Asterisk Administrator Guide

Asterisk Development Team <asteriskteam@digium.com>

Sangoma and Digium Join Together FAQ

Is Asterisk going away?

A: No. Asterisk, having been around nearly 20 years, isn't going away. Asterisk is a core component of the business strategies of both the Sangoma and Digium businesses and is one of the biggest reasons for this union.

Is FreePBX going away?

A: No, FreePBX has been available under the GPL license since 2004, and continues to be available today at both git.freepbx.org and github.com/freepbx. Like Asterisk, FreePBX is a core component of the Sangoma and Digium business strategy.

Will Asterisk or FreePBX be closed-source after this change?

A: No. Both projects have been distributed under the GPL license since their inception. None of that code can ever be taken out of that license and closed up. They will always be available for anyone to use, for free.

Will the project leads for Asterisk and FreePBX change?

A: No.

Will the distribution model for Asterisk or FreePBX change?

A: No. Developers who want to use Asterisk as a toolkit will continue to be able to download a tarball directly from the Asterisk website and downloads site. Users who want something a little more high-level, with helpful admin and user tools, or who want a PBX, will still be able to download the FreePBX distribution.

Will Asterisk become RPMs only?

A: No. Asterisk will continue to be able to be downloaded as a tarball, direct from the Asterisk website and downloads site.

Will Asterisk become an ISO only?

A: No. Asterisk's long-term strategy is to increase its usefulness as a toolkit for building communications solutions. Distributing it only as an ISO runs counter to this strategy. Therefore, Asterisk will continue to be able to be downloaded as a tarball, direct from the Asterisk website and downloads site.

Will Asterisk be distributed as packages as well?

A: Asterisk itself is distributed as tarballs. Within the FreePBX Distro, Asterisk is provided as RPMs and SRPMs, as a part of the distribution itself.

Do all of the licensing grants for contributing code remain in force?

A: Yes. All existing Asterisk and FreePBX Contributors Licenses remain in force and do not need to be re-signed.

What does this mean for the future of the Asterisk and FreePBX projects?

A: Asterisk and FreePBX are open source projects with massive communities and have always been distributed under open source licenses. These business changes have no negative impact on the future of these great projects. Their licenses will continue. Tarballs and source code will continue. Support will continue. The development teams remain intact and are still focused on delivering the world's greatest platform for open source communications solutions.

Will card drivers from Sangoma and others get integrated into the DAHDI project?

A: DAHDI, as it pertains to Asterisk via chan_dahdi, works with existing Sangoma telephony cards as-is, with no changes anticipated. The extensive software support provided for Sangoma's telephony cards for applications other than Asterisk (e.g. data protocol and Windows support) mean that pulling them directly into DAHDI isn't a practicable solution at this time.

What happens to libss7 (due to Sangoma already having commercial stacks) or opepnr2?

A: libss7 has been provided as-is for a number of years. As open-source code useful for certain SS7 related tasks, it will always be available. Openr2 has been released as GPL by Sangoma for years and maintained by Sangoma and used in other Sangoma commercial products such as the Vega Gateways. Therefore, efforts needed for openr2 such as bug fixes or features will continue to be handled as they have always been.

Will there be more people made available to the Asterisk or FreePBX teams to work on these open source projects?

A: We anticipate that this announcement results in many improvements in internal efficiencies, potentially resulting in more eyeballs working on both projects.

Will you start charging for access to new versions of Asterisk (like other PBX companies)?

A: No. Asterisk has never operated under a model whereby only some types of user receive access to new versions.

What will happen to AsteriskNOW?

A: AsteriskNOW will be folded into the existing FreePBX Distro, which will continue to benefit from the resources of the combined company by ensuring that users have a stable, go to distribution for their needs. However, we understand that not everyone uses the distro and Asterisk/FreePBX will continue to be available in tarball form for people to install in their distribution of choice. Existing users of AsteriskNOW can continue to use and upgrade it.

What are the Asterisk commercial licensing implications? Does the license survive the deal?

A: All commercial license agreements maintain their existing terms. Existing licenses are not terminated upon this merger and are assigned to the new company.

What does this mean for certified Asterisk?

A: Certified Asterisk is the product for customers that require SLA-backed engineering-level support of Asterisk and will continue to be offered moving forward.

How are FreePBX and Switchvox planning on coexisting?

A: FreePBX and Switchvox service two different markets but have the same common goal of providing a full UC solution. We would expect to see some features being shared between the products as the teams work and collaborate more closely together, thereby improving software development efficiencies.

Will I still be able to get my voice prompts from Allison?

A: Yes. Allison Smith has been the voice of Asterisk and FreePBX users for the better part of two decades. You will still be able to get wonderful prompts from her for your IVR needs since both Sangoma and Digium offer purchasing of those prompts already.

If I want to engage with the Asterisk and FreePBX communities, where do I go?

A: The various entry points for users into the ecosystem are designed to best-serve the types of users entering via those means. Users with questions about Asterisk should continue to use the Asterisk Community Forums (community.asterisk.org). Users with questions about FreePBX should continue to use the FreePBX Community Forums (community.freepbx.org). Issue reports should continue to be made at the respective issue trackers: issues.asterisk.org and issues.freepbx.org. Users who prefer mailing lists can use the Asterisk mailing lists and users who prefer IRC can use the #asterisk or #freepbx IRC channels. Some of this may change in the future, but until that happens, please communicate the same way you do today.

How did Digium and Sangoma end up together after such a long period of competition?

A: Having collaborated together for years for the good of open source communications, the combined Sangoma and Digium organization will have greater financial and competitive strength to meet market challenges. This is a consolidation of the major players in the Asterisk-related ecosystem - who can now work together fully instead of competing against each other.

Will Astricon still exist? Is Astricon still happening this year?

A: Yes. Astricon is still happening in Orlando, FL from October 9-11, 2018. We hope to see you there! Register Now!

Is there any impact to trademark licensing?

A: No. Existing trademark license agreements survive this announcement.

Where should I go for questions about commercial products?

A: Contact your Sangoma and Digium account managers.

About the Project

A Brief History of the Asterisk Project

Linux Support Services

Way, way back in 1999 a young man named Mark Spencer was finishing his Computer Engineering degree at Auburn University when he hit on an interesting business concept. 1999 was the high point in the .com revolution (aka bubble), and thousands of businesses world-wide were discovering that they could save money by using the open source Linux operating system in place of proprietary operating systems. The lure of a free operating system with open access to the source code was too much to pass up. Unfortunately there was little in the way of commercial support available for Linux at that time. Mark decided to fill this gap by creating a company called "Linux Support Services". LSS offered a support hotline that IT professionals could (for a fee) call to get help with Linux.

The idea took off. Within a few months, Mark had a small office staffed with Linux experts. Within a few more months the growth of the business expanded demanded a "real" phone system that could distribute calls evenly across the support team, so Mark called up several local phone system vendors and asked for quotes. Much to his surprise, the responses all came back well above $50,000 -- far more than Mark had budgeted for the project. Far more than LSS could afford.

Finding a Solution

Rather than give in and take out a small business loan, Mark made a pivotal decision. He decided to write his own phone system. Why not? A phone system is really just a computer running phone software, right? Fortunately for us, Mark had no idea how big a project he had take on. If he had known what a massive undertaking it was to build a phone system from the ground up might have gritted his teeth, borrowed the money and spent the next decade doing Linux support. But he didn't know what he didn't know, and so he started to code. And he coded. And he coded.

Mark had done his engineering co-op at Adtran, a communications and networking device manufacturer in Huntsville, AL. There he had cut his teeth on telecommunications system development, solving difficult problems generating a prodigious amount of complex code in short time. This experience proved invaluable as he began to frame out the system which grew into Asterisk. In only a few months Mark crafted the original Asterisk core code. As soon as he had a working prototype he published the source code on the Internet, making it available under the GPL license (the same license used for Linux).

Within a few months the idea of an "open source PBX" caught on. There had been a few other open source communications projects, but none had captured the imagination of the global population of communications geeks like Asterisk. As Mark labored on the core system, hundreds (now thousands) of developers from all over the world began to submit new features and functions.

Digium

What became of Linux Support Services? In 2001, Linux Support Services changed its name to Digium. Digium continued to develop Asterisk in collaboration with the community, provide services to support the development community, as well as build commercial products and services around Asterisk which have fueled growth in both Digium and the Asterisk project. You can find out more about Digium at the Sangoma website and on wikipedia.

Asterisk in the Present

Asterisk is constantly evolving to meet the needs of the project's user-base. It's difficult to summarize the vast scope of everything that Asterisk can do as a communications toolkit. We'll list some resources that give you an idea of what is going on in the Asterisk project at present.

- Asterisk Versions :Shows release time lines, support and EOL schedules
- Roadmap section :Information from developer conferences and planning sessions
- CHANGES :A document in Asterisk trunk, shows functionality changes between major versions
- UPGRADE :A document in Asterisk trunk, shows breaking changes, deprecation of specific features and important info on upgrading.
- Mailing lists :The dev list is a great list to see what hot topics the developers are discussing in real-time.

Asterisk as a Swiss Army Knife of Telephony

What Is Asterisk?

People often tend to think of Asterisk as an "open source PBX" because that was the focus of the original development effort. But calling Asterisk a PBX is both selling it short (it is much more) and overstating it (it can be much less). It is true that Asterisk started out as a phone system for a small business (see the "Brief History" section for the juicy details) but in the decade since it was originally released it has grown into a universal tool for building communications applications. Today Asterisk powers not only IP PBX systems but also VoIP gateways, call center systems, conference bridges, voicemail servers and all kinds of other applications that involve real-time communications.

Asterisk is not a PBX but is the engine that powers PBXs. Asterisk is not an IVR but is the engine that powers IVRs. Asterisk is not a call center ACD but is the engine that powers ACD/queueing systems.

Asterisk is to communications applications what the Apache web server is to web applications. Apache is a web server. Asterisk is a communication server. Apache handles all the low-level details of sending and receiving data using the HTTP protocol. Asterisk handles all the low level details of sending and receiving data using lots of different communication protocols. When you install Apache, you have a web server but its up to you to create the web applications. When you install Asterisk, you have a communications server but its up to you to create the communications applications.

Web applications are built out of HTML pages, CSS style sheets, server-side processing scripts, images, databases, web services, etc. Asterisk communications applications are built out Dialplan scripts, configuration files, audio recordings, databases, web services, etc. For a web application to work, you need the web server connected to the Internet. For a communications application to work, you need the communications server connected to communication services (VoIP or PSTN). For people to be able to access your web site you need to register a domain name and set up DNS entries that point "www.yourdomain.com" to your server. For people to access your communications system you need phone numbers or VoIP URIs that send calls to your server.

In both cases the server is the plumbing that makes your application work. The server handles the low-level complexities and allows you, the application developer, to concentrate on the application logic and presentation. You don't have to be an expert on HTTP to create powerful web applications, and you don't have to be an expert on SIP or Q.931 to create powerful communications applications.

Here's a simple example. The following HTML script, installed on a working web server, prints "Hello World" in large type:

```
<html>
  <head>
    <title>Hello World Demo</title>
  </head>
  <body>
    <h1>Hello World!</h1>
  </body>
</html>
```

The following Dialplan script answers the phone, waits for one second, plays back "hello world" then hangs up.

```
exten => 100,1,Answer()
exten => 100,n,Wait(1)
exten => 100,n,Playback(hello-world)
exten => 100,n,Hangup()
```

In both cases the server components are handling all of the low level details of the underlying protocols. Your application doesn't have to worry about the byte alignment, the packet size, the codec or any of the thousands of other critical details that make the application work. This is the power of an engine.

Who Uses Asterisk?

Asterisk is created by communication system developers, for communication system developers. As an open source project, Asterisk is a collaboration between many different individuals and companies, all of which need a flexible communications engine to power their applications.

Asterisk Versions

There are multiple supported feature frozen releases of Asterisk. Once a release series is made available, it is supported for some period of time. During this initial support period, releases include changes to fix bugs that have been reported. At some point, the release series will be deprecated and only maintained with fixes for security issues. Finally, the release will reach its End of Life, where it will no longer receive changes of any kind.

The type of release defines how long it will be supported. A Long Term Support (LTS) release will be fully supported for 4 years, with one additional year of maintenance for security fixes. Standard releases are supported for a shorter period of time, which will be at least one year of full support and an additional year of maintenance for security fixes.

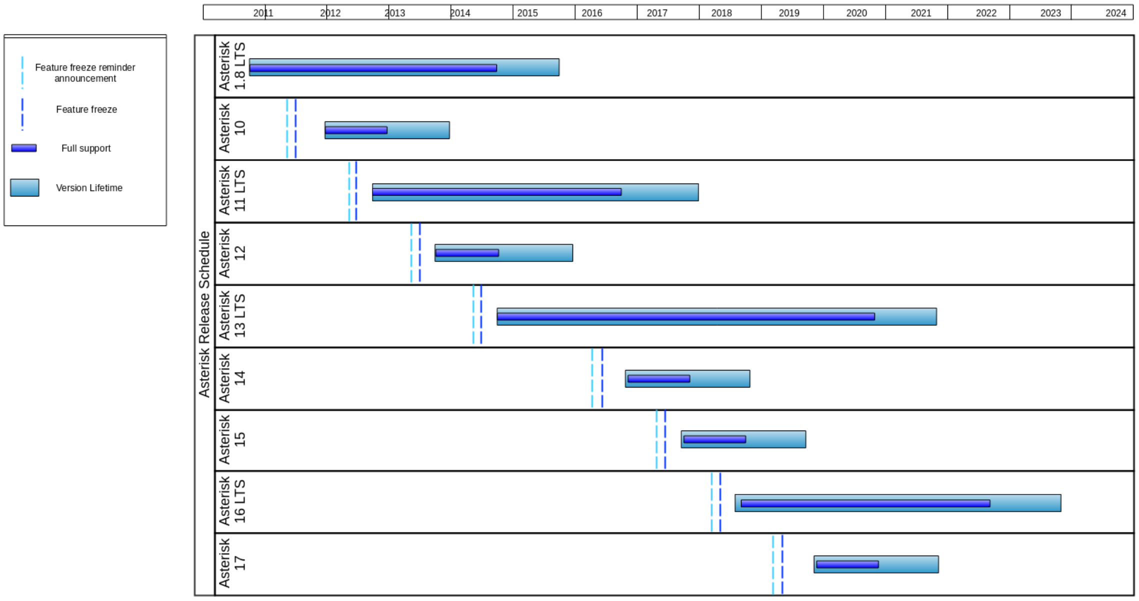
The following table shows the release time lines for all releases of Asterisk, including those that have reached End of Life.

Release Series	Release Type	Release Date	Security Fix Only	EOL
1.2.X		2005-11-21	2007-08-07	2010-11-21
1.4.X	LTS	2006-12-23	2011-04-21	2012-04-21
1.6.0.X	Standard	2008-10-01	2010-05-01	2010-10-01
1.6.1.X	Standard	2009-04-27	2010-05-01	2011-04-27
1.6.2.X	Standard	2009-12-18	2011-04-21	2012-04-21
1.8.X	LTS	2010-10-21	2014-10-21	2015-10-21
10.X	Standard	2011-12-15	2012-12-15	2013-12-15
11.x	LTS	2012-10-25	2016-10-25	2017-10-25
12.x	Standard	2013-12-20	2014-12-20	2015-12-20
13.x	LTS	2014-10-24	2020-10-24	2021-10-24
14.x	Standard	2016-09-26	2017-09-26	2018-09-26
15.x	Standard	2017-10-03	2018-10-03	2019-10-03
16.x	LTS	2018-10-09	2022-10-09	2023-10-09
17.x	Standard	2019-10-28	2020-10-28	2021-10-28

New releases of Asterisk will be made roughly once a year, alternating between standard and LTS releases. Within a given release series that is fully supported, bug fix updates are provided roughly every 4 weeks. For a release series that is receiving only maintenance for security fixes, updates are made on an as needed basis.

If you're not sure which one to use, choose either the latest release for the most up to date features, or the latest LTS release for a platform that may have less features, but will usually be around longer.

The schedule for Asterisk releases is visualized below (which is subject to change at any time):



For developers, it is useful to be aware of when the feature freeze for a particular branch will occur. The feature freeze for a branch will occur 3 months prior to the release of a new Asterisk version, and a reminder announcement will be posted to the asterisk-dev mailing list approximately 60 days prior to the feature freeze. Asterisk versions are slated to be released the 3rd Wednesday of October. The feature freeze for a branch will occur the 3rd Wednesday of July. An announcement reminder will be posted to the asterisk-dev mailing list the 3rd Wednesday of May. Feature freeze consists of the creation of two branches: One for the release series and one for the initial release. Features can continue to be placed into the release series branch according to policy but the initial release branch will be frozen.

Feature Freeze Announcement Reminder	3rd Wednesday of May
Feature Freeze of Asterisk Branch	3rd Wednesday of July
First Release of Asterisk from Branch	3rd Wednesday of October

License Information

Asterisk 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 products (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 several management/control protocols. This includes the Asterisk Manager Interface (AMI), the Asterisk Gateway Interface (AGI), and the Asterisk REST Interface (ARI). 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.

The 'Asterisk' name and logos are trademarks owned by Digium, Inc., and use of them is subject to our trademark licensing policies. If you wish to use these trademarks for purposes other than simple redistribution of Asterisk source code obtained from Digium, you should contact our licensing department to determine the necessary steps you must take. For more information on this policy, please read: https://www.sangoma.com/legal/

If you have any questions regarding our licensing policy, please contact us:

+1.877.344.4861 (via telephone in the USA)
+1.256.428.6000 (via telephone outside the USA)
+1.256.864.0464 (via FAX inside or outside the USA)
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445 Jan Davis Drive NW
Huntsville, AL 35806
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Asterisk Sounds

License information for Asterisk sounds can be found in the Voice Prompts and Music on Hold License section.

Frequently Asked Questions about Licensing

What is an open-source license?

Wikipedia has a great article on open-source licenses, and opensource.org is a pretty definitive resource.

What is the GNU General Public License?

The GPL is a specific open-source license. Reading the preamble at this link is a great introduction, and below that is the full license text.

What if I want to distribute or license Asterisk under a different license?

Digium distributes Asterisk under a multi-licensing model often referred to as Dual-licensing and is additionally made possible by a Contributors License Agreement. This allows Digium to provide Asterisk under licenses other than the GPL. Digium provides information on their alternative commercial licensing at their website.

How can I contribute to Asterisk?

Documentation, new features, bug fixes, testing, protocol and programming expertise,, and general feedback are all welcome to the project. There is an overview that points to many resources for developers, also you can see the guidelines for contribution to see how it works.

Voice Prompts and Music on Hold License

Voice Prompts

All voice prompt contributions distributed with Asterisk or available on the Asterisk downloads site are licensed as Creative Commons Attribution-Share Alike 3.0. The process for contributing sound files can be found in the Asterisk Sounds Submission Process section.

Music On Hold

The Hold (on hold) music included with the Asterisk distribution has been sourced from opsound.org which itself distributes the music under Creative Commons Attribution-ShareAlike 2.5 license.

Supported Platforms

The Asterisk software can be installed on a wide range of platforms including various Linux distributions. As a project, however, we are only able to test and support a subset of them. The Asterisk project supports 32-bit and 64-bit x86 platforms using non-end of life CentOS, RHEL, Fedora, Ubuntu, and Debian Linux distributions. Support for other platforms and Linux distributions is best effort and is provided by the community. Any changes to allow such platforms must not hinder or break the project supported Linux distributions.

> ⓘ Note that due to changes and improvements in compilers it is possible for Linux distribution upgrades to result in old versions of Asterisk no longer building. If this occurs it is recommended to upgrade to the latest supported version of Asterisk.

Getting Started

When learning Asterisk it is important to start off on the right foot, so this section of the wiki covers orientation for learning Asterisk as well as installation and a simple Hello World style tutorial. These items are foundational, as knowing how to install Asterisk right the first time and where to locate the right help resources will save you a ton of time down the road.

Those interested in Asterisk training courses and certifications may visit http://www.asterisk.org/products/training

Beginning Asterisk

Asterisk is…

- an Open Source software development project
- written in the C Programming Language
- running on Linux (or other types of Unix)
- powering Business Telephone Systems
- connecting many different Telephony protocols
- a toolkit for building many things:
    - an IP PBX with many powerful features and applications
    - VoIP Gateways
    - Conferencing systems
    - and much, much more
- supporting VoIP Phones as well as PSTN and POTS
- speaking SIP , the most common VoIP protocol, among others

YouTube Videos

- Systm 5 Episode on Asterisk (from 2006 - see Asterisk Wiki for current installation instructions)
- Official Asterisk Channel
- Asterisk 123: Intro to Asterisk from Astricon 10
- Asterisk 12 Overview from Astricon 10

Resources for understanding

- Acronyms and Terminology
    - Telephony Terminology
    - Asterisk Terms Glossary
    - Telecom Acronyms (very comprehensive)
- Telephony Protocols
    - IP Telephony Protocols Overview
    - SIP Overview
    - A Hitchhiker's Guide to SIP
- Linux & Unix
    - Linux Newbie Guide
    - Beginner Tutorials
    - Unix Beginner Tutorial
- Installing and Configuring Asterisk
    - Asterisk: The Definitive Guide 3rd Edition
    - The Asterisk Wiki
- C Programming
    - C Programming Tutorial
    - Interactive C Tutorial
    - C Programming Quick Guide

Where to get help

- Email Lists and Live Chat (IRC)
    - Asterisk Mailing List and IRC
- Web Discussion Forums
    - Asterisk Community Forums
- Online Community
    - Voip Users Conference main site and on Google+

Avoiding obsolete or incorrect information

When reading about Asterisk on the web, you may come across old or incorrect information.

