?? ()
#8  0x180b0818 in ?? ()
#9  0x08068008 in ast_stopstream (tmp=0x40758d38) at file.c:180
#10 0x000000a0 in ?? ()
#11 0x000000a0 in ?? ()
#12 0x00000000 in ?? ()
#13 0x407513c3 in confcall_careful_stream (conf=0x8180bf8, filename=0x8181de8 "DAHDI/pseudo-1324221520") at app_meet
#14 0x40751332 in streamconfthread (args=0x8180bf8) at app_meetme.c:1965
#15 0xbcdffbe0 in ?? ()
#16 0x40028e51 in pthread_start_thread () from /lib/libpthread.so.0
#17 0x401ec92a in clone () from /lib/libc.so.6
(gdb)
```

That output tells us crucial information about each thread.

Now, just create an output.txt file and dump your "bt full" (and/or "bt") ALONG WITH "thread apply all bt" into it.

Note: Please ATTACH your output, DO NOT paste it as a note.

And you're ready for upload on the bug tracker.

If you have questions or comments regarding this documentation, feel free to pass by the #asterisk-bugs channel on irc.freenode.net.