- Check which version of Asterisk is mentioned. There are significant changes in every version.
- Check the published date of the article if the Asterisk version isn't provided.
- Take things with a grain of salt until checked with another resource or proven correct through your own testing.
- Refer to the Asterisk Wiki and the Official Asterisk Youtube Channel for the most accurate and up to date details on the specific version of Asterisk you are using.

Please note that it is always possible that even the official documentation does not match what is written into the source code itself. If you find something lacking or incorrect in the Asterisk documentation, please communicate it through comments on the Asterisk Wiki or by filing an issue through the Asterisk Issues Tracker .

Installing Asterisk

Now that you know a bit about Asterisk and how it is used, it's time to get you up and running with your own Asterisk installation. There are various ways to get started with Asterisk on your own system:

- Install FreePBX, the Asterisk-based distribution. This takes care of installing Linux, Asterisk, and a web-based management interface all at the same time. FreePBX is the easiest way to get started if you're new to Linux and/or Asterisk.
- If you're already familiar with Linux or Unix, you can simply install packages for Asterisk and its related tools using the package manager in your operating system. We'll cover this in more detail below in Alternate Install Methods.
- For the utmost in control of your installation, you can compile and install Asterisk (and its related tools) from source code. We'll explain how to do this in Installing Asterisk From Source.

Installing Asterisk From Source

One popular option for installing Asterisk is to download the source code and compile it yourself. While this isn't as easy as using package management or using an Asterisk-based Linux distribution, it does let you decide how Asterisk gets built, and which Asterisk modules are built.

In this section, you'll learn how to download and compile the Asterisk source code, and get Asterisk installed.

What to Download?

Asterisk

Downloads of Asterisk are available at https://downloads.asterisk.org/pub/telephony/asterisk/. The currently supported versions of Asterisk will each have a symbolic link to their related release on this server, named **asterisk-{version}-current.tar.gz**. All releases ever made for the Asterisk project are available at https://downloads.asterisk.org/pub/telephony/asterisk/releases/.

The currently supported versions of Asterisk are documented on the Asterisk Versions page. It is highly recommended that you install one of the currently supported versions, as these versions continue to receive bug and security fixes.

> **Which version should I install?**
>
> - If you want a rock solid communications framework, choose the latest **Long Term Support (LTS)** release.
> - If you want the latest cool features and capabilities, choose the latest release of Asterisk. If that is a **Standard** release, note that these releases may have larger changes made in them than LTS releases.
>
> Unless otherwise noted, for the purposes of this section we will assume that Asterisk 14 is being installed.

Review Asterisk's System Requirements in order to determine what needs to be installed for the version of Asterisk you are installing. While Asterisk will look for any missing system requirements during compilation, it's often best to install these prior to configuring and compiling Asterisk.

Asterisk does come with a script, **install_prereq**, to aid in this process. If you'd like to use this script, download Asterisk first, then see Checking Asterisk Requirements for instructions on using this script to install prerequisites for your version of Asterisk.

On this Page

Downloading Asterisk

Browse to https://downloads.asterisk.org/pub/telephony/asterisk, select asterisk-14-current.tar.gz, and save the file on your file system.

You can also get the latest releases from the downloads page on asterisk.org.

Alternatively, you can use `wget` to retrieve the latest release:

```
[root@server:/usr/local/src]# wget https://downloads.asterisk.org/pub/telephony/asterisk/asterisk-14-current.tar.gz
--2017-04-28 15:45:36--  https://downloads.asterisk.org/pub/telephony/asterisk/asterisk-14-current.tar.gz
Resolving downloads.asterisk.org (downloads.asterisk.org)... 76.164.171.238
Connecting to downloads.asterisk.org (downloads.asterisk.org)|76.164.171.238|:443... connected.
HTTP request sent, awaiting response... 200 OK
Length: 40692588 (39M) [application/x-gzip]
Saving to: ‘asterisk-14-current.tar.gz’

asterisk-14-current.tar.gz            100%[======================================================================>]  38.81M
3.32MB/s    in 12s

2017-04-28 15:45:47 (3.37 MB/s) - ‘asterisk-14-current.tar.gz’ saved [40692588/40692588]
```

Other Projects

libpri

The **libpri** library allows Asterisk to communicate with ISDN connections.You'll only need this if you are going to use DAHDI with ISDN interface hardware (such as T1/E1/J1/BRI cards).

DAHDI

The **DAHDI** library allows Asterisk to communicate with analog and digital telephones and telephone lines, including connections to the Public Switched Telephone Network, or PSTN.

DAHDI stands for Digium Asterisk Hardware Device Interface, and is a set of drivers and utilities for a number of analog and digital telephony cards, such as those manufactured by Digium. The DAHDI drivers are independent of Asterisk, and can be used by other applications. DAHDI was previously called Zaptel, as it evolved from the Zapata Telephony Project.

The DAHDI code can be downloaded as individual pieces (**dahdi-linux** for the DAHDI drivers, and **dahdi-tools** for the DAHDI utilities. They can also be downloaded as a complete package called **dahdi-linux-complete**, which contains both the Linux drivers and the utilities.

You will only need to install DAHDI if you are going to utilize DAHDI compatible analog or digital telephony interface boards.

✓ **Why is DAHDI split into different pieces?**
DAHDI has been split into two pieces (the Linux drivers and the tools) as third parties have begun porting the DAHDI drivers to other operating systems, such as FreeBSD. Eventually, we may have dahdi-linux, dahdi-freebsd, and so on.

Download Locations

Project	Location
Asterisk	https://downloads.asterisk.org/pub/telephony/asterisk/asterisk-14-current.tar.gz
libpri	https://downloads.asterisk.org/pub/telephony/libpri/libpri-current.tar.gz
dahdi-linux	https://downloads.asterisk.org/pub/telephony/dahdi-linux/dahdi-linux-current.tar.gz
dahdi-tools	https://downloads.asterisk.org/pub/telephony/dahdi-tools/dahdi-tools-current.tar.gz
dahdi-complete	https://downloads.asterisk.org/pub/telephony/dahdi-linux-complete/dahdi-linux-complete-current.tar.gz

Untarring the Source

When you download the source for libpri, DAHDI, and Asterisk you'll typically end up with files with a .tar.gz or .tgz file extension. These files are affectionately known as tarballs. The name comes from the tar Unix utility, which stands for tape archive. A tarball is a collection of other files combined into a single file for easy copying, and then often compressed with a utility such as GZip.

To extract the source code from the tarballs, we'll use the tar command. The commands below assume that you've downloaded the tarballs for libpri, DAHDI, and Asterisk to the /usr/local/src directory on a Linux machine. (You'll probably need to be logged in as the root user to be able to write to that directory.) We're also going to assume that you'll replace the letters X, Y, and Z with the actual version numbers from the tarballs you downloaded. Also please note that the command prompt may be slightly different on your system than what we show here. Don't worry, the commands should work just the same.

First, we'll change to the directory where we downloaded the source code:

```
[root@server ~]# cd /usr/local/src
```

Next, let's extract the source code from each tarball using the tar command. The -zxvf parameters to the tar command tell it what we want to do with the file. The z option tells the system to unzip the file before continuing, the x option tells it to extract the files from the tarball, the v option tells it to be verbose (write out the name of every file as it's being extracted, and the f option tells the tar command that we're extracting the file from a tarball file, and not from a tape.

```
[root@server src]# tar -zxvf libpri-current.tar.gz

[root@server src]# tar -zxvf dahdi-linux-complete-2.X.Y+2.X.Y.tar.gz

[root@server src]# tar -zxvf asterisk-14-current.tar.gz
```

You should now notice that a new sub-directory was created for each of the tarballs, each containing the extracted files from the corresponding tarball. We can now compile and install each of the components.

Building and Installing DAHDI

Overview

Let's install DAHDI!

On Linux, we will use the **DAHDI-linux-complete** tarball, which contains the DAHDI Linux drivers, DAHDI tools, and board firmware files. Again, we're assuming that you've untarred the tarball in the `/usr/local/src` directory, and that you'll replace X and Y with the appropriate version numbers.

See What to Download? for more information on downloading the DAHDI tarballs.

Install DAHDI before libpri
libpri 1.4.13 and later source code depends on DAHDI include files. So, one must install DAHDI before installing libpri.

Don't need DAHDI?
If you are not integrating with any traditional telephony equipment and you are not planning on using the MeetMe dialplan application, then you do not have to install DAHDI or libpri in order to use Asterisk.

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Starting with DAHDI-Linux-complete version 2.8.0+2.8.0, all files necessary to install DAHDI are available in the complete tarball. Therefore, all you need to do to install DAHDI is:

```
[root@server src]# cd dahdi-linux-complete-2.X.Y+2.X.Y

[root@server dahdi-linux-complete-2.X.Y+2.X.Y]# make

[root@server dahdi-linux-complete-2.X.Y+2.X.Y]# make install

[root@server dahdi-linux-complete-2.X.Y+2.X.Y]# make config
```

Building and Installing LibPRI

⚠ **Have you installed DAHDI?**
Before you can build **libpri**, you'll need to Build and Install DAHDI.

As in the other build and install sections, we'll assume that you'll replace the letters X, Y, and Z with the actual version numbers from the tarballs you downloaded.

```
[root@server src]# cd libpri-1.X.Y
```

This command changes directories to the **libpri** source directory.

```
[root@server libpri-1.X.Y]# make
```

This command compiles the **libpri** source code into a system library.

```
[root@server libpri-1.X.Y]# make install
```

This command installs the **libpri** library into the proper system library directory

PJSIP-pjproject

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Overview

Asterisk currently contains two SIP stacks: the original **chan_sip** SIP channel driver which is a complete standalone implementation, has been present in all previous releases of Asterisk *and no longer receives core support*, and the newer **chan_pjsip** SIP stack that is based on Teluu's "pjproject" SIP stack. While the pjproject stack allows us to move a significant amount of code out of Asterisk, it *is* a separate, actively maintained, library that we integrate very tightly to. This presents challenges in making sure that the versions of Asterisk and pjproject currently installed on a system are compatible. For this reason, we've elected to "bundle" a stable, tested version of pjproject with the Asterisk distribution and integrate it into the Asterisk build process. This does not prevent you from using an external pjproject installation but it will not be supported by the Asterisk team. See PJSIP-pjproject below for more info.

Using the Bundled Version of pjproject

Beginning with Asterisk 13.8.0, a stable version of pjproject is included in Asterisk's ./third-party directory and is enabled with the `--with-pjproject-bundled` option to `./configure`. Beginning with Asterisk 15.0.0, it is enabled by default but can be disabled with the `--without-pjproject-bundled` option to `./configure`.

The actual pjproject source code is NOT distributed with Asterisk. Instead the Asterisk build process downloads the official pjproject tarball then patches, configures and builds pjproject when you build Asterisk.

Why use the bundled version?

- **Predictability**: When built with the bundled pjproject, you're always certain of the version you're running against, no matter where it's installed.
- **Scalability**: The default pjproject configuration is optimized for client applications. The bundled version's configuration is optimized for server use.
- **Usability**: Several feature patches, which have been submitted upstream to pjproject but not yet released, are usually included in the bundled version.
- **Safety**: If a security or critical issue is identified in pjproject, it can be patched and made available with a new release of Asterisk instead of having to waiting for a new release of pjproject.
- **Maintainability**: You don't need to build and install separate packages.
- **Supportability**: When asking others for help, there's no question about which version of pjproject you're using and what options it was compiled with.
- **Debugability**: The Asterisk `DONT_OPTIMIZE` and `MALLOC_DEBUG` compile flags, which are essential for troubleshooting crashes and deadlocks, are automatically passed to the pjproject build process.
- **Compatibility**: This is especially important from a development perspective because it means we can be sure that new pjproject APIs that have been introduced or old ones that have been deprecated, are handled and tested appropriately in Asterisk.
- **Reliability**: You can be sure that Asterisk was tested against the bundled version.

Usage

First, run `./contrib/scripts/install_prereq`. Building the bundled pjproject requires the python development libraries which install_prereq installs. All you have to do now is add the `--with-pjproject-bundled` option to your Asterisk `./configure` command line and remove any other `--with-pjproject` option you may have specified.

```
$ cd /path/asterisk-source-dir
# For Asterisk 13 and 14...
$ ./configure --with-pjproject-bundled
# For Asterisk 15+...
$ ./configure
$ make && make install
```

The configure and make processes will download the correct version of pjproject, patch it, configure it, build it, and finally link Asterisk to it statically. No changes in runtime configuration are required. You can leave your system-installed version of pjproject in place if needed. Once compiled with the `--with-pjproject-bundled` option, Asterisk will ignore any other installed versions of pjproject.

Using the bundled version of pjproject doesn't necessarily mean you need internet access to download the pjproject tarball every time you build. There are 2 ways to specify an alternate location from which to retrieve it. First, assuming version 2.6 of pjproject is needed and `/tmp/downloads` is the directory you're going to save to, download the following files to the local directory:

```
$ mkdir /tmp/downloads
$ wget -O /tmp/downloads/pjproject-2.6.tar.bz2
http://www.pjsip.org/release/2.6/pjproject-2.6.tar.bz2
$ wget -O /tmp/downloads/pjproject-2.6.md5
http://www.pjsip.org/release/2.6/MD5SUM.txt
```

It's important that both files be named `pjproject-<version>.tar.bz2` and `pjproject-<version>.md5` respectively.

Now perform either of the following 2 steps:

a. Run ./configure with the `--with-externals-cache=/tmp/downloads` option. ./configure will check there first and only download if the files aren't already there or the tarball checksum doesn't match what's in the md5 file. This is similar to the `--with-sounds-cache` option. BTW, the `--with-externals-cache` mechanism works for the precompiled codecs and the Digium Phone Module for Asterisk as well. As of Asterisk 13.18, 14.7 and 15.0, the `--with-download-cache` option can be used to specify both the externals and sounds cache directory.

b. Set the `PJPROJECT_URL` environment variable to any valid URL (including file:// URLs) where `./configure` can find the tarball and checksum files. The variable can be set in your environment and exported or specified directly on the `./configure` command line. As of Asterisk 13.18, 14.7 and 15.0, the `AST_DOWNLOAD_CACHE` environment variable can be used to specify both the externals and sounds cache directory.

Building and Installing pjproject from Source

> **Installing pjproject from source or from packages is no longer a supported configuration for Asterisk versions that contain the bundled version of pjproject.** Reports of pjproject-related Asterisk issues may only be made against the bundled version. The bundled version inherits flags like DONT_OPTIMIZE and MALLOC_DEBUG from Asterisk which allows us to accurately diagnose issues across both Asterisk and pjproject.

Because earlier releases of pjproject cannot build shared object libraries, some changes were required in order to use it with Asterisk 12. As such, **Asterisk requires pjproject version 2.4 or later (2.6 is current)**. Alternatively, an Asterisk compatible version of pjproject is available on github , or - depending on your Linux distribution - available as a package.

Earlier versions of pjproject downloaded from www.pjsip.org will **not** work with Asterisk 12 or greater.

> If you have previously installed a version of pjproject, you **must** remove that version of pjproject prior to building and installing the Asterisk 12+ compatible version of pjproject. See Uninstalling pjproject for more information.

Downloading pjproject

Obtaining pjproject from Teluu:

Use `wget` to pull the latest version (currently 2.6) from `www.pjsip.org`. Note that the instructions assume that this is 2.6; for the latest version, refer to `www.pjsip.org`:

```
# wget http://www.pjsip.org/release/2.6/pjproject-2.6.tar.bz2

# tar -xjvf pjproject-2.6.tar.bz2
```

Obtaining the latest pjproject from the svn repo:

Use `svn` to install the latest version from www.pjsip.org.

```
# svn co http://svn.pjsip.org/repos/pjproject/trunk/ pjproject-trunk
```

Obtaining (old asterisk) pjproject from the github repo:

If you do not have git, install git on your local machine.

> Downloading and installing `git` is beyond the scope of these instructions, but for Debian/Ubuntu systems, it should be as simple as:
>
> ```
> apt-get install git
> ```
>
> And for RedHat/CentOS systems:
>
> ```
> yum install git
> ```

Checkout the Asterisk 12-compatible pjproject from the Asterisk github repo:

```
# git clone https://github.com/asterisk/pjproject pjproject
```

And that's it!

Building and Installing pjproject

The first step in building and installing pjproject is configuring it using **configure**. For Asterisk, this is arguably the **most** important step in this process. pjproject embeds a number of third party libraries which can conflict with versions of those libraries that may already be installed on your system. Asterisk **will not** use the embedded third party libraries within pjproject. As an example, if you are going to build the res_srtp module in Asterisk, then you **must** specify "--with-external-srtp" when configuring pjproject to point to an external srtp library.

Additionally, Asterisk **REQUIRES** two or three options to be passed to **configure**:

- `--enable-shared` - Instruct pjproject to build shared object libraries. Asterisk will only use shared objects from pjproject.
- `--prefix` - Specify root install directory for pjproject. This will be dependent on your distribution of Linux; typically this is `/usr` for most systems. The default is `/usr/local`
- `--libdir` - Specify the installation location for object code libraries. This may need to be set to `/usr/lib64` for some 64-bit systems such as CentOS.

> Failure to build Asterisk with shared pjproject object libraries **WILL** result in seemingly random crashes. For Asterisk to work properly with pjproject, pjproject **MUST** be built with shared object libraries.

Compiler DEFINEs

- Users who expect to deal with Contact URIs longer than 256 characters or hostnames longer than 128 characters should set `PJSIP_MAX_URL_SIZE` and `PJ_MAX_HOSTNAME` as appropriate.
- IPv6 support in pjproject is, by default, disabled. To enable it, set `PJ_HAS_IPV6` to `1`.
- The default configuration of pjproject enables "assert" functions which can cause Asterisk to crash unexpectedly. To disable the asserts, set `NDEBUG` to `1`.
- The default number of TCP/TLS incoming connections allowed is 64. If you plan on having more than that you'll need to set `PJ_IOQUEUE_MAX_HANDLES` to the new limit.

With the exception of `PJ_IOQUEUE_MAX_HANDLES`, the options can be set in `CFLAGS` and passed to configure as follows: '`./configure CFLAGS="-DNDEBUG=1 -DPJ_HAS_IPV6=1"`', etc. A better way is to create or edit the `pjlib/include/pj/config_site.h` file and set them all there. You should use the bundled version of the `config_site.h` file in `third-party/pjproject/patches` as a starting point. Below is a copy of the file at the time of this writing.

pjlib/include/pj/config_site.h

```
/*
 * Asterisk config_site.h
 */

#include <sys/select.h>

/*
 * Since both pjproject and asterisk source files will include config_site.h,
 * we need to make sure that only pjproject source files include asterisk_malloc_debug.h.
 */
#if defined(MALLOC_DEBUG) && !defined(_ASTERISK_ASTMM_H)
#include "asterisk_malloc_debug.h"
#endif

/*
 * Defining PJMEDIA_HAS_SRTP to 0 does NOT disable Asterisk's ability to use srtp.
 * It only disables the pjmedia srtp transport which Asterisk doesn't use.
 * The reason for the disable is that while Asterisk works fine with older libsrtp
 * versions, newer versions of pjproject won't compile with them.
 */
#define PJMEDIA_HAS_SRTP 0

#define PJ_HAS_IPV6 1
#define NDEBUG 1
#define PJ_MAX_HOSTNAME (256)
#define PJSIP_MAX_URL_SIZE (512)
#ifdef PJ_HAS_LINUX_EPOLL
#define PJ_IOQUEUE_MAX_HANDLES    (5000)
#else
#define PJ_IOQUEUE_MAX_HANDLES    (FD_SETSIZE)
#endif
#define PJ_IOQUEUE_HAS_SAFE_UNREG 1
#define PJ_IOQUEUE_MAX_EVENTS_IN_SINGLE_POLL (16)

#define PJ_SCANNER_USE_BITWISE    0
#define PJ_OS_HAS_CHECK_STACK    0

#ifndef PJ_LOG_MAX_LEVEL
#define PJ_LOG_MAX_LEVEL        6
#endif

#define PJ_ENABLE_EXTRA_CHECK    1
#define PJSIP_MAX_TSX_COUNT        ((64*1024)-1)
#define PJSIP_MAX_DIALOG_COUNT    ((64*1024)-1)
#define PJSIP_UDP_SO_SNDBUF_SIZE    (512*1024)
#define PJSIP_UDP_SO_RCVBUF_SIZE    (512*1024)
#define PJ_DEBUG            0
#define PJSIP_SAFE_MODULE        0
#define PJ_HAS_STRICMP_ALNUM        0
```

```
/*
 * Do not ever enable PJ_HASH_USE_OWN_TOLOWER because the
algorithm is
 * inconsistently used when calculating the hash value and
doesn't
 * convert the same characters as pj_tolower()/tolower().
Thus you
 * can get different hash values if the string hashed has
certain
 * characters in it.  (ASCII '@', '[', '\\', ']', '^', and
'_')
 */
#undef PJ_HASH_USE_OWN_TOLOWER

/*
  It is imperative that PJSIP_UNESCAPE_IN_PLACE remain 0 or
undefined.
  Enabling it will result in SEGFAULTS when URIs containing
escape sequences are encountered.
*/
#undef PJSIP_UNESCAPE_IN_PLACE
#define PJSIP_MAX_PKT_LEN            32000

#undef PJ_TODO
#define PJ_TODO(x)

/* Defaults too low for WebRTC */
#define PJ_ICE_MAX_CAND 32
#define PJ_ICE_MAX_CHECKS (PJ_ICE_MAX_CAND *
PJ_ICE_MAX_CAND)

/* Increase limits to allow more formats */
#define    PJMEDIA_MAX_SDP_FMT   64
#define    PJMEDIA_MAX_SDP_BANDW   4
#define    PJMEDIA_MAX_SDP_ATTR   (PJMEDIA_MAX_SDP_FMT*2 +
4)
#define    PJMEDIA_MAX_SDP_MEDIA   16

/*
 * Turn off the periodic sending of CRLNCRLN.  Default is
on (90 seconds),
 * which conflicts with the global section's
keep_alive_interval option in
 * pjsip.conf.
 */
#define PJSIP_TCP_KEEP_ALIVE_INTERVAL    0
#define PJSIP_TLS_KEEP_ALIVE_INTERVAL    0
```

Other common **configure** options needed for pjproject are listed below:

Library	Configure option	Notes
libspeex shared objects	`--with-external-speex`	Make sure that the library development headers are accessible from pjproject. The CFLAGS and LDFLAGS environment variables may be used to set the include/lib paths.
libsrtp shared objects	`--with-external-srtp`	Make sure that the library development headers are accessible from pjproject. The CFLAGS and LDFLAGS environment variables may be used to set the include/lib paths.
GSM codec	`--with-external-gsm`	Make sure that the library development headers are accessible from pjproject. The CFLAGS and LDFLAGS environment variables may be used to set the include/lib paths.
Disable sound	`--disable-sound`	Let Asterisk perform sound manipulations.
Disable resampling	`--disable-resample`	Let Asterisk perform resample operations.
Disable video	`--disable-video`	Disable video support in pjproject's media libraries. This is not used by Asterisk.
Disable AMR	--disable-opencore-amr	Disable AMR codec support. This is not used by Asterisk

These are some of the more common options used to disable third party libraries in pjproject. However, other options may be needed depending on your system - see **`configure --help`** for a full list of configure options you can pass to pjproject.

a. Now that you understand the pjproject configure options available, change directories to the pjproject source directory:

```
# cd pjproject
```

b. In the pjproject source directory, run the configure script with the options needed for your system:

```
# ./configure --prefix=/usr --enable-shared --disable-sound
--disable-resample --disable-video --disable-opencore-amr
CFLAGS='-O2 -DNDEBUG'
```

A few recommended options are shown. That includes setting a couple important CFLAGS, -O2 for common optimizations and -DNDEBUG to disable debugging code and assertions.

c. Build pjproject:

```
# make dep
# make
```

d. Install pjproject

```
# make install
```

e. Update shared library links.

```
# ldconfig
```

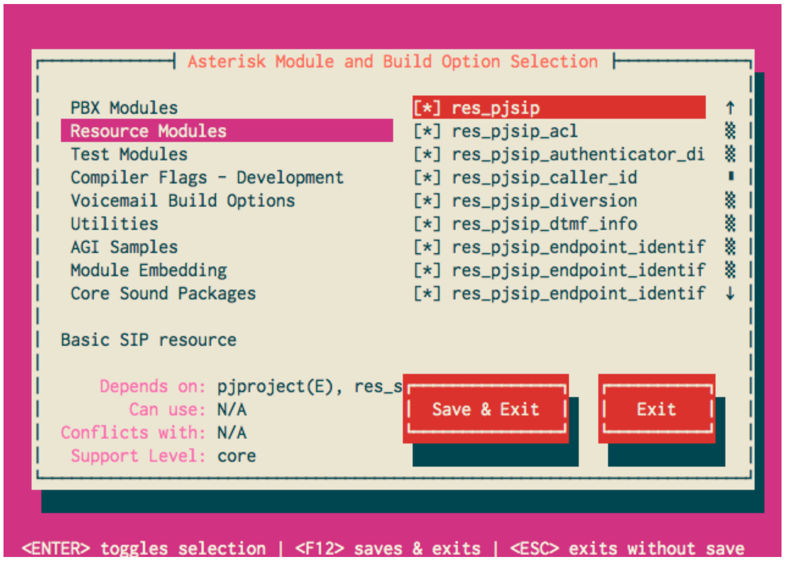
f. Verify that pjproject has been installed in the target location by looking for, and finding the various pjproject modules:

```
# ldconfig -p | grep pj
 libpjsua.so (libc6,x86-64) => /usr/lib/libpjsua.so
 libpjsip.so (libc6,x86-64) => /usr/lib/libpjsip.so
 libpjsip-ua.so (libc6,x86-64) => /usr/lib/libpjsip-ua.so
 libpjsip-simple.so (libc6,x86-64) => /usr/lib/libpjsip-simple.so
 libpjnath.so (libc6,x86-64) => /usr/lib/libpjnath.so
 libpjmedia.so (libc6,x86-64) => /usr/lib/libpjmedia.so
 libpjmedia-videodev.so (libc6,x86-64) =>
/usr/lib/libpjmedia-videodev.so
 libpjmedia-codec.so (libc6,x86-64) => /usr/lib/libpjmedia-codec.so
 libpjmedia-audiodev.so (libc6,x86-64) =>
/usr/lib/libpjmedia-audiodev.so
 libpjlib-util.so (libc6,x86-64) => /usr/lib/libpjlib-util.so
 libpj.so (libc6,x86-64) => /usr/lib/libpj.so
```

g. Finally, verify that Asterisk detects the pjproject libraries. In your Asterisk source directory:

```
# ./configure
# make menuselect
```

h. Browse to the **Resource Modules** category and verify that the `res_pjsip` modules are enabled:



```
Asterisk Module and Build Option Selection

  PBX Modules                          [*] res_pjsip
  Resource Modules                     [*] res_pjsip_acl
  Test Modules                         [*] res_pjsip_authenticator_di
  Compiler Flags - Development         [*] res_pjsip_caller_id
  Voicemail Build Options              [*] res_pjsip_diversion
  Utilities                            [*] res_pjsip_dtmf_info
  AGI Samples                          [*] res_pjsip_endpoint_identif
  Module Embedding                     [*] res_pjsip_endpoint_identif
  Core Sound Packages                  [*] res_pjsip_endpoint_identif

 Basic SIP resource

     Depends on: pjproject(E), res_s
        Can use: N/A
 Conflicts with: N/A
  Support Level: core
```

Save & Exit
Exit
<ENTER> toggles selection | <F12> saves & exits | <ESC> exits without save

i. You're all done! Now, build and install Asterisk as your normally would.

If you need pjsua (for the testsuite, for example), then you may also need to take a look at Installing the Asterisk Test Suite#pjsua_installationPJSUAInstallation to set that up externally as well.

Troubleshooting

First, if you're using Asterisk 13.8.0 or greater, consider switching to the Bundled Version of pjproject

Asterisk fails to detect pjproject libraries

After building and installing pjproject, Asterisk fails to detect any of the libraries - the various res_pjsip components cannot be selected in Asterisk's menuselect

Solution

Verify that Asterisk's config.log shows the following:

```
configure:23029: checking for PJPROJECT
configure:23036: $PKG_CONFIG --exists --print-errors "libpjproject"
Package libpjproject was not found in the pkg-config search path.
Perhaps you should add the directory containing `libpjproject.pc'
to the PKG_CONFIG_PATH environment variable
No package 'libpjproject' found
```

a. Make sure you have `pkg-config` installed on your system.
b. pjproject will install the package config file in `/usr/lib/pkgconfig` . Some distributions, notably Fedora, will instead look for the library in `/usr/lib64` . Update your `PKG_CONFIG_PATH` environment variable with `/usr/lib/pkgconfig` and re-run Asterisk's `configure` script.

pjproject fails to build: errors related to opencore_amr

When building pjproject, errors about opencore_amr are displayed, e.g.:

```
output/pjmedia-codec-x86_64-unknown-linux-gnu/opencore_amr.o:(.rodata+0x60):
multiple definition of `pjmedia_codec_amrnb_framelenbits'
output/pjmedia-codec-x86_64-unknown-linux-gnu/opencore_amr.o:(.rodata+0x60): first
defined here
output/pjmedia-codec-x86_64-unknown-linux-gnu/opencore_amr.o:(.rodata+0x80):
multiple definition of `pjmedia_codec_amrnb_framelen'
output/pjmedia-codec-x86_64-unknown-linux-gnu/opencore_amr.o:(.rodata+0x80): first
defined here
output/pjmedia-codec-x86_64-unknown-linux-gnu/opencore_amr.o:(.rodata+0x20):
multiple definition of `pjmedia_codec_amrwb_framelenbits'
output/pjmedia-codec-x86_64-unknown-linux-gnu/opencore_amr.o:(.rodata+0x20): first
defined here
output/pjmedia-codec-x86_64-unknown-linux-gnu/opencore_amr.o:(.rodata+0x40):
multiple definition of `pjmedia_codec_amrwb_framelen'
output/pjmedia-codec-x86_64-unknown-linux-gnu/opencore_amr.o:(.rodata+0x40): first
defined here
...
```

Solution

You already have the AMR codec installed. Run `configure` with the `--disable-opencore-amr` option specified.

pjproject fails to build: video linker errors

When building pjproject, linker errors referring to various video methods are displayed, e.g.:

```
/home/mjordan/projects/pjproject/pjmedia/lib/libpjmedia-videodev.so: undefined
reference to `pjmedia_format_init_video'
/home/mjordan/projects/pjproject/pjmedia/lib/libpjmedia.so: undefined reference to
`pjmedia_video_format_mgr_instance'
/home/mjordan/projects/pjproject/pjmedia/lib/libpjmedia-videodev.so: undefined
reference to `pjmedia_format_get_video_format_detail'
/home/mjordan/projects/pjproject/pjmedia/lib/libpjmedia-videodev.so: undefined
reference to `pjmedia_get_video_format_info'
```

Solution

Run `configure` with either or both `--disable-video` or `--disable-v4l2`

ldconfig fails to display pjproject libraries

After building pjproject, the dump provided by `ldconfig -p` doesn't display any libraries.

Solution

Run `ldconfig` to re-configure dynamic linker run-time bindings. This will need to be run with super user permissions.

pjproject fails to build on Raspberry Pi

pjproject/Asterisk fails to compile on your Raspberry Pi (raspbian) due to pjproject configure scripts not detecting endianness:

```
/usr/include/pj/config.h:243:6: error: #error Endianness must be declared for this
processor
In file included from /usr/include/pj/types.h:33:0,
                 from /usr/include/pjsip/sip_config.h:27,
                 from /usr/include/pjsip/sip_types.h:34,
                 from /usr/include/pjsip.h:24,
                 from conftest.c:290:
/usr/include/pj/config.h:1161:4: error: #error "PJ_IS_LITTLE_ENDIAN is not
defined!"
/usr/include/pj/config.h:1165:4: error: #error "PJ_IS_BIG_ENDIAN is not defined!"
```

Solution

a. Edit `/usr/include/pj/config.h` (using the editor of your choice)
b. Replace this code:

```
/*
     * ARM, bi-endian, so raise error if endianness is not
configured
     */
#   undef PJ_M_ARMV4
#   define PJ_M_ARMV4           1
#   define PJ_M_NAME            "armv4"
#   define PJ_HAS_PENTIUM       0
#   if !PJ_IS_LITTLE_ENDIAN && !PJ_IS_BIG_ENDIAN
#       error Endianness must be declared for this
processor
#   endif
```

With this:

```
/*
     * ARM, bi-endian, so raise error if endianness is not
configured
     */
#   undef PJ_M_ARMV4
#   define PJ_M_ARMV4           1
#   define PJ_M_NAME            "armv4"
#   define PJ_HAS_PENTIUM       0
#   define PJ_IS_LITTLE_ENDIAN  1
#   define PJ_IS_BIG_ENDIAN     0
```

Then recompile. This workaround was taken from issue ASTERISK-23315.

Uninstalling a Previous Version of pjproject

Typically, other versions of pjproject will be installed as static libraries. These libraries are not compatible with Asterisk and can confuse the build process for Asterisk 12. As such, any static libraries must be removed prior to installing the compatible version of pjproject.

pjproject provides an `uninstall` make target that will remove previous installations. It can be called from the pjproject source directory like:

```
# make uninstall
```

If you don't have an "uninstall" make target, you may need to fetch and merge the latest pjproject from https://github.com/asterisk/pjproject

Alternatively, the following should also remove all previously installed static libraries:

```
# rm -f /usr/lib/libpj*.a /usr/lib/libmilenage*.a
/usr/lib/pkgconfig/libpjproject.pc
```

Finally, you will need to update shared library links:

```
# ldconfig
```

If you want to run a sanity check, you can verify that pjproject has been uninstalled by ensuring no pjproject modules remain on the system:

```
# ldconfig -p | grep pj
```

If running the above command yields no results, that's it! You have successfully uninstalled pjproject from your system. If there are results, you may need to remove other pjproject-related items from /usr/lib as well.

Checking Asterisk Requirements

Configuring Asterisk

Now it's time to compile and install Asterisk. Let's change to the directory which contains the Asterisk source code.

```
[root@server]# cd /usr/local/src/asterisk-14.X.Y
```

Next, we'll run a command called **./configure**, which will perform a number of checks on the operating system, and get the Asterisk code ready to compile on this particular server.

```
[root@server asterisk-14.X.Y]# ./configure
```

This will run for a couple of minutes, and warn you of any missing system libraries or other dependencies. Unless you've installed all of the System Requirements for your version of Asterisk, the **configure** script is likely to fail. If that happens, resolve the missing dependency manually, or use the install_prereq script to resolve all of the dependencies on your system.

Once a dependency is resolved, run **configure** again to make sure the missing dependency is fixed.

> If you have many missing dependencies, you may find yourself running **configure** a lot. If that is the case, you'll do yourself a favour by checking the System Requirements or installing all dependencies via the `install_prereq` script.

On this Page

- Configuring Asterisk
- Using install_prereq

Upon successful completion of **./configure**, you should see a message that looks similar to the one shown below. (Obviously, your host CPU type may be different than the below.)

```
                .$$$$$$$$$$$$$$$=..
              .$7$7..        .7$$7:.
            .$7$7..           .7$$7:.
          .$$:.                 ,$7.7
        .$7.     7$$$$           .$$77
     ..$$.       $$$$$            .$$$7
    ..7$   .?.   $$$$$   .?.       7$$$.
   $.$.   .$$$7. $$$$7 .7$$$.      .$$$.
 .777.   .$$$$$$77$$$77$$$$$7.      $$$,
 $$$~      .7$$$$$$$$$$$$$7.       .$$$.
.$$7          .7$$$$$$$7:          ?$$$.
$$$          ?7$$$$$$$$$$I        .$$$7
$$$       .7$$$$$$$$$$$$$$$$      :$$$.
$$$       $$$$$$7$$$$$$$$$$$$    .$$$.
$$$        $$$   7$$$7  .$$$    .$$$.
$$$$             $$$$7         .$$$.
7$$$7            7$$$$        7$$$
 $$$$$                        $$$
  $$$$7.                       $$  (TM)
   $$$$$$$.           .7$$$$$$  $$
     $$$$$$$$$$$$7$$$$$$$$$.$$$$$$
       $$$$$$$$$$$$$$$$.

configure: Package configured for:&nbsp;
configure: OS type &nbsp;: linux-gnu
configure: Host CPU : x86_64
configure: build-cpu:vendor:os: x86_64 : unknown : linux-gnu :
configure: host-cpu:vendor:os: x86_64 : unknown : linux-gnu :
```

> **Cached Data**
> The **./configure** command caches certain data to speed things up if it's invoked multiple times. To clear all the cached data, you can use the following command to completely clear out any cached data from the Asterisk build system.
>
> ```
> [root@server asterisk-14.X.Y]# make distclean
> ```
>
> You can then re-run **./configure**.

Using install_prereq

The **install_prereq** script is included with every release of Asterisk in the `contrib/scripts` subdirectory. The script has the following options:

- **test** - print only the libraries to be installed.

- **install** - install package dependencies only. Depending on your distribution of Linux, version of Asterisk, and capabilities you wish to use, this may be sufficient.
- **install-unpacakged** - install dependencies that don't have packages but only have tarballs. You may need these dependencies for certain capabilities in Asterisk.

> You should always use your operating system's package management tools to ensure that your system is running the latest software **before** running `install_prereq`. Ubuntu 14's libsnmp-dev package, for instance, has an issue where it will attempt to remove critical system packages if the system isn't updated before an attempt is made to install that package.

```
[root@server asterisk-14.X.Y]# cd contrib/scripts

[root@server asterisk-14.X.Y/contrib/scripts]# ./install_prereq install

[root@server asterisk-14.X.Y/contrib/scripts]# ./install_prereq install-unpackaged
```

Using Menuselect to Select Asterisk Options

Using Menuselect

The next step in the build process is to tell Asterisk which modules to compile and install, as well as set various compiler options. These settings are all controlled via a menu-driven system called **Menuselect**. To access the Menuselect system, type:

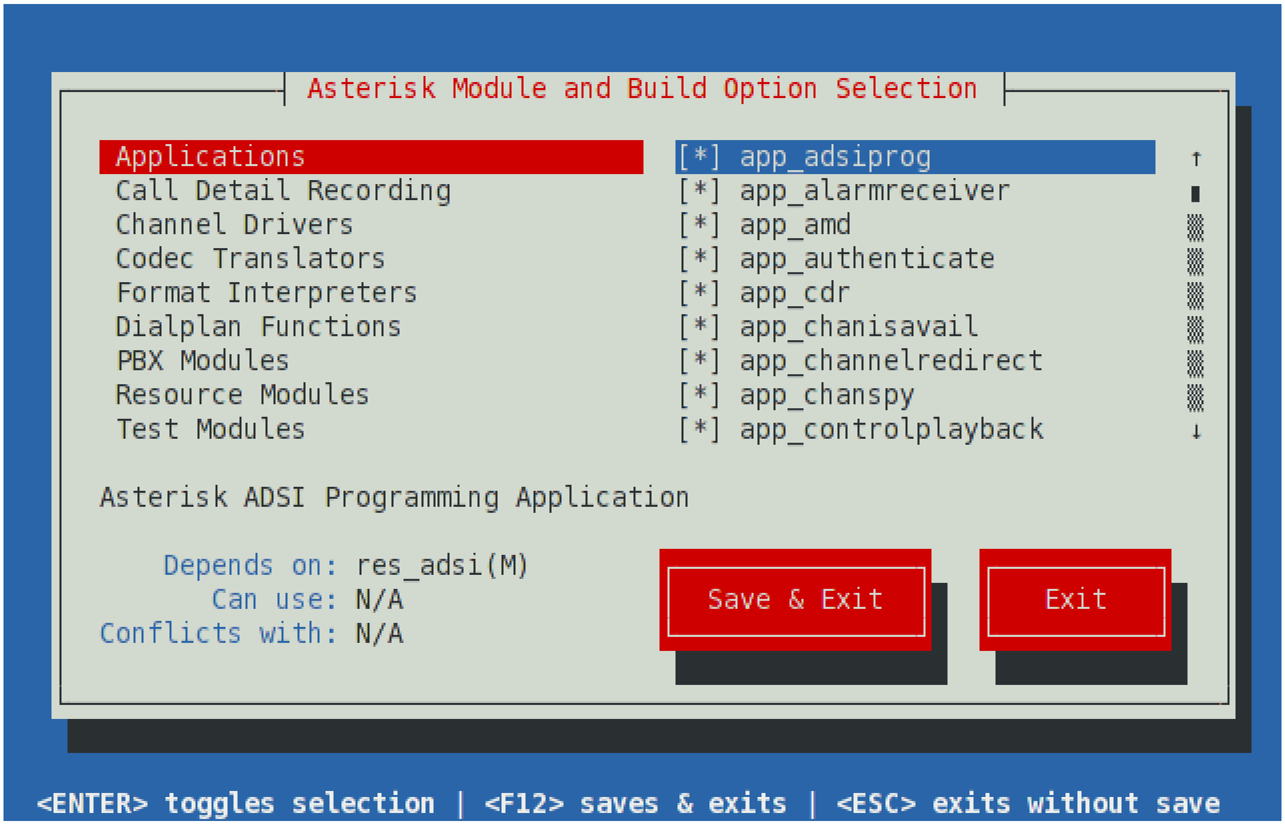
```
[root@server asterisk-14.X.Y]# make menuselect
```

Terminal Window
Your terminal window size must be at least eighty characters wide and twenty-seven lines high, or Menuselect will not work. Instead, you'll get an error message stating

```
Terminal must be at least 80 x 27.
```

The **Menuselect** menu should look like the screen-shot below. On the left-hand side, you have a list of categories, such as **Applications**, **Channel Drivers** , and **PBX Modules**. On the right-hand side, you'll see a list of modules that correspond with the select category. At the bottom of the screen you'll see two buttons. You can use the **Tab** key to cycle between the various sections, and press the **Enter** key to select or unselect a particular module. If you see **[*]** next to a module name, it signifies that the module has been selected. If you see ***XXX** next to a module name, it signifies that the select module cannot be built, as one of its dependencies is missing. In that case, you can look at the bottom of the screen for the line labeled **Depends upon**: for a description of the missing dependency.

On this Page



```
            Asterisk Module and Build Option Selection

  Applications                      [*] app_adsiprog             ↑
  Call Detail Recording             [*] app_alarmreceiver
  Channel Drivers                   [*] app_amd
  Codec Translators                 [*] app_authenticate
  Format Interpreters               [*] app_cdr
  Dialplan Functions                [*] app_chanisavail
  PBX Modules                       [*] app_channelredirect
  Resource Modules                  [*] app_chanspy
  Test Modules                      [*] app_controlplayback      ↓

  Asterisk ADSI Programming Application

     Depends on: res_adsi(M)
        Can use: N/A              Save & Exit        Exit
  Conflicts with: N/A

<ENTER> toggles selection | <F12> saves & exits | <ESC> exits without save
```

When you're first learning your way around Asterisk on a test system, you'll probably want to stick with the default settings in **Menuselect**. If you're building a production system, however, you may not wish to build all of the various modules, and instead only build the modules that your system is using.

When you are finished selecting the modules and options you'd like in **Menuselect**, press **F12** to save and exit, or highlight the **Save and Exit** button and press enter.

Module Support Levels

Menuselect will also show the support level for a selected module or build option. The support level will always be one of `core`, `extended`, or `deprecated`. For more information on these support levels, see Asterisk Module Support States.

Menuselect Categories

Category	Description
Add-ons	Modules that link with libraries that have licensing restrictions beyond what is allowed via the GPLv2 and Asterisk's dual licensing model. See README-addons.txt, delivered with Asterisk, for more information.
Applications	Modules that provide call functionality to the system. An application might answer a call, play a sound prompt, hang up a call, and so forth.
Bridging Modules	Modules that provide various bridge mixing technologies and other bridge related functionality.
Call Detail Recording	Modules that provide Call Detail Record (CDR) drivers for various permanent storage backends.
Channel Event Logging	Modules that provide Channel Event Logging (CEL) drivers for various permanent storage backends.
Channel Drivers	Modules that provide communications with devices outside of Asterisk, and translate that particular signalling or protocol to the core.
Codec Translators	Modules that provide encoding/decoding for audio or video. Typically codecs are used to encode media so that it takes less bandwidth.
Format Interpreters	Modules used to save media to disk in a particular file format, and to convert those files back to media streams on the network.
Dialplan Functions	Modules that are used to retrieve or set various settings on a call. A function might be used to set the Caller ID on an outbound call, for example.
PBX Modules	Modules that implement dialplan functionality or enhancements.
Resource Modules	Modules that provide additional resources to Asterisk. This can includes music on hold, calendar integration, database integration, various protocol stacks, etc.
Test Modules	Unit test modules. These are typically only available when Asterisk has: • Been configured with the `--enable-dev-mode` setting • The `TEST_FRAMEWORK` compilation option has been selected in **Compiler Flags - Development**
Compiler Flags - Development	Various compilation flags that alter Asterisk's behaviour. These flags are often useful in debugging Asterisk, or obtaining information for Asterisk developers. **Easier Debugging of Asterisk Crashes** As much as we may hate to admit it, Asterisk *may* sometimes have problems. If you're finding that Asterisk is crashing on you, there's are settings under **Compiler Flags - Development** that are critical for developers attempting to assist you. For detailed instructions on enabling these settings, see Getting a Backtrace (Asterisk versions < 13.14.0 and 14.3.0).
Voicemail Build Options	Compilation flags that enable different Voicemail (via `app_voicemail`) storage backends.
Utilities	Various utilities for Asterisk. These include Asterisk Database upgrade utilities, Asterisk monitoring utilities, and other potentially useful tools.
AGI Samples	Sample AGI applications.

Module Embedding	Compilation flags to enable embedding of Asterisk dynamic modules into the Asterisk binary.
Core Sound Packages	Core sounds used by Asterisk. Different sound formats can be selected in this menu; when Asterisk is installed, these sounds will be downloaded and installed.
Music On Hold File Packages	Sample Music on Hold media used by Asterisk. Different formats can be selected in this menu; when Asterisk is installed, the various media samples will be downloaded and installed.
Extras Sound Packages	Extra sounds that can be used by Asterisk integrators. Different sound formats can be selected in this menu; when Asterisk is installed, these sounds will be downloaded and installed.

Controlling Menuselect

Options in **Menuselect** can be controlled from the command line. **Menuselect** can be built without invoking the user interface via the `menuselect.makeopts` target:

```
[root@server asterisk-14.X.Y]# make menuselect.makeopts
```

Available options can be viewed using the `--help` command line parameter:

```
[root@server asterisk-14.X.Y]# menuselect/menuselect --help
```

Some of the more common options are shown below.

Menuselect Output
Asterisk expects all **Menuselect** options to be written to the **menuselect.makeopts** file. When enabling/disabling **Menuselect** options via the command line, your output should typically be to that file.

Listing Options

To list all options in **Menuselect**, use the `--list-options` command line parameter:

```
[root@server asterisk-14.X.Y]# menuselect/menuselect --list-options
```

To list only the categories, use the `--category-list` command line parameter:

```
[root@server asterisk-14.X.Y]# menuselect/menuselect --category-list
MENUSELECT_ADDONS
MENUSELECT_APPS
MENUSELECT_BRIDGES
MENUSELECT_CDR
MENUSELECT_CEL
MENUSELECT_CHANNELS
MENUSELECT_CODECS
MENUSELECT_FORMATS
MENUSELECT_FUNCS
MENUSELECT_PBX
MENUSELECT_RES
MENUSELECT_TESTS
MENUSELECT_CFLAGS
MENUSELECT_OPTS_app_voicemail
MENUSELECT_UTILS
MENUSELECT_AGIS
MENUSELECT_EMBED
MENUSELECT_CORE_SOUNDS
MENUSELECT_MOH
MENUSELECT_EXTRA_SOUNDS
```

To list the options in a category, use the `--list-category` command line parameter:

```
[root@server asterisk-14.X.Y]# menuselect/menuselect --list-category MENUSELECT_OPTS_app_voicemail
+ FILE_STORAGE
- ODBC_STORAGE
- IMAP_STORAGE
```

Enabling an Option

To enable an option in Menuselect, use the `--enable` command line parameter:

```
[root@server asterisk-14.X.Y]# menuselect/menuselect --enable IMAP_STORAGE menuselect.makeopts
```

✓ **Chaining Options**

Multiple options can be chained together:

```
[root@server asterisk-14.X.Y]# menuselect/menuselect --enable app_voicemail --enable IMAP_STORAGE menuselect.makeopts
```

Disabling an Option

To disable an option in **Menuselect**, use the `--disable` command line parameter:

```
[root@server asterisk-14.X.Y]# menuselect/menuselect --disable app_voicemail menuselect.makeopts
```

Enabling a Category

An entire category can be enabled in **Menuselect** using the `--enable-category` command line parameter:

```
[root@server asterisk-14.X.Y]# menuselect/menuselect --enable-category MENUSELECT_ADDONS menuselect.makeopts
```

Building and Installing Asterisk

Build and Install Instructions

Now we can compile and install Asterisk. To compile Asterisk, simply type make at the Linux command line.

```
[root@server asterisk-14.X.Y]# make
```

The compiling step will take several minutes, and you'll see the various file names scroll by as they are being compiled. Once Asterisk has finished compiling, you'll see a message that looks like:

```
+--------- Asterisk Build Complete ---------+
+ Asterisk has successfully been built, and +
+ can be installed by running:              +
+                                           +
+                make install               +
+-------------------------------------------+
+--------- Asterisk Build Complete ---------+
```

On this Page

As the message above suggests, our next step is to install the compiled Asterisk program and modules. To do this, use the **make install** command.

```
[root@server asterisk-14.X.Y]# make install
```

When finished, Asterisk will display the following warning:

```
+---- Asterisk Installation Complete -------+
+                                           +
+    YOU MUST READ THE SECURITY DOCUMENT    +
+                                           +
+ Asterisk has successfully been installed. +
+ If you would like to install the sample   +
+ configuration files (overwriting any      +
+ existing config files), run:              +
+                                           +
+                make samples               +
+                                           +
+-------------------------------------------+
+---- Asterisk Installation Complete -------+
```

Security Precautions
As the message above suggests, we very strongly recommend that you read the security documentation before continuing with your Asterisk installation. Failure to read and follow the security documentation can leave your system vulnerable to a number of security issues, including toll fraud.

Advanced Build and Install Options

Customizing the Build/Installation

In some environments, it may be necessary or useful to modify parts of the build or installation process. Some common scenarios are listed here

Passing compilation and linkage flags to gcc

Specific flags can be passed to `gcc` when Asterisk is configured, using the `CFLAGS` and `LDFLAGS` environment variables:

```
[root@server asterisk-14.X.Y]# ./configure CFLAGS=-pg LDFLAGS=-pg
```

Debugging compilation

To see all of the flags passed to gcc, build using the `NOISY_BUILD` setting set to `YES`:

```
[root@server asterisk-14.X.Y]# make NOISY_BUILD=yes
```

Building for non-native architectures

Generally, Asterisk attempts to optimize itself for the machine on which it is built on. On some virtual machines with virtual CPU architectures, the defaults chosen by Asterisk's compilation options will cause Asterisk to build but fail to run. To disable native architecture support, disable the `BUILD_NATIVE` option in menuselect:

```
[root@server asterisk-14.X.Y]# menuselect/menuselect --disable BUILD_NATIVE menuselect.makeopts

[root@server asterisk-14.X.Y]# make
```

Installing to a custom directory

While there are multiple ways to sandbox an instance of Asterisk, the preferred mechanism is to use the `--prefix` option with the `configure` script:

```
[root@server asterisk-14.X.Y]# ./configure --prefix=/usr/local/my_special_folder
```

Note that the default value for `prefix` is `/usr/local`.

Other Make Targets

Target	Description
	Executing `make` with no target is equivalent to the `all` target.
all	Compiles everything everything selected through the `configure` and `menuselect` scripts.
full	This is equivalent to `make` or `make all`, save that it will perform a more thorough investigation of the source code for documentation. This is needed to generate AMI event documentation. Note that your system must have Python in order for this make target to succeed. **Version Notice** This build target is only available in Asterisk 11 and later versions.
install	Installs Asterisk, building Asterisk if it has not already been built. In general, this should be executed after Asterisk has successfully compiled.
uninstall	Removes Asterisk binaries, sounds, man pages, headers, modules and firmware builds from the system.
uninstall-all	Same as the `uninstall` target, but additionally removes configuration, spool directories and logs. All traces of Asterisk. As just noted, this will remove all Asterisk configuration from your system. Do not execute `uninstall-all` unless you are sure that is what you want to do.
clean	Remove all files generated by make.
dist-clean	Remove pretty much all files generated by make and configure.
samples	Install all sample configuration files (.conf files) to `/etc/asterisk/`. Overwrites existing config files.
config	Install init scripts (startup scripts) on your system.
progdocs	Uses doxygen to locally generate HTML development documentation from the source code. Generated in the `doc/` subdirectory of the source; see `doc/index.html`.

Installing Sample Files

Asterisk Sample Configs: not a sample PBX configuration
For many of the sample configuration files that **make samples** installs, the configuration contains more than just an example configuration. The sample configuration files historically were used predominately for documentation of available options. As such, they contain many examples of configuring Asterisk that may not be ideal for standard deployments.

While installing the sample configuration files may be a good starting point for some people, they should not be viewed as recommended configuration for an Asterisk system.

To install a set of sample configuration files for Asterisk, type:

```
[root@server asterisk-14.X.Y]# make samples
```

Any existing sample files which have been modified will be given a **.old** file extension. For example, if you had an existing file named **extensions.conf**, it would be renamed to **extensions.conf.old** and the sample dialplan would be installed as **extensions.conf**.

Installing Initialization Scripts

Now that you have Asterisk compiled and installed, the last step is to install the initialization script, or `initscript`. This script starts Asterisk when your server starts, will monitor the Asterisk process in case anything *bad* happens to it, and can be used to stop or restart Asterisk as well. To install the `initscript`, use the **make config** command.

```
[root@server asterisk-14.X.Y]# make config
```

As your Asterisk system runs, it will generate logfiles. It is recommended to install the `logrotation` script in order to compress and rotate those files, to save disk space and to make searching them or cataloguing them easier. To do this, use the **make install-logrotate** command.

```
[root@server asterisk-14.X.Y]# make install-logrotate
```

Validating Your Installation

Before continuing on, let's check a few things to make sure your system is in good working order. First, let's make sure the DAHDI drivers are loaded. You can use the **lsmod** under Linux to list all of the loaded kernel modules, and the **grep** command to filter the input and only show the modules that have **dahdi** in their name.

```
[root@server asterisk-14.X.Y]# lsmod | grep dahdi
```

If the command returns nothing, then DAHDI has not been started. Start DAHDI by running:

```
[root@server asterisk-14.X.Y]# /etc/init.d/dadhi start
```

Different Methods for Starting Initscripts
Many Linux distributions have different methods for starting initscripts. On most Red Hat based distributions (such as Red Hat Enterprise Linux, Fedora, and CentOS) you can run:

```
[root@server asterisk-14.X.Y]# service dahdi start
```

Distributions based on Debian (such as Ubuntu) have a similar command, though it's not commonly used:

```
[root@server asterisk-14.X.Y]# invoke-rc.d dahdi start
```

If you have DAHDI running, the output of **lsmod | grep dahdi** should look something like the output below. (The exact details may be different, depending on which DAHDI modules have been built, and so forth.)

```
[root@server asterisk-14.X.Y]# lsmod | grep dahdi
dahdi_transcode  7928 1 wctc4xxp
dahdi_voicebus  40464 2 wctdm24xxp,wcte12xp
dahdi          196544 12 wctdm24xxp,wcte11xp,wct1xxp,wcte12xp,wct4xxp
crc_ccitt        2096 1 dahdi
```

Now that DAHDI is running, you can run **dahdi_hardware** to list any DAHDI-compatible devices in your system. You can also run the **dahdi_tool** utility to show the various DAHDI-compatible devices, and their current state.

To check if Asterisk is running, you can use the Asterisk `initscript`.

```
[root@server asterisk-14.X.Y]# /etc/init.d/asterisk status
asterisk is stopped
```

To start Asterisk, we'll use the `initscript` again, this time giving it the start action:

```
[root@server asterisk-14.X.Y]# /etc/init.d/asterisk start
Starting asterisk:
```

When Asterisk starts, it runs as a background service (or daemon), so you typically won't see any response on the command line. We can check the status of Asterisk and see that it's running using the command below. (The process identifier, or pid, will obviously be different on your system.)

```
[root@server asterisk-14.X.Y]# /etc/init.d/asterisk status
asterisk (pid 32117) is running...
```

And there you have it! You've compiled and installed Asterisk, DAHDI, and libpri from source code.

libsrtp

libsrtp is a critical part of providing secure calling with Asterisk but there are some very old versions floating around and even still being made available by major distributions. It's also a library that's used by both Asterisk itself and pjproject. To make matters even worse, pjproject bundles a version with it's tarball.

As of November 2017, the minimum supported version of libsrtp supported by Asterisk is 1.5.4. Earlier versions may allow Asterisk to compile but there were enough issues that earlier versions MAY CRASH, will NOT BE SUPPORTED and are used at your own risk.

Both Asterisk and pjproect do support libsrtp 2.x but we've not tested extensively with it so your safe bet is to stick with 1.5.4 for production builds and use ./configure --with-pjproject-bundled (of course) to make sure both Asterisk and pjproject are in sync with respect to library versions.

Exploring Sound Prompts

Asterisk comes with a wide variety of pre-recorded sound prompts. When you install Asterisk, you can choose to install both core and extra sound packages in several different file formats. Prompts are also available in several languages. To explore the sound files on your system, simply find the sounds directory (this will be **/var/lib/asterisk/sounds** on most systems) and look at the filenames. You'll find useful prompts ("Please enter the extension of the person you are looking for..."), as well as as a number of off-the-wall prompts (such as "Weasels have eaten our phone system", "The office has been overrun with iguanas", and "Try to spend your time on hold not thinking about a blue-eyed polar bear") as well.

✓ **Sound Prompt Formats**

Sound prompts come in a variety of file formats, such as **.wav** and **.ulaw** files. When asked to play a sound prompt from disk, Asterisk plays the sound prompt with the file format that can most easily be converted to the CODEC of the current call. For example, if the inbound call is using the **alaw** CODEC and the sound prompt is available in **.gsm** and **.ulaw** format, Asterisk will play the **.ulaw** file because it requires fewer CPU cycles to transcode to the **alaw** CODEC.

You can type the command **core show translation** at the Asterisk CLI to see the transcoding times for various CODECs. The times reported (in Asterisk 1.6.0 and later releases) are the number of microseconds it takes Asterisk to transcode one second worth of audio. These times are calculated when Asterisk loads the codec modules, and often vary slightly from machine to machine. To perform a current calculation of translation times, you can type the command **core show translation recalc 60**.

Alternate Install Methods

If you already have a Linux system that you can dedicate to Asterisk, simply use the package manager in your operating system to install Asterisk, DAHDI, and libpri. Most modern Linux distributions such as Debian, Ubuntu, and Fedora have these packages in their repositories.

Linux distro maintained packages may be old, so watch out for that. There are no currently maintained official repositories for Asterisk packages.

Asterisk Packages

⚠ There is currently no official repository for asterisk packages. Asterisk source code is distributed by Digium via tarballs and Git. Various community distributions of Asterisk may utilize packages provided and hosted by the distribution maintainer.

Read through Installing Asterisk for more detail on installing Asterisk via source.

Historical Packaging Information

At one time, Asterisk packages were also available for Ubuntu. Currently, packages are not being made by the Asterisk project for this distribution. Information detailing the Ubuntu build environment has been moved onto this page for historical purposes.

Prerequisites

All of Ubuntu's Code, Translations, Packages, Bugs, access control lists, team information, etc. live in Launchpad. So for you to be able to contribute to bug discussions, upload packages, contribute code and translations, it's important that you:

- Create an account on launchpad.
- Create a GPG key and import it.
- Create aSSH key and import it.

Create a Build Environment

Install Ubuntu 10.04 (Lucid)

Installing Ubuntu 10.04 (Lucid)

Enable Backports

```
$ sudo apt-get install python-software-properties
$ sudo add-apt-repository "deb http://ca.archive.ubuntu.com/ubuntu/ $(lsb_release --short
--codename)-backports main universe"
```

Upgrade Lucid to the latest release:

```
$ sudo apt-get update
$ sudo apt-get dist-upgrade
$ sudo apt-get autoremove
$ sudo reboot
```

Install required software

```
$ sudo apt-get install build-essential pbuilder debian-archive-keyring ccache
```

pbuilder

```
$ sudo mkdir -p /var/cache/pbuilder/ccache
$ sudo mkdir -p /var/cache/pbuilder/hook.d
```

```
$ sudo vi /etc/pbuilder/pbuilderrc
```

/etc/pbuilder/pbuilderrc

```
export CCACHE_DIR="/var/cache/pbuilder/ccache"
export PATH="/usr/lib/ccache:${PATH}"
EXTRAPACKAGES="ccache"
BINDMOUNTS="${CCACHE_DIR}"

# Codenames for Debian suites according to their alias. Update these when
# needed.
UNSTABLE_CODENAME="sid"
TESTING_CODENAME="wheezy"
STABLE_CODENAME="squeeze"
```

```
OLDSTABLE_CODENAME="lenny"
STABLE_BACKPORTS_SUITE="$STABLE_CODENAME-backports"

# List of Debian suites.
DEBIAN_SUITES=($UNSTABLE_CODENAME $TESTING_CODENAME $STABLE_CODENAME $OLDSTABLE_CODENAME
    "unstable" "testing" "stable" "oldstable")

# List of Ubuntu suites. Update these when needed.
UBUNTU_SUITES=("oneiric" "natty" "maverick" "lucid")

# Mirrors to use. Update these to your preferred mirror.
DEBIAN_MIRROR="ftp.us.debian.org"
UBUNTU_MIRROR="mirrors.kernel.org"

# Optionally use the changelog of a package to determine the suite to use if
# none set.
if [ -z "${DIST}" ] && [ -r "debian/changelog" ]; then
    DIST=$(dpkg-parsechangelog | awk '/^Distribution: / {print $2}')
    # Use the unstable suite for certain suite values.
    if $(echo "experimental UNRELEASED" | grep -q $DIST); then
        DIST="$UNSTABLE_CODENAME"
    fi
fi

# Optionally set a default distribution if none is used. Note that you can set
# your own default (i.e. ${DIST:="unstable"}).
: ${DIST:="$(lsb_release --short --codename)"}

# Optionally change Debian release states in $DIST to their names.
case "$DIST" in
    unstable)
        DIST="$UNSTABLE_CODENAME"
        ;;
    testing)
        DIST="$TESTING_CODENAME"
        ;;
    stable)
        DIST="$STABLE_CODENAME"
        ;;
    oldstable)
        DIST="$OLDSTABLE_CODENAME"
        ;;
esac

# Optionally set the architecture to the host architecture if none set. Note
# that you can set your own default (i.e. ${ARCH:="i386"}).
: ${ARCH:="$(dpkg --print-architecture)"}

NAME="$DIST"
if [ -n "${ARCH}" ]; then
    NAME="$NAME-$ARCH"
    DEBOOTSTRAPOPTS=("--arch" "$ARCH" "${DEBOOTSTRAPOPTS[@]}")
fi

DEBBUILDOPTS="-b"
if [ "${ARCH}" == "i386" ]; then
    DEBBUILDOPTS="-B"
fi
```

```
BASETGZ="/var/cache/pbuilder/$NAME-base.tgz"
# Optionally, set BASEPATH (and not BASETGZ) if using cowbuilder
# BASEPATH="/var/cache/pbuilder/$NAME/base.cow/"
DISTRIBUTION="$DIST"
BUILDRESULT="/var/cache/pbuilder/$NAME/result/"
APTCACHE="/var/cache/pbuilder/$NAME/aptcache/"
BUILDPLACE="/var/cache/pbuilder/build/"

if $(echo ${DEBIAN_SUITES[@]} | grep -q $DIST); then
    # Debian configuration
    MIRRORSITE="http://$DEBIAN_MIRROR/debian/"
    COMPONENTS="main contrib non-free"
    DEBOOTSTRAPOPTS=("${DEBOOTSTRAPOPTS[@]}"
"--keyring=/usr/share/keyrings/debian-archive-keyring.gpg")
elif $(echo ${UBUNTU_SUITES[@]} | grep -q $DIST); then
    # Ubuntu configuration
    MIRRORSITE="http://$UBUNTU_MIRROR/ubuntu/"
    COMPONENTS="main universe"
    DEBOOTSTRAPOPTS=("${DEBOOTSTRAPOPTS[@]}"
"--keyring=/usr/share/keyrings/ubuntu-archive-keyring.gpg")
else
    echo "Unknown distribution: $DIST"
    exit 1
fi
```

Debian

```
$ for x in unstable testing stable; do for y in i386 amd64; do sudo DIST=${x} ARCH=${y}
pbuilder create; done; done
```

Ubuntu

```
$ for x in lucid maverick natty; do for y in i386 amd64; do sudo DIST=${x} ARCH=${y}
pbuilder create; done; done
```

svn-buildpackage

```
$ vi ~/.svn-buildpackage.conf
```

```
svn-builder=debuild
svn-noautodch
```

quilt

```
$ vi ~/.quiltrc
```

```
QUILT_PATCHES="debian/patches"

QUILT_PATCH_OPTS="--unified-reject-files"
QUILT_REFRESH_ARGS="-p ab --no-timestamps --no-index"
QUILT_DIFF_OPTS="--show-c-function"
QUILT_DIFF_ARGS="-p ab --no-timestamps --no-index --color=auto"
```

devscripts

```
$ vi ~/.devscripts
```

```
DEBCHANGE_RELEASE_HEURISTIC=changelog
DEBCHANGE_MULTIMAINT_MERGE=yes
DEBCHANGE_MAINTTRAILER=no
DEBUILD_ROOTCMD=fakeroot
DEBUILD_LINTIAN=yes
DEBUILD_LINDA=yes
DEFAULT_DEBRELEASE_DEBS_DIR=../build-area/
USCAN_DESTDIR=../tarballs
```

Create a GPG Key

https://help.ubuntu.com/community/GnuPrivacyGuardHowto

```
$ vi ~/.bashrc
```

```
export DEBFULLNAME='Paul Belanger'
export DEBEMAIL='pabelanger@digium.com'
export GPGKEY=8C3B0FA6
export EDITOR=vi
```

See also

- Ubuntu Packaging Guide

Updating an Ubuntu Package

New upstream release

Checkout source

```
$ mkdir -p ~/digium
$ cd ~/digium
$ svn http://blah.org/svn/blah
```

Upstream tarball

```
$ uscan --verbose
```

Update the changelog file

```
$ dch -e
```

Update patches

```
$ while quilt push; do quilt refresh; done
```

Release package

```
$ dch -r
```

Build package source

```
$ svn-buildpackage -S
```

Compile package

```
$ dput ppa:pabelanger/testing ../build-area/*.changes
```

rebuildd

Introduction

Prerequisites

Creating a Build Environment

Getting started

```
sudo apt-get install rebuildd reprepro apache2
```

reprepro

```
$ sudo adduser --system --shell /bin/bash --gecos 'Reprepro Daemon' --group
--disabled-password reprepro
```

```
$ sudo su reprepro
$ cd ~
$ mkdir bin conf incoming
```

```
$ vi ~/conf/distributions
```

distributions

```
Suite: lucid-proposed
Version: 10.04
Codename: lucid-proposed
Architectures: i386 amd64 source
Components: main
SignWith: yes
Log: logfile
  --changes ~/bin/build_sources
```

```
$ vi ~/conf/incoming
```

incoming

```
Name: incoming
IncomingDir: incoming
Allow: lucid-proposed
Cleanup: on_deny on_error
TempDir: tmp
```

```
$ vi ~/conf/apache.conf
```

apache.conf

```
Alias /debian /home/reprepro/
<Directory /home/reprepro>
        Options +Indexes
        AllowOverride None
        order allow,deny
        allow from all
</Directory>
```

```
$ vi ~/bin/build_sources
```

```
#!/bin/bash

action=$1
release=$2
package=$3
version=$4
changes_file=$5

# Only care about packages being added
if [ "$action" != "accepted" ]
then
 exit 0
fi

# Only care about source packages
echo $changes_file | grep -q _source.changes
if [ $? = 1 ]
then
 exit 0
fi

# Kick off the job
echo "$package $version 1 $release"  | sudo rebuildd-job add
```

```
$ reprepro -V -b . createsymlinks
$ reprepro -V -b . processincoming incoming
```

```
$ exit
```

rebuildd

```
$ sudo vi /etc/default/rebuildd
```

```
START_REBUILDD=1
START_REBUILDD_HTTPD=1
DISTS="lucid"
```

Also see

- http://alioth.debian.org/scm/viewvc.php/*checkout*/mirrorer/docs/manual.html?revision=HEAD&root=mirrorer
- http://inodes.org/2009/09/14/building-a-private-ppa-on-ubuntu/

Working with Source Packages

```
$ sudo apt-get build-dep asterisk
```

```
$ DEB_BUILD_OPTIONS="debug" apt-get -b source asterisk
```

Installing Asterisk on Non-Linux Operating Systems

Sub-pages here should provide guidance for installation on Non-Linux operating systems. Contributions are welcome!

Asterisk on (Open)Solaris

Asterisk on Solaris 10 and OpenSolaris

On this page

Digium's Support Status

According to the README file from 1.6.2: "Asterisk has also been 'ported' and reportedly runs properly on other operating systems as well, including Sun Solaris, Apple's Mac OS X, Cygwin, and the BSD variants." Digium's developers have also been doing a good job of addressing build and run-time issues encountered with Asterisk on Solaris.

Build Notes

Prerequisites

The following packages are recommend for building Asterisk 1.6 and later on OpenSolaris:

- SUNWlibm (math library)
- gcc-dev (compiler and several dependencies)
- SUNWflexlex (GNU flex)
- SUNWggrp (GNU grep)
- SUNWgsed (GNU sed)
- SUNWdoxygen (optional; needed for "make progdocs")
- SUNWopenldap (optional; needed for res_config_ldap; see below)
- SUNWgnu-coreutils (optional; provides GNU install; see below)

Caution: installing SUNW gnu packages will change the default application run when the user types 'sed' and 'grep' from /usr/bin/sed to /usr/gnu/bin/sed. Just be aware of this change, as there are differences between the Sun and GNU versions of these utilities.

LDAP dependencies

Because OpenSolaris ships by default with Sun's LDAP libraries, you must install the SUNWopenldap package to provide OpenLDAP libraries. Because of namespace conflicts, the standard LDAP detection will not work.

There are two possible solutions:

1. Port res_config_ldap to use only the RFC-specified API. This should allow it to link against Sun's LDAP libraries.
   - The problem is centered around the use of the OpenLDAP-specific ldap_initialize() call.
2. Change the detection routines in configure to use OpenSolaris' layout of OpenLDAP.
   - This seems doubtful simply because the filesystem layout of SUNWopenldap is so non-standard.

Despite the above two possibilities, there is a workaround to make Asterisk compile with res_config_ldap.

- Modify the "configure" script, changing all instances of "-lldap" to "-lldap-2.4".
  - At the time of this writing there are only 4 instances. This alone will make configure properly detect LDAP availability. But it will not compile.
- When running make, specify the use of the OpenLDAP headers like this:

```
"make LDAP_INCLUDE=-I/usr/include/openldap"
```

Makefile layouts

This has been fixed in Asterisk 1.8 and is no longer an issue.

In Asterisk 1.6 the Makefile overrides any usage of --prefix. I suspect the assumptions are from back before configure provided the ability to set the installation prefix. Regardless, if you are building on OpenSolaris, be aware of this behavior of the Makefile!

If you want to alter the install locations you will need to hand-edit the Makefile. Search for the string "SunOS" to find the following section:

```
# Define standard directories for various platforms
# These apply if they are not redefined in asterisk.conf
ifeq ($(OSARCH),SunOS)
  ASTETCDIR=/etc/asterisk
  ASTLIBDIR=/opt/asterisk/lib
  ASTVARLIBDIR=/var/opt/asterisk
  ASTDBDIR=$(ASTVARLIBDIR)
  ASTKEYDIR=$(ASTVARLIBDIR)
  ASTSPOOLDIR=/var/spool/asterisk
  ASTLOGDIR=/var/log/asterisk
  ASTHEADERDIR=/opt/asterisk/include/asterisk
  ASTBINDIR=/opt/asterisk/bin
  ASTSBINDIR=/opt/asterisk/sbin
  ASTVARRUNDIR=/var/run/asterisk
  ASTMANDIR=/opt/asterisk/man
else
```

Note that, despite the comment, these definitions have build-time and run-time implications. Make sure you make these changes BEFORE you build!

FAX support with SpanDSP

I have been able to get this to work reliably, including T.38 FAX over SIP. If you are running Asterisk 1.6 note Ticket 16342 if you do not install SpanDSP to the default locations (/usr/include and /usr/lib).

There is one build issue with SpanDSP that I need to document (FIXME)

Gotchas

Runtime issues

- WAV and WAV49 files are not written correctly (see Ticket 16610)
- 32-bit binaries on Solaris are limited to 255 file descriptors by default. (see http://developers.sun.com/solaris/articles/stdio_256.html)

Build issues

- bootstrap.sh does not correctly detect OpenSolaris build tools (see Ticket 16341)
- Console documentation is not properly loaded at startup (see Ticket 16688)
- Solaris sed does not properly create AEL parser files (see Ticket 16696; workaround is to install GNU sed with SUNWgsed)
- Asterisk's provided install script, install-sh, is not properly referenced in the makeopts file that is generated during the build. One workaround is to install GNU install from the SUNWgnu-coreutils package. (See Ticket 16781)

Finally, Solaris memory allocation seems far more sensitive than Linux. This has resulted in the discovery of several previously unknown bugs related to uninitialized variables that Linux handled silently. Note that this means, until these bugs are found and fixed, you may get segfaults.

At the time of this writing I have had a server up and running reasonably stable. However, there are large sections of Asterisk's codebase I do not use and likely contain more of these uninitialized variable problems and associated potential segfaults.

Hello World

You've just installed Asterisk and you have read about basic configuration. Now let's quickly get a phone call working so you can get a taste for a simple phone call to Asterisk.

Hello World with Asterisk and SIP

Requirements

This tutorial assumes the following:

- You have a SIP phone plugged into the same LAN where the Asterisk server is plugged in, or can install the Zoiper softphone used in the example
- If you use your own hardware phone, we assume both the phone and Asterisk can reach each other and are on the same subnet.
- When you built Asterisk, you should have made sure to build the SIP channel driver you wanted to use, which may imply other requirements. For example if you want to use chan_pjsip, then make sure you followed the Installing pjproject guide.

Configuration files needed

You should have already run "make samples" if you installed from source, otherwise you may have the sample config files if you installed from packages.

If you have no configuration files in **/etc/asterisk/** then grab the sample config files from the source directory by navigating to it and running "make samples".

Files needed for this example:

- asterisk.conf
- modules.conf
- extensions.conf
- sip.conf or pjsip.conf

You can use the defaults for asterisk.conf and modules.conf, we'll only need to modify extensions.conf and sip.conf or pjsip.conf.

To get started, go ahead and move to the /etc/asterisk/ directory where the files are located.

```
cd /etc/asterisk
```

Configure extensions.conf

Backup the sample extensions.conf and create a new one

```
mv extensions.conf extensions.sample
vim extensions.conf
```

I'm assuming you use the VI/VIM editor here, after all, it is the best.

We are going to use a very simple dialplan. A dialplan is simply instructions telling Asterisk what to do with a call.

Edit your blank extensions.conf to reflect the following:

```
[from-internal]
exten = 100,1,Answer()
same = n,Wait(1)
same = n,Playback(hello-world)
same = n,Hangup()
```

When a phone dials extension 100, we are telling Asterisk to **Answer** the call, **Wait** one second, then Play (**Playback**) a sound file (hello-world) to the channel and **Hangup**.

Configure a SIP channel driver

Depending on the version of Asterisk in use, you may have the option of more than one SIP channel driver. You'll have to pick one to use for the example.

- Asterisk 11 and previous: chan_sip is the primary option.
- Asterisk 12 and beyond: You'll probably want to use chan_pjsip (the newest driver), but you still have the option of using chan_sip as well

Follow the instructions below for the channel driver you chose.

Configure chan_sip

Backup and edit a new blank **sip.conf**, just like you did with extensions.conf.

Then add the following to your sip.conf file:

```
[general]
context=default

[6001]
type=friend
context=from-internal
host=dynamic
secret=unsecurepassword
disallow=all
allow=ulaw
```

Basic configuration will be explained in more detail in other sections of the wiki. For this example to work, just make sure you have everything exactly as written above. For the sake of terminology, it is useful to note that though we have this SIP configuration configured with "type=friend", most people refer to this as configuring a SIP peer.

Configure chan_pjsip

Backup and edit a new blank **pjsip.conf**, just like you did with extensions.conf.

Then add the following to your pjsip.conf file:

```
[transport-udp]
type=transport
protocol=udp
bind=0.0.0.0

[6001]
type=endpoint
context=from-internal
disallow=all
allow=ulaw
auth=6001
aors=6001

[6001]
type=auth
auth_type=userpass
password=unsecurepassword
username=6001

[6001]
type=aor
max_contacts=1
```

Configure your SIP phone

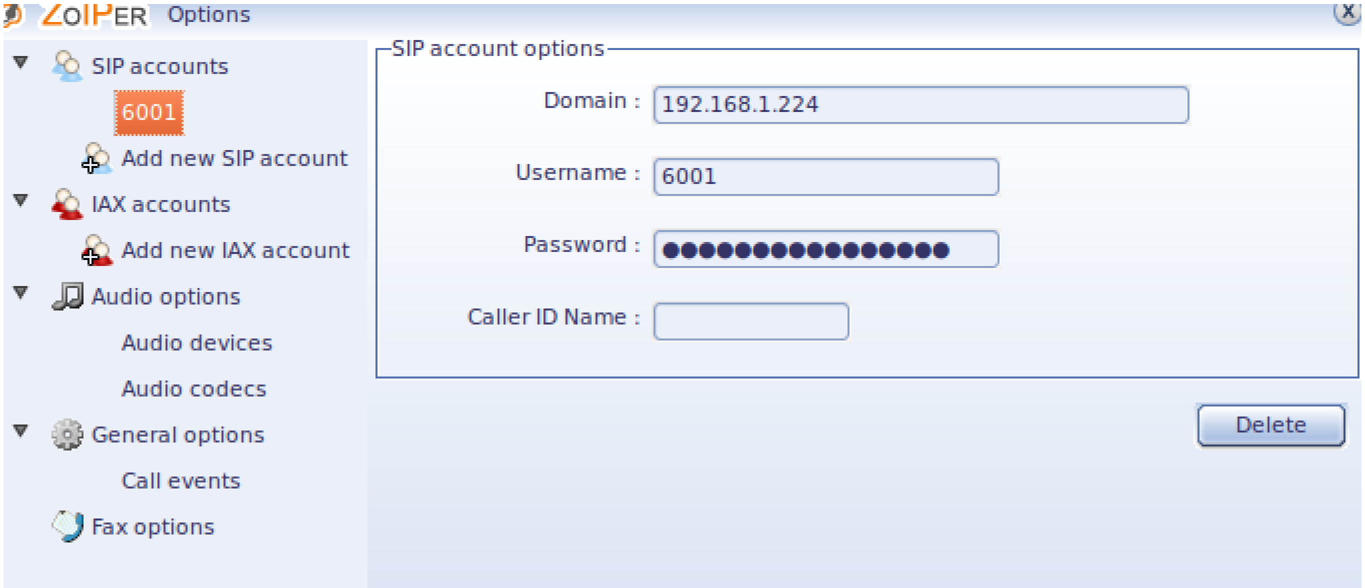
You can use any SIP phone you want of course, but for this demonstration we'll use Zoiper, a Softphone which just happens to be easily demonstrable.

You can find the latest version of Zoiper for your platform at their website. You can install it on the same system you are running Asterisk on, or it may make more sense to you if you install on another system on the same LAN (though you might find complication with software firewalls in that case).

Once you have Zoiper installed. Configure a new SIP account in Zoiper.

1. Once Zoiper is opened, click the wrench icon to get to settings.
2. Click "Add new SIP account"
3. Enter 6001 for the account name, click OK
4. Enter the IP address of your Asterisk system in the **Domain** field
5. Enter 6001 in the **Username** field
6. Enter your SIP peer's password in the **Password** field

7. Enter whatever you like in **Caller ID Name** or leave it blank
8. Click OK



SIP account options

Domain : 192.168.1.224

Username : 6001

Password : ●●●●●●●●●●●●●●●●●

Caller ID Name :

Your results should look like the above screen shot.

Start Asterisk

Back at the Linux shell go ahead and start Asterisk. We'll start Asterisk with a control console (-c) and level 5 verbosity (vvvvv).

```
asterisk -cvvvvv
```

Or if Asterisk is already running, restart Asterisk from the shell and connect to it.

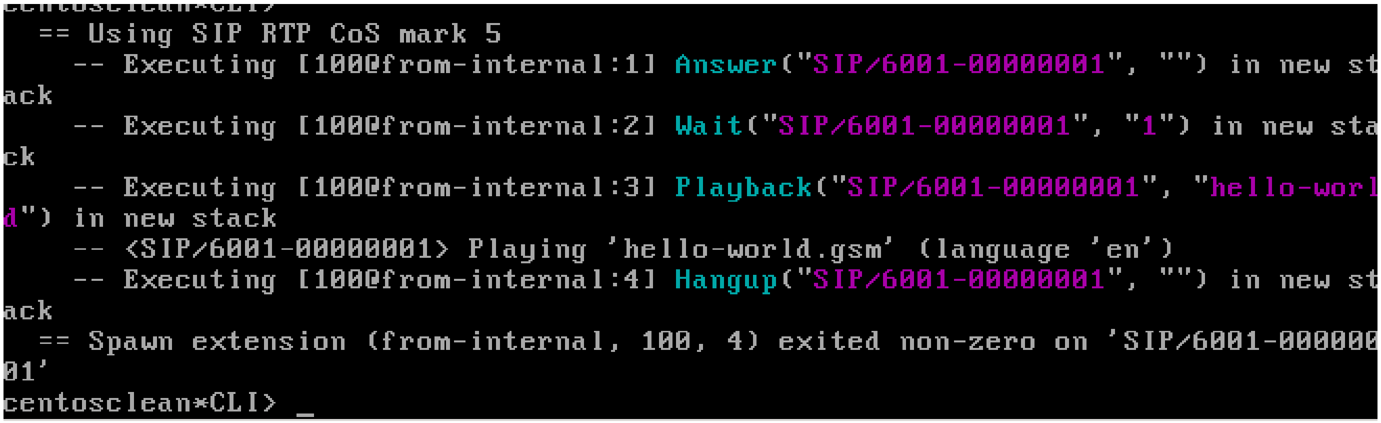
```
asterisk -rx "core restart now"
asterisk -rvvvvv
```

Make the call

Go back to the main Zoiper interface, and make sure the account is registered. Select the account from the drop down list and click the **Register** button next to it. If it says registered, you are good to go. If it doesn't register, then double check your configuration.

Once registered, enter extension 100 and click the Dial button. The call should be made and you should hear the sound file hello-world!

On the Asterisk CLI, you should see something like:



```
centosclean*CLI>
  == Using SIP RTP CoS mark 5
    -- Executing [100@from-internal:1] Answer("SIP/6001-00000001", "") in new st
ack
    -- Executing [100@from-internal:2] Wait("SIP/6001-00000001", "1") in new sta
ck
    -- Executing [100@from-internal:3] Playback("SIP/6001-00000001", "hello-worl
d") in new stack
    -- <SIP/6001-00000001> Playing 'hello-world.gsm' (language 'en')
    -- Executing [100@from-internal:4] Hangup("SIP/6001-00000001", "") in new st
ack
  == Spawn extension (from-internal, 100, 4) exited non-zero on 'SIP/6001-000000
01'
centosclean*CLI> _
```

Now that you have made a very simple call, you may want to start reading through the other sections on the wiki to learn more about Operation, Fundamentals and Configuration.

Fundamentals

Asterisk Architecture

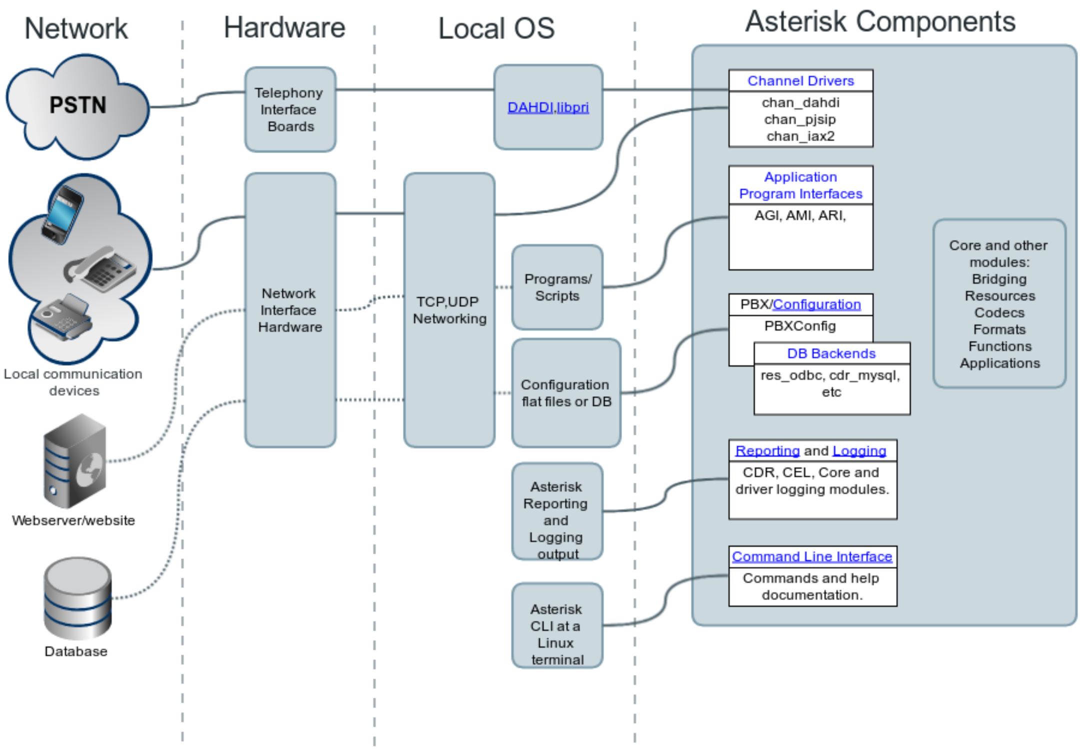
From an architectural standpoint, Asterisk is made up of many different modules. This modularity gives you an almost unlimited amount of flexibility in the design of an Asterisk-based system. As an Asterisk administrator, you have the choice on which modules to load and the configuration of each module. Each module that you load provides different capabilities to the system. For example, one module might allow your Asterisk system to communicate with analog phone lines, while another might add call reporting capabilities. In this section, we'll discuss the overall relationships of some Asterisk component, the various types of modules and their capabilities.

Asterisk Architecture, The Big Picture

Before we dive too far into the various types of modules, let's first take a step back and look at the overall architecture of Asterisk.

Asterisk a big program with many components, with complex relationships. To be able to use it, you don't have to know how everything relates in extreme detail. Below is a simplified diagram intended to illustrate the relationships of some major components to each other and to entities outside Asterisk. It is useful to understand how a component may relate to things outside Asterisk as Asterisk is not typically operating without some connectivity or interaction with other network devices or files on the local system.

An Asterisk System



Remember this is not an exhaustive diagram. It covers only a few of the common relationships between certain components.

On this Page

Asterisk Architecture

Asterisk has a **core** that can interact with many **modules**. Modules called channel drivers provide **channels** that follow Asterisk **dialplan** to execute programmed behavior and facilitate communication between devices or programs outside Asterisk. Channels often use **bridging** infrastructure to interact with other channels. We'll describe some of these concepts in brief below.

The Core

The heart of any Asterisk system is the **core**. The PBX core is the essential component that provides a lot of infrastructure. Among many functions it reads the configuration files, including dialplan and loads all the other **modules**, distinct components that provide more functionality.

The core loads and builds the dialplan, which is the logic of any Asterisk system. The dialplan contains a list of instructions that Asterisk should follow to know how to handle incoming and outgoing **calls** on the system.

Modules

Other than functionality provided by the core of Asterisk, modules provide all other functionality. The source for many modules is distributed with Asterisk, though other modules may be available from community members or even businesses that make commercial modules. The modules distributed with Asterisk can be optionally be built when Asterisk is built.

Modules are not only optionally built, but you can affect at load-time whether they will be loaded at all, the loading order or even unload/load them during run-time. Most modules are independently configurable and have their own configuration files. Some modules have support for configuration to be read statically or dynamically(realtime) from database backends.

From a logistical standpoint, these modules are typically files with a **.so** file extension, which live in the Asterisk modules directory (which is typically **/usr/lib/asterisk/modules**). When Asterisk starts up, it loads these files and adds their functionality to the system.

Asterisk modules which are part of the core have a file name that look like **pbx_xxxxx.so**. All of the modules types are discussed in the section Types of Asterisk Modules.

✓ **A Plethora of Modules**
Take just a minute and go look at the Asterisk modules directory on your system. You should find a wide variety of modules. A default installation of Asterisk has over one hundred fifty different modules!

A Few Module Examples

- **chan_pjsip** uses **res_pjsip** and many other res_pjsip modules to provide a SIP stack for SIP devices to interact with Asterisk and with each other through Asterisk.
- **app_voicemail** provides traditional PBX-type voicemail features.
- **app_confbridge** provides conference bridges with many optional features.
- **res_agi** provides the Asterisk Gateway Interface, an API that allows call control from external scripts and programs.

Calls and Channels

As was mentioned in the Asterisk as a Swiss Army Knife of Telephony section, the primary purpose of Asterisk is being an engine for building Real Time Communication systems and applications.

In most but not all cases this means you'll deal with the concept of "calls". Calls in telephony terminology typically refer to one phone communicating with (calling) another phone over a medium, such as a PSTN line. However in the case of Asterisk a call typically references one or more **channels** existing in Asterisk.

Here are some example "calls".

- A phone calling another phone through Asterisk.
- A phone calling many phones at once (for example, paging) through Asterisk.
- A phone calls an application or the reverse happens. e.g., app_voicemail or app_queue
- A local channel is created and interacts with an application or another channel.

Note that I primarily use phones as an example, however you could refer to any channel or group of channels as a call. It doesn't matter if the devices are phones or something else, like an alarm system sensor or garage door opener.

Channels

Channels are created by Asterisk using Channel Drivers. They can utilize other resources in the Asterisk system to facilitate various types of communication between one or more devices. Channels can be **bridged** to other channels and be affected by **applications** and **functions**. Channels can make use of many other resources provided by other modules or external libraries. For example SIP channels when passing audio will make use of the **codec** and **format** modules. Channels may interact with many different modules at once.

Dialplan

Dialplan is the one main method of directing Asterisk behavior. Dialplan exists as text files (for example extensions.conf) either in the built-in dialplan scripting language, AEL or LUA formats. Alternatively dialplan could be read from a database, along with other module configuration. When writing dialplan, you will make heavy use of **applications** and **functions** to affect channels, configuration and features.

Dialplan can also call out through other interfaces such as AGI to receive call control instruction from external scripts and programs. The Dialplan section of the wiki goes into detail on the usage of dialplan.

Types of Asterisk Modules

There are many different types of modules, each providing their own functionality and capabilities to Asterisk. Configuration of loading is described in Configuring the Asterisk Module Loader.

Use the CLI command **module show** to see all the loaded modules in your Asterisk system. See the command usage for details on how to filter the results shown with a pattern.

Click here for example "module show" output...

```
mypbx*CLI> module show
Module                         Description                             Use Count  Status      Support Level
app_adsiprog.so                Asterisk ADSI Programming Application    0          Running          extended
app_agent_pool.so              Call center agent pool applications      0          Running              core
app_alarmreceiver.so           Alarm Receiver for Asterisk              0          Running          extended
app_amd.so                     Answering Machine Detection Application  0          Running          extended
app_authenticate.so            Authentication Application               0          Running              core
```

Various Module Types

- **Channel Drivers**

Channel drivers communicate with devices outside of Asterisk, and translate that particular signaling or protocol to the core.

- **Dialplan Applications**

Applications provide call functionality to the system. An application might answer a call, play a sound prompt, hang up a call or provide more complex behavior such as queuing, voicemail or conferencing feature sets.

- **Dialplan Functions**

Functions are used to retrieve, set or manipulate various settings on a call. A function might be used to set the Caller ID on an outbound call, for example.

- **Resources**

As the name suggests, resources provide resources to Asterisk and its modules. Common examples of resources include music on hold and call parking.

- **CODECs**

A CODEC (which is an acronym for COder/DECoder) is a module for encoding or decoding audio or video. Typically codecs are used to encode media so that it takes less bandwidth. These are essential to translating audio between the audio codecs and payload types used by different devices.

- **File Format Drivers**

File format drivers are used to save media to disk in a particular file format, and to convert those files back to media streams on the network.

- **Call Detail Record (CDR) Drivers**

CDR drivers write call logs to a disk or to a database.

- **Call Event Log (CEL) Drivers**

Call event logs are similar to call detail records, but record more detail about what happened inside of Asterisk during a particular call.

- **Bridge Drivers**

Bridge drivers are used by the bridging architecture in Asterisk, and provide various methods of bridging call media between participants in a call.

The next sub-sections will include detail on each of the module types.

Channel Driver Modules

All calls from the outside of Asterisk go through a channel driver before reaching the core, and all outbound calls go through a channel driver on their way to the external device.

The PJSIP channel driver (chan_pjsip), for example, communicates with external devices using the SIP protocol. It translates the SIP signaling into the core. This means that the core of Asterisk is signaling agnostic. Therefore, Asterisk isn't just a SIP or VOIP communications engine, it's a multi-protocol engine.

For more information on the various channel drivers, see the configuration section for Channel Drivers.

All channel drivers have a file name that look like **chan_xxxxx.so**, such as **chan_pjsip.so** or **chan_dahdi.so**.

Dialplan Application Modules

The application modules provide call functionality to the system. These applications are then scripted sequentially in the dialplan. For example, a call might come into Asterisk dialplan, which might use one application to answer the call, another to play back a sound prompt from disk, and a third application to allow the caller to leave voice mail in a particular mailbox.

For more information on dialplan applications, see Applications.

All application modules have file names that looks like **app_xxxxx.so**, such as **app_voicemail.so**, however applications and functions can also be provided by the core and other modules. Modules like res_musiconhold and res_xmpp provide applications related to their own functionality.

Dialplan Function Modules

Dialplan functions are somewhat similar to dialplan applications, but instead of doing work on a particular channel or call, they simply retrieve or set a particular setting on a channel, or perform text manipulation. For example, a dialplan function might retrieve the Caller ID information from an incoming call, filter some text, or set a timeout for caller input.

For more information on dialplan functions, see PBX Features.

All dialplan function modules have file names that looks like **func_xxxxx.so**, such as **func_callerid.so**, however applications and functions can also be provided by the core and other modules. Modules like res_musiconhold and res_xmpp provide applications related to their own functionality.

Resource Modules

Resources provide functionality to Asterisk that may be called upon at any time during a call, even while another application is running on the channel. Resources are typically used as asynchronous events such as playing hold music when a call gets placed on hold, or performing call parking.

Resource modules have file names that looks like **res_xxxxx.so**, such as **res_musiconhold.so**.

Resource modules can provide Asterisk 1.8 Dialplan Applications and Asterisk 1.8 Dialplan Functions even if those apps or functions don't have separate modules.

Codec Modules

CODEC modules have file names that look like **codec_xxxxx.so**, such as **codec_alaw.so** and **codec_ulaw.so**.

CODECs represent mathematical algorithms for encoding (compressing) and decoding (decompression) media streams. Asterisk uses CODEC modules to both send and recieve media (audio and video). Asterisk also uses CODEC modules to convert (or transcode) media streams between different formats.

Modules Provided by Default

Asterisk is provided with CODEC modules for the following media types:

- ADPCM, 32kbit/s
- G.711 A-law, 64kbit/s
- G.711 µ-law, 64kbit/s
- G.722, 64kbit/s
- G.726, 32kbit/s
- GSM, 13kbit/s
- LPC-10, 2.4kbit/s

Other Formats and Modules

The Asterisk core provides capability for 16 bit Signed Linear PCM, which is what all of the CODECs are encoding from or decoding to. There is another CODEC module, **codec_resample** which allows re-sampling of Signed Linear into different sampling rates 12,16,24,32,44,48,96 or 192 kHz to aid translation.

Various other CODEC modules will be built and installed if their dependencies are detected during Asterisk compilation.

- If the DAHDI drivers are detected then **codec_dahdi** will installed.
- If the Speex (www.speex.org) development libraries are detected, **codec_speex** will also be installed.
- If the iLBC (www.ilbcfreeware.org) development libraries are detected, **codec_ilbc** will also be installed.

Support for the patent-encumbered G.729A or G.723.1 CODECs is provided by Digium on a commercial basis through software (G.729A) or hardware (G.729A and G.723.1) products. For more information about purchasing licenses or hardware to use the G.729A or G.723.1 CODECs with Asterisk, please see Digium's website.

Support for Polycom's patent-encumbered but free G.722.1 Siren7 and G.722.1C Siren14 CODECs, or for Skype's SILK CODEC, can be enabled in Asterisk by downloading the binary CODEC modules from Digium's website.

✓ On the Asterisk Command Line Interface, use the command "core show translation" to show the translation times between all registered audio formats.

File Format Drivers

Asterisk uses file format modules to take media (such as audio and video) from the network and save them on disk, or retrieve said files from disk and convert them back to a media stream. While often related to CODECs, there may be more than one available on-disk format for a particular CODEC.

File format modules have file names that look like **format_xxxxx.so**, such as **format_wav.so** and **format_jpeg.so**.

Below is a list of format modules included with recent versions of Asterisk:

- format_g719
- format_g723
- format_g726
- format_g729
- format_gsm
- format_h263
- format_h264
- format_ilbc
- format_jpeg
- format_ogg_vorbis
- format_pcm
- format_siren7
- format_siren14
- format_sln
- format_vox
- format_wav_gsm
- format_wav

Call Detail Record (CDR) Drivers

CDR modules are used to store Call Detail Records (CDR) in a variety of formats. Popular storage mechanisms include comma-separated value (CSV) files, as well as relational databases such as MySQL or PostgreSQL. Call detail records typically contain one record per call, and give details such as who made the call, who answered the call, the amount of time spent on the call, and so forth.

Call detail record modules have file names that look like **cdr_xxxxx.so**, such as **cdr_csv.so**. The recommended module to use for connecting to CDR Storage Backends is **cdr_adaptive_odbc.so**.

Call Event Log (CEL) Driver Modules

Call Event Logs record the various actions that happen on a call. As such, they are typically more detailed that call detail records. For example, a call event log might show that Alice called Bob, that Bob's phone rang for twenty seconds, then Bob's mobile phone rang for fifteen seconds, the call then went to Bob's voice mail, where Alice left a twenty-five second voicemail and hung up the call. The system also allows for custom events to be logged as well.

For more information about Call Event Logging, see Call Event Logging.

Call event logging modules have file names that look like **cel_xxxxx.so**, such as **cel_custom.so** and **cel_adaptive_odbc.so**.

Bridging Modules

Beginning in Asterisk 1.6.2, Asterisk introduced a new method for bridging calls together. It relies on various bridging modules to control how the media streams should be mixed for the participants on a call. The new bridging methods are designed to be more flexible and more efficient than earlier methods.

Bridging modules have file names that look like **bridge_xxxxx.so**, such as **bridge_simple.so** and **bridge_multiplexed.so**.

Directory and File Structure

The top level directories used by Asterisk can be configured in the asterisk.conf configuration file.

Here we'll describe what each directory is used for, and what sub-directories Asterisk will place in each by default. Below each heading you can also see the correlating configuration line in asterisk.conf.

Asterisk Configuration Files

```
astetcdir => /etc/asterisk
```

This location is used to store and read Asterisk configuration files. That is generally files with a .conf extension, but other configuration types as well, for example .lua and .ael.

Asterisk Modules

```
astmoddir => /usr/lib/asterisk/modules
```

Loadable modules in Shared Object format (.so) installed by Asterisk or the user should go here.

Various Libraries

```
astvarlibdir => /var/lib/asterisk
```

Additional library elements and files containing data used in runtime are put here.

On This Page

Database Directory

```
astdbdir => /var/lib/asterisk
```

This location is used to store the data file for Asterisk's internal database. In Asterisk versions using the SQLite3 database, the file will be named astdb.sqlite3.

Encryption Keys

```
astkeydir => /var/lib/asterisk
```

When configuring key-based encryption, Asterisk will look in the **keys** subdirectory of this location for the necessary keys.

System Data Directory

```
astdatadir => /var/lib/asterisk
```

By default, Asterisk sounds are stored and read from the **sounds** subdirectory at this location.

AGI(Asterisk Gateway Interface) Directory

```
astagidir => /var/lib/asterisk/agi-bin
```

When using various AGI applications, Asterisk looks here for the AGI scripts by default.

Spool Directories

```
astspooldir => /var/spool/asterisk
```

This directory is used for storing spool files from various core and module-provided components of Asterisk.

Most of them use their own subdirectories, such as the following:

- dictate
- meetme
- monitor
- outgoing
- recording
- system
- tmp
- voicemail

Running Process Directory

```
astrundir => /var/run/asterisk
```

When Asterisk is running, you'll see two files here, **asterisk.ctl** and **asterisk.pid**. That is the control socket and the PID(Process ID) files for Asterisk.

Logging Output

```
astlogdir => /var/log/asterisk
```

When Asterisk is configured to provide log file output, it will be stored in this directory.

System Binary Directory

```
astsbindir => /usr/sbin
```

By default, Asterisk looks in this directory for any system binaries that it uses, if you move the Asterisk binary itself or any others that it uses, you'll need to change this location.

Asterisk Configuration

The top-level page for all things related to Asterisk configuration

Asterisk Configuration Files

This Asterisk Configuration Files section covers the following:

- The Config File Format and syntax.
- How to use Comments in the files.
- Using The include, tryinclude and exec Constructs to include file content into other files or get external program output into a file
- Adding to an existing section settings from other configuration sections
- The syntax and usage of Templates for avoiding redundant configuration.

See also

If you haven't read it already, the Asterisk Architecture section will help you to understand the context within which the configuration files are used. The Directory and File Structure will tell you exactly where to find the configuration files which we generalize in this section. See the Configuration section for information on how to configure files related to **specific** components of Asterisk.

Config File Format

Asterisk is a very flexible telephony engine. With this flexibility, however, comes a bit of complexity. Asterisk has quite a few configuration files which control almost every aspect of how it operates. The format of these configuration files, however, is quite simple. The Asterisk configuration files are plain text files, and can be edited with any text editor.

Sections and Settings

The configuration files are broken into various section, with the section name surrounded by square brackets. Section names should not contain spaces, and are case sensitive. Inside of each section, you can assign values to various settings. Note that settings are also referred to as configuration options or just, options. In general, settings in one section are independent of values in another section. Some settings take values such as true or false, while other settings have more specific settings. The syntax for assigning a value to a setting is to write the setting name, an equals sign, and the value, like this:

```
[section-name]
setting=true

[another_section]
setting=false
setting2=true
```

Additionally here is closer to real-life example from the pjsip.conf.sample file:

```
[transport-udp-nat]
type=transport
protocol=udp
bind=0.0.0.0
local_net=192.0.2.0/24
external_media_address=203.0.113.1
external_signaling_address=203.0.113.1
```

Objects

Some Asterisk configuration files also create objects. The syntax for objects is slightly different than for settings. To create an object, you specify the type of object, an arrow formed by the equals sign and a greater-than sign (=>), and the settings for that object.

```
[section-name]
some_object => settings
```

✓ Confused by Object Syntax?
In order to make life easier for newcomers to the Asterisk configuration files, the developers have made it so that you can also create objects with an equal sign. Thus, the two lines below are functionally equivalent.

```
some_object => settings
some_object=settings
```

It is common to see both versions of the syntax, especially in online Asterisk documentation and examples. This book, however, will denote objects by using the arrow instead of the equals sign.

```
[section-name]
label1=value1
label2=value2
object1 => name1

label1=value0
label3=value3
object2 => name2
```

In this example, **object1** inherits both **label1** and **label2**. It is important to note that **object2** also inherits **label2**, along with **label1** (with the new overridden value **value0**) and **label3**.

In short, objects inherit all the settings defined above them in the current section, and later settings override earlier settings.

Comments

We can (and often do) add comments to the Asterisk configuration files. Comments help make the configuration files easier to read, and can also be used to temporarily disable certain settings.

Comments on a Single Line

Single-line comments begin with the semicolon (;) character. The Asterisk configuration parser treats everything following the semicolon as a comment. To expand on our previous example:

```
[section-name]
setting=true

[another_section]
setting=false ; this is a comment
; this entire line is a comment
;awesome=true
; the semicolon on the line above makes it a
; comment, disabling the setting
```

Block Comments

Asterisk also allows us to create block comments. A block comment is a comment that begins on one line, and continues for several lines. Block comments begin with the character sequence

```
;--
```

and continue across multiple lines until the character sequence

```
--;
```

is encountered. The block comment ends immediately after --; is encountered.

```
[section-name]
setting=true
;-- this is a block comment that begins on this line
and continues across multiple lines, until we
get to here --;
```

Using The include, tryinclude and exec Constructs

include, tryinclude and exec

You might have arrived here looking for Include Statements specific to Asterisk dialplan.

There are two other constructs we can use within all of our configuration files. They are **#include** and **#exec**.

The **#include** construct tells Asterisk to read in the contents of another configuration file, and act as though the contents were at this location in this configuration file. The syntax is **#include filename**, where **filename** is the name of the file you'd like to include. This construct is most often used to break a large configuration file into smaller pieces, so that it's more manageable. The asterisk/star character will be parsed in the path, allowing for the inclusion of an entire directory of files. If the target file specified does not exist, then Asterisk will not load the module that contains configuration with the #include directive.

The **#tryinclude** construct is the same as #include except it won't stop Asterisk from loading the module when the target file does not exist.

The **#exec** takes this one step further. It allows you to execute an external program, and place the output of that program into the current configuration file. The syntax is **#exec program**, where **program** is the name of the program you'd like to execute.

The **#exec**, **#include**, and #**tryinclude** constructs do not work in the following configuration files:

- asterisk.conf
- modules.conf

Enabling #exec Functionality

The #exec construct is not enabled by default, as it has some risks both in terms of performance and security. To enable this functionality, go to the **asterisk.conf** configuration file (by default located in */etc/asterisk*) and set **execincludes=yes** in the **[options]** section. By default both the **[options]** section heading and the **execincludes=yes** option have been commented out, you you'll need to remove the semicolon from the beginning of both lines.

Examples

Let's look at example of both constructs in action. This is a generic example meant to illustrate the syntax usage inside a configuration file.

```
[section-name]
setting=true
#include otherconfig.conf      ; include another configuration file
#include my_other_files/*.conf ; include all .conf files in the subdirectory
my_other_files
#exec otherprogram             ; include output of otherprogram
```

You can use #tryinclude if there is any chance the target file may not exist and you still want Asterisk to load the configuration for the module.

Here is a more realistic example of how #exec might be used with real-world commands.

```
#exec /usr/bin/curl -s http://example.com/mystuff > /etc/asterisk/mystuff
#include mystuff
```

Adding to an existing section

If you want to add settings to an existing section of a configuration file (either later in the file, or when using the **#include** and **#exec** constructs), add a plus sign in parentheses after the section heading, as shown below:

```
[section-name]
setting1=value1

[section-name](+)
setting2=value2
```

This example shows that the **setting2** setting was added to the existing section of the configuration file.

If the section you're adding to appears more than once in the config, such as an endpoint and aor named the same in a pjsip.conf file, the section added to will be the first one defined unless you add a filter qualifier.

Without a qualifier:

This will fail because default_expiration isn't valid for an endpoint

```
[101]
type=endpoint
allow=ulaw

[101]
type=aor
default_expiration=3600

[101](+)
default_expiration=1200
```

With qualifiers:

This works because the filters ensure that the additions are to the correct objects.

```
[101]
type=endpoint
allow=ulaw

[101]
type=aor
default_expiration=3600

[101](+type=aor)
default_expiration=1200

[101](+type=endpoint)
allow=g722
```

You're not limited to filtering by the type parameter and you can even use regular expressions in the name or value.

A weird and not so useful example

```
[101]
type=endpoint
allow=ulaw

[101]
type=aor
default_expiration=3600

[101](+default_.*=36[0-9][0-9])
default_expiration=1200

[101](+type=endpoint)
allow=g722
```

You can also include multiple filters.

Another weird and not so useful example

```
[101]
type=endpoint
allow=ulaw

[101]
type=aor
default_expiration=3600

[101](+type=aor&default_.*=36[0-9][0-9])
default_expiration=1200

[101](+type=endpoint)
allow=g722
```

And finally, you can elect to include or restrict parameters inherited from templates in the search.

The final weird and not so useful example. This will NOT match because default_expiration is defined in the parent template.

```
[101]
type=endpoint
allow=ulaw

[aor_template](!)
type=aor
default_expiration=3600

[101](aor_template)

[101](+TEMPLATES=restrict&default_.*=36[0-9][0-9])
default_expiration=1200

[101](+type=endpoint)
allow=g722
```

Templates

Another construct we can use within most Asterisk configuration files is the use of templates. A template is a section of a configuration file that is only used as a base (or template, as the name suggests) to create other sections from.

Template Syntax

To define a section as a template **only** (not to be loaded for use as configuration by itself), place an exclamation mark in parentheses after the section heading, as shown in the example below.

```
[template-name](!)
setting=value
```

Alternatively the Using Templates page will also discuss how to have a section inherit another section's settings without defining a template. In effect, using an "active" or "live" configuration section as your template.

Using Templates

To use a template when creating another section, simply put the template name in parentheses after the section heading name, as shown in the example below. If you want to inherit from multiple templates, use commas to separate the template names).

```
[template-name](!)
setting=value

[template-2](!)
setting2=value2

[not-a-template]
setting4=value4

[section-name](template-name,template-2,not-a-template)
setting3=value3
```

This works even when the section name referenced in parentheses is **not defined as a template** as in the case of the "not-a-template" section.

The newly-created section will inherit all the values and objects defined in the template(s), as well as any new settings or objects defined in the newly-created section. The settings and objects defined in the newly-created section override settings or objects of the same name from the templates. Consider this example:

```
[test-one](!)
permit=192.168.0.2
host=alpha.example.com
deny=192.168.0.1

[test-two](!)
permit=192.168.1.2
host=bravo.example.com
deny=192.168.1.1

[test-three](test-one,test-two)
permit=192.168.3.1
host=charlie.example.com
```

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

```
[test-three]
permit=192.168.0.2
host=alpha.example.com
deny=192.168.0.1
permit=192.168.1.2
host=bravo.example.com
deny=192.168.1.1
permit=192.168.3.1
host=charlie.example.com
```

chan_sip Template Example

Here is a more extensive and realistic example from the chan_sip channel driver's sample configuration file.

```
[basic-options](!)                ; a template
        dtmfmode=rfc2833
        context=from-office
        type=friend

[natted-phone](!,basic-options)   ; another template inheriting basic-options
        nat=yes
        directmedia=no
        host=dynamic

[public-phone](!,basic-options)   ; another template inheriting basic-options
        nat=no
        directmedia=yes

[my-codecs](!)                    ; a template for my preferred codecs
        disallow=all
        allow=ilbc
        allow=g729
        allow=gsm
        allow=g723
        allow=ulaw

[ulaw-phone](!)                   ; and another one for ulaw-only
        disallow=all
        allow=ulaw

; and finally instantiate a few phones
;
; [2133](natted-phone,my-codecs)
;        secret = peekaboo
; [2134](natted-phone,ulaw-phone)
;        secret = not_very_secret
; [2136](public-phone,ulaw-phone)
;        secret = not_very_secret_either
```

Database Support Configuration

Top-level page for information about Database support.

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:

1. ODBC: Support for UnixODBC, integrated into Asterisk The UnixODBC subsystem supports many different databases, please check www.unixodbc.org for more information.
2. MySQL: Native support for MySQL, integrated into Asterisk
3. 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.

> To use static realtime with certain core configuration files (e.g. `features.conf`, `cdr.conf`, `cel.conf`, `indications.conf`, etc.) the realtime backend you wish to use must be preloaded in `modules.conf`.
>
> ```
> [modules]
> preload => res_odbc.so
> preload => res_config_odbc.so
> ```

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>,<res_<driver>.conf class 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
- sipregs - SIP registrations
- iaxpeers, iaxusers - IAX2 peers and users
- voicemail - Voicemail accounts
- extensions - Realtime extensions (switch)
- meetme - MeetMe conference rooms
- queues - Queues
- queue_members - Queue members
- musiconhold - Music On Hold classes
- queue_log - Queue logging

Voicemail storage with the support of ODBC described in ODBC Voicemail Storage.

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.

You may notice a performance issue under high load using UnixODBC. The UnixODBC driver supports threading but you must specifically enable threading within the UnixODBC configuration file like below for each engine:

```
Threading = 2
```

This will enable the driver to service many requests at a time, rather than serially.

Notes on use of the sipregs family

The community provided some additional recommendations on the JIRA issue ASTERISK-21315:

- It is a good idea to avoid using sipregs altogether by NOT enabling it in extconfig. Using a writable sipusers table should be enough. If you cannot write to your base sipusers table because it is readonly, you could consider making a separate sipusers view that joins the readonly table with a writable sipregs table.

SIP Realtime, MySQL table structure

Here is the table structure used by MySQL for Realtime SIP friends

```
#
# Table structure for table `sipfriends`
#

CREATE TABLE IF NOT EXISTS `sipfriends` (
      `id` int(11) NOT NULL AUTO_INCREMENT,
      `name` varchar(10) NOT NULL,
      `ipaddr` varchar(15) DEFAULT NULL,
      `port` int(5) DEFAULT NULL,
      `regseconds` int(11) DEFAULT NULL,
      `defaultuser` varchar(10) DEFAULT NULL,
      `fullcontact` varchar(35) DEFAULT NULL,
      `regserver` varchar(20) DEFAULT NULL,
      `useragent` varchar(20) DEFAULT NULL,
      `lastms` int(11) DEFAULT NULL,
      `host` varchar(40) DEFAULT NULL,
      `type` enum('friend','user','peer') DEFAULT NULL,
      `context` varchar(40) DEFAULT NULL,
      `permit` varchar(40) DEFAULT NULL,
      `deny` varchar(40) DEFAULT NULL,
      `secret` varchar(40) DEFAULT NULL,
      `md5secret` varchar(40) DEFAULT NULL,
      `remotesecret` varchar(40) DEFAULT NULL,
      `transport` enum('udp','tcp','udp,tcp','tcp,udp') DEFAULT NULL,
      `dtmfmode` enum('rfc2833','info','shortinfo','inband','auto') DEFAULT NULL,
      `directmedia` enum('yes','no','nonat','update') DEFAULT NULL,
      `nat` enum('yes','no','never','route') DEFAULT NULL,
      `callgroup` varchar(40) DEFAULT NULL,
      `pickupgroup` varchar(40) DEFAULT NULL,
      `language` varchar(40) DEFAULT NULL,
      `allow` varchar(40) DEFAULT NULL,
      `disallow` varchar(40) DEFAULT NULL,
      `insecure` varchar(40) DEFAULT NULL,
      `trustrpid` enum('yes','no') DEFAULT NULL,
      `progressinband` enum('yes','no','never') DEFAULT NULL,
      `promiscredir` enum('yes','no') DEFAULT NULL,
      `useclientcode` enum('yes','no') DEFAULT NULL,
      `accountcode` varchar(40) DEFAULT NULL,
      `setvar` varchar(40) DEFAULT NULL,
      `callerid` varchar(40) DEFAULT NULL,
      `amaflags` varchar(40) DEFAULT NULL,
      `callcounter` enum('yes','no') DEFAULT NULL,
      `busylevel` int(11) DEFAULT NULL,
      `allowoverlap` enum('yes','no') DEFAULT NULL,
      `allowsubscribe` enum('yes','no') DEFAULT NULL,
      `videosupport` enum('yes','no') DEFAULT NULL,
      `maxcallbitrate` int(11) DEFAULT NULL,
      `rfc2833compensate` enum('yes','no') DEFAULT NULL,
      `mailbox` varchar(40) DEFAULT NULL,
      `session-timers` enum('accept','refuse','originate') DEFAULT NULL,
      `session-expires` int(11) DEFAULT NULL,
      `session-minse` int(11) DEFAULT NULL,
      `session-refresher` enum('uac','uas') DEFAULT NULL,
      `t38pt_usertpsource` varchar(40) DEFAULT NULL,
      `regexten` varchar(40) DEFAULT NULL,
      `fromdomain` varchar(40) DEFAULT NULL,
      `fromuser` varchar(40) DEFAULT NULL,
      `qualify` varchar(40) DEFAULT NULL,
      `defaultip` varchar(40) DEFAULT NULL,
      `rtptimeout` int(11) DEFAULT NULL,
      `rtpholdtimeout` int(11) DEFAULT NULL,
      `sendrpid` enum('yes','no') DEFAULT NULL,
      `outboundproxy` varchar(40) DEFAULT NULL,
      `callbackextension` varchar(40) DEFAULT NULL,
      `registertrying` enum('yes','no') DEFAULT NULL,
      `timert1` int(11) DEFAULT NULL,
      `timerb` int(11) DEFAULT NULL,
      `qualifyfreq` int(11) DEFAULT NULL,
      `constantssrc` enum('yes','no') DEFAULT NULL,
      `contactpermit` varchar(40) DEFAULT NULL,
      `contactdeny` varchar(40) DEFAULT NULL,
      `usereqphone` enum('yes','no') DEFAULT NULL,
      `textsupport` enum('yes','no') DEFAULT NULL,
      `faxdetect` enum('yes','no') DEFAULT NULL,
      `buggymwi` enum('yes','no') DEFAULT NULL,
      `auth` varchar(40) DEFAULT NULL,
      `fullname` varchar(40) DEFAULT NULL,
      `trunkname` varchar(40) DEFAULT NULL,
      `cid_number` varchar(40) DEFAULT NULL,
      `callingpres`
enum('allowed_not_screened','allowed_passed_screen','allowed_failed_screen','allowed','prohib_not_screened','prohib_passed_screen
','prohib_failed_screen','prohib') DEFAULT NULL,
      `mohinterpret` varchar(40) DEFAULT NULL,
      `mohsuggest` varchar(40) DEFAULT NULL,
      `parkinglot` varchar(40) DEFAULT NULL,
      `hasvoicemail` enum('yes','no') DEFAULT NULL,
      `subscribemwi` enum('yes','no') DEFAULT NULL,
      `vmexten` varchar(40) DEFAULT NULL,
      `autoframing` enum('yes','no') DEFAULT NULL,
      `rtpkeepalive` int(11) DEFAULT NULL,
      `call-limit` int(11) DEFAULT NULL,
      `g726nonstandard` enum('yes','no') DEFAULT NULL,
      `ignoresdpversion` enum('yes','no') DEFAULT NULL,
      `allowtransfer` enum('yes','no') DEFAULT NULL,
      `dynamic` enum('yes','no') DEFAULT NULL,
      PRIMARY KEY (`id`),
      UNIQUE KEY `name` (`name`),
      KEY `ipaddr` (`ipaddr`,`port`),
      KEY `host` (`host`,`port`)
) ENGINE=MyISAM;
```

Sorcery

Under Construction

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Sorcery Overview

Added in Asterisk 12, Asterisk has a data abstraction and object persistence CRUD API called Sorcery. Sorcery provides Asterisk modules with a useful abstraction on top of the many storage mechanisms in Asterisk. Such as the:

- Asterisk Database
- Static Configuration Files
- Asterisk Realtime Architecture
- In-Memory

Sorcery also provides a caching service as well as the capability for push configuration through the Asterisk REST Interface. See the section ARI Push Configuration for more information on that topic.

Modules Supporting Sorcery

The PJSIP modules and resources were the first to use the Sorcery DAL. All future modules which utilize Sorcery for object persistence must have a column named id within their schema when using the Sorcery realtime module. This column must be able to contain a string of up to 128 characters in length.

Sorcery API Actions

AMI actions existing at the time of Asterisk 14.2.1

- SorceryMemoryCacheExpire
- SorceryMemoryCacheExpireObject
- SorceryMemoryCachePopulate
- SorceryMemoryCacheStale
- SorceryMemoryCacheStaleObject

Sorcery Functions

Sorcery functions existing at the time of Asterisk 14.2.1

- AST_SORCERY()

Sorcery Mapping Configuration

Users can configure a hierarchy of data storage layers for specific modules in sorcery.conf.

You can view the sorcery.conf sample in your configs/samples/ Asterisk source subdirectory. Or you can check it out on github: https://github.com/asterisk/asterisk/blob/master/configs/samples/sorcery.conf.sample

We've included roughly the same instructions below while taking advantage of wiki formatting.

Constructing a Mapping

To allow configuration of where and how an object is persisted, object mappings can be defined within sorcery.conf on a per-module basis. The mapping consists of the **object type**, **options**, **wizard name**, and **wizard configuration data**.

Format

The basic format follows:

```
[module_name]    ;The brackets around the module name are literal, just as in most other Asterisk configuration files.
object_type[/options] = wizard_name[,wizard_configuration_data]   ;Bracketed items here are optional
```

Module name

Object/Wizard mappings are defined within sections denoted by the module name in brackets. The section name must match the module.

Object types

Note that an object type can have multiple mappings defined. Each mapping will be consulted in the order in which it appears within the configuration file. This means that if you are configuring a wizard as a cache it should appear as the first mapping so the cache is consulted before all other mappings.

Object types available depend on the modules loaded and what objects they provide. There are PJSIP types for all the configuration objects in PJSIP, such as endpoint, auth,aor, etc. You can find a more exhaustive list of PJSIP objects in the Sorcery Caching page.

Wizards

Wizards are the persistence mechanism for objects. They are loaded as Asterisk modules and register themselves with the sorcery core. All implementation specific details of how objects are persisted is isolated within wizards.

A wizard can optionally be marked as an object cache by adding "/cache" to the object type within the mapping. If an object is returned from a non-object cache it is immediately given to the cache to be created. Multiple object caches can be configured for a single object type.

Wizards available at the time of writing:

- astdb
- config
- memory
- realtime
- memory_cache (For further details on this wizard type see the documentation here)

Example Mapping Configurations

The following object mappings are used by the unit test to test certain functionality of sorcery.

```
[test_sorcery_section]
test=memory
[test_sorcery_cache]
test/cache=test
test=memory
```

The following object mapping is the default mapping of external MWI mailbox objects to give persistence to the message counts.

```
[res_mwi_external]
mailboxes=astdb,mwi_external
```

The following object mappings set PJSIP objects to use realtime database mappings from extconfig with the table names used when automatically generating configuration from the alembic script.

```
[res_pjsip]
endpoint=realtime,ps_endpoints
auth=realtime,ps_auths
aor=realtime,ps_aors
domain_alias=realtime,ps_domain_aliases
contact=realtime,ps_contacts

[res_pjsip_endpoint_identifier_ip]
identify=realtime,ps_endpoint_id_ips
```

PJSIP Default Wizard Configurations

When configuring PJSIP sorcery mappings it can be useful to allow both the configuration file and other wizards to be used. The below configuration matches the default configuration for the PJSIP sorcery usage.

```
[res_pjsip]
auth=config,pjsip.conf,criteria=type=auth
domain_alias=config,pjsip.conf,criteria=type=domain_alias
global=config,pjsip.conf,criteria=type=global
system=config,pjsip.conf,criteria=type=system
transport=config,pjsip.conf,criteria=type=transport
aor=config,pjsip.conf,criteria=type=aor
endpoint=config,pjsip.conf,criteria=type=endpoint
contact=astdb,registrator

[res_pjsip_endpoint_identifier_ip]
identify=config,pjsip.conf,criteria=type=identify

[res_pjsip_outbound_publish]
outbound-publish=config,pjsip.conf,criteria=type=outbound-publish

[res_pjsip_outbound_registration]
registration=config,pjsip.conf,criteria=type=registration
```

Sorcery Caching

Since Asterisk 12, Asterisk has had a generic data access/storage layer called "sorcery", with pluggable "wizards" that each create, retrieve, update, and delete data from various backends. For instance, there is a sorcery wizard that reads configuration data from .conf files. There is a sorcery wizard that uses the Asterisk Realtime Architecture to interface with databases and other alternative backends. There are also sorcery wizards that use the AstDB and a simple in-memory container.

Starting in Asterisk 13.5.0, a new "memory_cache" wizard has been created. This allows for a cached copy of an object to be stored locally in cases where retrieval from a remote backend (such as a relational database) might be expensive. Memory caching is a flexible way to provide per object type caching, meaning that you are not forced into an all-or-nothing situation if you decide to cache. Caching also provides configuration options to allow for cached entries to automatically be updated or expired.

Cachable Objects

Not all configurable objects are managed by sorcery. The following is a list of objects that are managed by the sorcery subsystem in Asterisk.

- PJSIP endpoint
- PJSIP AOR
- PJSIP contact
- PJSIP identify
- PJSIP ACL
- PJSIP resource_list
- PJSIP phoneprov
- PJSIP registration
- PJSIP subscription_persistence
- PJSIP inbound-publication
- PJSIP asterisk-publication
- PJSIP system
- PJSIP global
- PJSIP auth
- PJSIP outbound-publish
- PJSIP transport
- External MWI mailboxes

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When Should I Use Caching?

First, if you are using default sorcery backends for objects (i.e. you have not altered `sorcery.conf` at all), then caching will likely not have any positive effect on your configuration. However, if you are using the "realtime" sorcery wizard or any other that retrieves data from outside the Asterisk process, then caching could be a good fit for certain object types.

There are two overall flavors of caching. The first type is a method that caches individually retrieved objects. In other words, when an object is retrieved from the backend, that object is also placed in the cache. That object can then be retrieved individually from the cache the next time it is needed. This type of caching works well for values that are

- Read more often than they are written
- Retrieved one-at-a-time.

For the first point, you will be able to know this better than anyone else. For instance, if you tend to configure PJSIP authentication very infrequently, but there are many calls, subscriptions, and qualifies that require authentication, then caching PJSIP auths is probably a good idea. If you are constantly tweaking PJSIP endpoint configuration for some reason, then you might find that caching isn't necessarily as good a fit for PJSIP endpoints.

For the second point, it may not always be obvious which types of objects are typically looked up one-at-a-time and which ones are typically looked up in multiples. The following object types are likely a bad fit for caching since they tend to be looked up in multiples:

- PJSIP contact
- PJSIP identify
- PJSIP global
- PJSIP system
- PJSIP registrations
- PJSIP ACLs
- PJSIP outbound-publishes
- PJSIP subscription_persistence

The rest of the objects listed are most typically retrieved one-at-a-time and would be good for caching in this manner.

The second type of caching instead pulls all objects from the database up front. These objects are all stored in memory, and since it is known that the cache has all objects, multiple objects can be retrieved from the cache at once. This means that **any** object type is a good fit for this type of caching.

How do I enable Caching?

If you are familiar with enabling realtime for a sorcery object, then enabling caching should not seem difficult. Here is an example of what it might look like if you have configured PJSIP endpoints to use a cache:

sorcery.conf

```
[res_pjsip]
endpoint/cache=memory_cache
endpoint=realtime,ps_endpoints
```

Let's break this down line-by-line. The first line starts with "endpoint/cache". "endpoint" is the name of the object type. "/cache" is a cue to sorcery that the wizard being specified on this line is a cache. And "memory_cache" is the name of the caching wizard that has been added in Asterisk 14.0.0. The second line is the familiar line that specifies that endpoints can be retrieved from realtime by following the "ps_endpoints" configuration line in `extconfig.conf`.

The order of the lines is important. You will want to specify the memory_cache wizard before the realtime wizard so that the memory_cache is looked in before realtime when retrieving an item.

How does the cache behave?

By default, the cache will simply store objects in memory. There will be no limits to the number of objects stored in the cache, and the items in the cache will never be updated or expire, no matter whether the backend has been updated to have new configuration values. The cache entry in `sorcery.conf` is configurable, though, so you can modify the behavior to suit your setup. Options for the memory cache are comma-separated on the line in `sorcery.conf` that defines the cache. For instance, you might have something like the following:

sorcery.conf

```
[res_pjsip]
endpoint/cache = memory_cache,maximum_objects=150,expire_on_reload=yes,object_lifetime_maximum=3600
endpoint = realtime,ps_endpoints
```

The following configuration options are recognized by the memory cache:

name

The name of a cache is used when referring to a specific cache when running an AMI or CLI command. If no name is provided for a cache, then the default is <configuration section>/<object type>. PJSIP endpoints, for instance, have a default cache name of "res_pjsip/endpoint".

maximum_objects

This option specifies the maximum number of objects that can be in the cache at a given time. If the cache is full and a new item is to be added, then the oldest item in the cache is removed to make room for the new item. If this option is not set or if its value is set to 0, then there is no limit on the number of objects in the cache.

object_lifetime_maximum

This option specifies the number of seconds an object may occupy the cache before it is automatically removed. This time is measured from when the object is initially added to the cache, not the time when the object was last accessed. If this option is not set or if its value is set to 0, then objects will stay in the cache forever.

object_lifetime_stale

This option specifies the number of seconds an object may occupy the cache until it is considered stale. When a stale object is retrieved from the cache, the stale object is given to the requestor, and a background task is initiated to update the object in the cache by querying whatever backend stores are configured. If a new object is retrieved from the backend, then the stale cached object is replaced with the new object. If the backend no longer has an object with the same ID as the one that has become stale, then the stale object is removed from the cache. If this option is not set or if its value is 0, then objects in the cache will never be marked stale.

expire_on_reload

This option specifies whether a reload of a module should automatically remove all of its objects from the cache. For instance, if this option is enabled, and you are caching PJSIP endpoints, then a module reload of `res_pjsip.so` would clear all PJSIP endpoints from the cache. By default this option is not enabled.

What AMI and CLI commands does the cache provide?

CLI

sorcery memory cache show <cache name>

This CLI command displays the configuration for the given cache and tells the number of items currently in the cache.

sorcery memory cache dump <cache name>

This CLI command displays all objects in the given cache. In addition to the name of the object, the command also displays the number of seconds until the object becomes stale and the number of seconds until the object will be removed from the cache.

sorcery memory cache expire <cache name> [object name]

This CLI command is used to remove objects from a given cache. If no object name is specified, then all objects in the cache are removed. If an object name is specified, then only the specified object is removed.

sorcery memory cache stale <cache name> [object_name]

This CLI command is used to mark an item in the cache as stale. If no object name is specified, then all objects in the cache are marked stale. If an object name is specified, then only the specified object is marked stale. For information on what it means for an object to be stale, see here

AMI

Since AMI commands are XML-documented in the source, there should be a dedicated wiki page with this information.

SorceryMemoryCacheExpireObject

This command has the following syntax:

```
Action: SorceryMemoryCacheExpireObject
Cache: <cache name>
Object: <object name>
```

Issuing this command will cause the specified object in the specified cache to be removed. Like all AMI commands, an optional ActionID may be specified.

SorceryMemoryCacheExpire

This command has the following syntax:

```
Action: SorceryMemoryCacheExpire
Cache: <cache name>
```

Issuing this command will cause all objects in the specified cache to be removed. Like all AMI commands, an optional ActionID may be specified.

SorceryMemoryCacheStaleObject

This command has the following syntax:

```
Action: SorceryMemoryCacheStaleObject
Cache: <cache name>
Object: <object name>
```

Issuing this command will cause the specified object in the specified cache to be marked as stale. For more information on what it means for an object to be stale, see here. Like all AMI commands, an optional ActionID may be specified.

SorceryMemoryCacheStale

This command has the following syntax:

```
Action: SorceryMemoryCacheStale
Cache: <cache name>
```

Issuing this command will cause all objects in the specified cache to be marked as stale. For more information on what it means for an object to be stale, see here. Like all AMI commands, an optional ActionID may be specified.

What are some caching strategies?

Hands-on or hands-off?

The hands-on approach to caching is that you set your cache to have no maximum number of objects, and objects never expire or become stale on their own. Instead, whenever you make changes to the backend store, you issue an AMI or CLI command to remove objects or mark them stale. The hands-off approach to caching is to fine-tune the maximum number of objects, stale timeout, and expire timeout such that you never have to think about the cache again after you set it up the first time.

The hands-on approach is a good fit either for installations where configuration rarely changes, or where there is some automation involved when configuration changes are made. For instance, if you are setting up a PBX for a small office where you are likely to make configuration changes a few times a year, then the hands-on approach may be a good fit. If your configuration is managed through a GUI that fires off a script when the "submit" button is pressed, then the hands-on approach may be a good fit since your scripts can be modified to manually expire objects or mark them stale. The main disadvantage to the hands-on approach is that if you forget to manually expire a cached object or if you make a mistake in your tooling, you're likely to have some big problems since configuration changes will seemingly not have any effect.

The hands-off approach is a good fit for configurations that change frequently or for deployments with inconsistent usage among users. If configuration is changing frequently, then it makes sense for objects in the cache to become stale and automatically get refreshed. If you have some users on the system that maybe use the system once a week, it makes sense for them to get removed from the cache as more frequent users occupy it. The biggest disadvantage to the hands-off approach is the potential for churn if your settings are overzealous. For instance, if you allow a maximum of 15 objects in a cache but it's common for 20 to be used, then the cache may constantly be shuffling which objects are stored in it. Similarly, if you set a stale object timeout low, then it is possible that objects in the cache will frequently be replacing themselves with identical copies.

There is also a hybrid approach. In the hybrid approach, you're mostly hands-off, but you can be hands-on for "emergency" changes. For instance, if there is a misconfiguration that is resulting in calls not being able to be sent to a user, then you may want to get that configuration updated and immediately remove the cached object so that the new configuration can be added to the cache instead.

Expire or Stale?

One question that may enter your mind is whether to have objects expire or whether they should become stale.

Letting objects expire has the advantage that they no longer are occupying cache space. For objects that are infrequently accessed, this can be a good thing since they otherwise will be taking up space and being useless. For objects that are accessed frequency, expiration is likely a bad choice. This is because if the object has been removed from the cache, then attempting to retrieve the object will require a cache miss, followed by a backend hit to retrieve the object. If the object configuration has not been altered, then this equates to a waste of cycles.

Letting objects become stale has the advantage that retrievals will always be quick. This is because even if the object is stale, the stale cached object is returned. It's left up to a background task to update the cached object with new data from the backend. The main disadvantage to objects being stale is that infrequently accessed objects will remain in the cache long after their useful lifetime.

One approach to take is a hybrid approach. You can set objects to become stale after an amount of time, and then later, the object will become expired. This way, objects that are retrieved frequently will stay up to date as they become stale, and objects that are rarely accessed will expire after a while.

An example configuration

Below is a sample sorcery.conf file that uses realtime as the backend store for some PJSIP objects.

sorcery.conf

```
 [res_pjsip]
endpoint/cache = memory_cache,object_lifetime_stale=600,object_lifetime_maximum=1800,expire_on_reload=yes
endpoint = realtime,ps_endpoints
auth/cache=memory_cache,expire_on_reload=yes
auth = realtime,ps_auths
aor/cache = memory_cache,object_lifetime_stale=1500,object_lifetime_maximum=1800,expire_on_reload=yes
aor = realtime,ps_aors
```

In this particular setup, the administrator has set different options for different object caches.

- For endpoints, the administrator decided that cached endpoint configuration may occasionally need updating. Endpoints therefore will be marked stale after 10 minutes. If an endpoint happens to make it 30 minutes without being retrieved, then the endpoint will be ejected from the cache entirely.
- For auths, the administrator realized that auth so rarely changes that there is no reason to set any sort of extra parameters. On those odd occasions where auth is updated, the admin will just manually expire the old auth.
- AORs, like endpoints, may require refreshing after a while, but because the AOR configurations are changed much more infrequently, it

takes 25 minutes for the object to become stale.
- All objects expire on a reload since a reload likely means that there was some large-scale change and everything should start from scratch.

This is just an example. It is not necessarily going to be a good fit for everyone's needs.

Pre-caching all objects

When introducing caching, we discussed a second form of caching, where all objects are pre-loaded from the realtime backend and placed in the cache. Why would this be necessary?

Consider if you have configured AORs to be cached. At some point, Asterisk tried to retrieve AOR "alice". This AOR was found in a database, and it was added to the cache. Now, Asterisk gets told to retrieve all AORs. If Asterisk just looks in the cache, all it will get is "alice". The cache has no way of knowing whether it has all values cached or not. Thus, rather than even asking the cache, Asterisk skips straight to going to the database directly. The cache did not serve us much good there.

However, Asterisk can be told to pre-load all objects of a certain type and cache those. This way, the cache knows that it has all objects of a certain type. Therefore, if multiple objects need to be retrieved, Asterisk can ask the cache for those items and not have to hit the realtime backend at all. Here's an example configuration:

sorcery.conf

```
[res_pjsip]
identify/cache =
memory_cache,object_lifetime_stale=600,object_lifetime_maximum=1800,expire_on_reload=yes,full_backend_cache=y
es
identify = realtime,ps_endpoint_id_ips
```

Just like with the previous section's configuration, we have configured an object to be retrieved from realtime and cached in memory. Notice, though, that we have added `full_backend_cache=yes` to the end of the line. This is what causes Asterisk to pre-cache the objects. Normally, PJSIP "identify" objects would be a bad fit for caching since we tend to retrieve them all at once rather than one-at-a-time. By pre-caching all objects though, Asterisk can now retrieve all of them directly from the cache. Also notice that the other caching options are still relevant here. Rather than having the options apply to individual objects, they now apply to all of the retrieved objects. So if Asterisk retrieved 10 identifys during pre-cache, when the stale lifetime rolls around, all 10 will be marked stale and Asterisk will once again retrieve all of the objects from the backend.

CLI

sorcery memory cache populate <cache name>

This CLI command is used to manually tell Asterisk to remove all objects from the cache and repopulate that cache with all objects from the backend.

AMI

SorceryMemoryCachePopulate

This command has the following syntax:

```
Action: SorceryMemoryCachePopulate
Cache: <cache name>
```

Issuing this command has the same effect as the CLI "sorcery memory cache populate" command. It will invalidate all cached entries from the particular cache and then repopulate it with all objects from the backend.

When to use this Caching method

Pre-caching the entire backend is a good idea if you find that caching individual objects is not working for you. The tradeoff is that you will use more memory this way since all objects are retrieved from the cache at once.

Asterisk Internal Database

Asterisk comes with a database that is used internally and made available for Asterisk programmers and administrators to use as they see fit.

Asterisk versions up to 1.8 used the Berkeley DB, and **in version 10** the project moved to the **SQLite3 database**. You can read about database migration between those major versions in the section SQLite3 astdb back-end.

Purpose of the internal database

The database really has two purposes:

1. Asterisk uses it to store information that needs to persist between reloads/restarts. Various modules use it for this purpose automatically.
2. Users can use it to store arbitrary data. This is done using a variety of dialplan applications and functions such as:
    - Functions:
        - DB
        - DB_DELETE
        - DB_EXISTS
        - DB_KEYS
    - Application: DBdeltree

The functions and applications for Asterisk 11 are linked above, but you should look at the documentation for the version you have deployed.

Database commands on the CLI

Sub-commands under the command "database" allow a variety of functions to be performed on or with the database.

```
*CLI> core show help database
database del                   -- Removes database key/value
database deltree               -- Removes database keytree/values
database get                   -- Gets database value
database put                   -- Adds/updates database value
database query                 -- Run a user-specified query on the astdb
database show                  -- Shows database contents
database showkey               -- Shows database contents
```

SQLite3 astdb back-end

⚠ Starting with **Asterisk 10** , Asterisk uses **SQLite3** for its internal database instead of the Berkeley DB database used by Asterisk 1.8 and previous versions.

Every effort has been made to make this transition as automatic and painless for users as possible. This page will describe the upgrade process, any potential problems, and the appropriate solutions to those problems.

The upgrade process

Asterisk 10 will attempt to upgrade any existing old-style Berkeley DB internal database to the new SQLite 3 database format. This conversion process is accomplished at run-time with the astdb2sqlite3 utility which builds by default in Asterisk 10. The astdb2sqlite3 utility will also be forcibly built even if deselected using menuselect if the build process determines that there is an old-style Berkeley DB and no new SQLite3 DB exists.

When Asterisk 10 is run, as part of the initialization process it checks for the existence of the SQLite3 database. If it doesn't exist and an old-style Berkeley DB does exist, it will attempt to convert the Berkeley DB to the SQLite3 format. If no existing database exists, a new SQLite 3 database will be created. If the conversion fails, a warning will be displayed with instructions describing possible fixes and Asterisk will exit.

ⓘ It is important that you perform the upgrade process at the same permission level that you expect Asterisk to run at. For example, if you upgrade as root, but run Asterisk as a user with lower permissions, the SQLite3 database created as part of the upgrade will not be able to be accessed by Asterisk.

Troubleshooting an upgrade

Symptoms

./configure displays the warning: *** Please install the SQLite3 development package.

Cause

To build Asterisk 10, the SQLite 3 development libraries must be installed.

Solution

On Debian-based distros including Ubuntu, these libraries may be installed by running 'sudo apt-get install libsqlite3-dev'. For Red Hat-based distros including Fedora and Centos these libraries may be installed by running (as root) 'yum install sqlite3-devel'.

Asterisk exits displaying the warning: *** Database conversion failed!

Cause

Asterisk 10 could not find the astdb2sqlite3 utility to convert the old Berkeley DB to SQLite 3.

Solution

Make sure that astdb2sqlite3 is selected for build in the Utilities section when running 'make menuselect'. Be sure to re-run 'make' and 'make install' after selecting astdb2sqlite3 for build.

Cause

Asterisk is unable to write to the directory specified in asterisk.conf as the 'astdbdir'

Solution

SQLite 3 creates a journal file in the 'astdbdir' specified in asterisk.conf. It is important that this directory is writable by the user Asterisk runs as. This involves either modifying the permissions of the 'astdbdir' directory listed in asterisk.conf, or changing the 'astdbdir' option to a directory for which the user running Asterisk already has write permission. This is generally only a problem if Asterisk is run as a non-root user.

Cause

If Asterisk 10 was installed via a distro-specific package, it is possible that the distro forgot to package the astdb2sqlite3 utility.

Solution

Run 'which astdb2sqlite3' from a terminal. If no filenames are displayed, then astd2sqlite3 has not be installed. Check if the distro includes it in another asterisk related package, or download the Asterisk 10 source from the Asterisk.org website and follow the normal build instructions. Instead of running 'make install', manually run 'utils/astdb2sqlite3 /var/lib/asterisk/astdb' from the Asterisk source directory, replacing '/var/lib/asterisk' with the 'astdbdir' directory listed in asterisk.conf. After the conversion, the distro-supplied Asterisk should successfully run.

Migrating back from Asterisk 10 to Asterisk 1.8

If migrating back to Asterisk 1.8 from Asterisk 10, it is possible to convert the SQLite 3 internal database back to the Berkeley DB format that Asterisk 1.8 uses by using the astdb2bdb utility found in the utils/ directory of the Asterisk 10 source. To build, make sure that astdb2bdb is selected in the Utilities section when running 'make menuselect'. Running 'utils/astdb2bdb /var/lib/asterisk/astdb.sqlite3' (replacing '/var/lib/asterisk' with the 'astdbdir' directory listed in asterisk.conf) will produce a file named 'astdb' in the current directory. Back up any existing astdb file in the astdbdir directory and replace it with the newly created astdb file.

Key Concepts

Under Construction

Top-level page for a section dealing with concepts of the key moving pieces in Asterisk that an **administrator** needs to understand. Channels, Bridges, Frames, etc.

Bridges

Overview

In Asterisk, a bridge is the construct that shares media among Channels. While a channel represents the path of communication between Asterisk and some device, a bridge is how that path of communication is shared. While channels are in a bridge, their media is exchanged in a manner dictated by the bridge's type. While we generally think of media being directed among channels, media can also be directed from Asterisk to the channels in a bridge. This can be the case in some conferences, where Music on Hold (MoH) or announcements are played for waiting channels.

On this Page

Creation

Generally, a bridge is created when Asterisk knows that two or more channels want to communicate. A variety of applications and API calls can cause a bridge to be created. Some of these include:

- Dial - a bridge is created for the two channels when the outbound channel answers. Both the inbound channel and the outbound channel are placed into the bridge.
    - DTMF feature invocations available from Dial() can create, modify, or destroy bridges.
- Bridge - this directly creates a new bridge and places two channels into the bridge. Unlike Dial, both channels have to already exist.
- BridgeWait (Asterisk 12+) - creates a special holding bridge and places a channel into it. Any number of channels may join the holding bridge, which can entertain them in a variety of ways.
- MeetMe/ConfBridge - both of these applications are used for conferencing, and can support multiple channels together in the same bridge.
- Page - a conferencing bridge (similar to MeetMe/ConfBridge) is used to direct the audio from the announcer to the many dialed channels.
- Parking (Asterisk 12+) - a special holding bridge is used for Parking, which entertains the waiting channel with hold music.

Asterisk 12+: Bridging Changed
In Asterisk 12, the bridging framework that ConfBridge was built on top of was extended to all bridges that Asterisk creates (with the exception of MeetMe). There are some new capabilities that this afforded Asterisk users; where applicable, this page will call out features that only apply to Asterisk 12 and later versions.

Destruction

Channels typically leave a bridge when the application that created the bridge is terminated (such as a conference leader ending a ConfBridge conference) or when the other side hangs up (such as in a two-party bridge created by Dial). When channels leave a bridge they can continue doing what they were doing prior to entering the bridge, continue executing dialplan, or be hung up.

Types

There are many types of bridges in Asterisk, each of which determine how the media is mixed between the participants of the bridge. In general, there are two categories of bridge types within Asterisk: two party and multiparty. Two party bridge variants include core bridges, local native bridges, and remote native bridges. Multiparty bridge variants include mixing and holding.

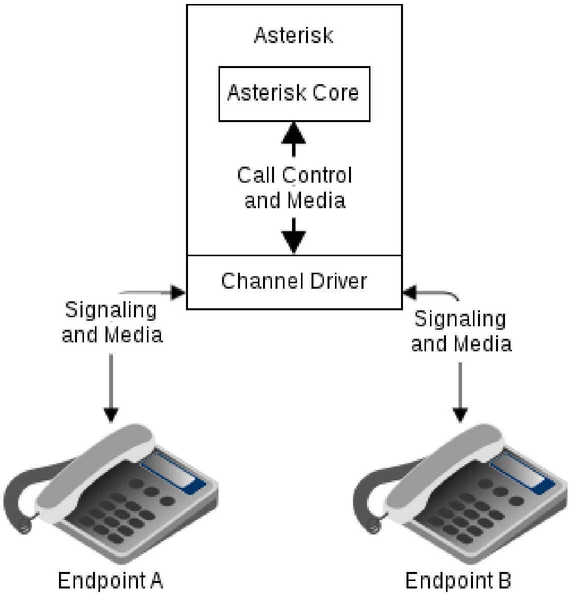
Asterisk 12+: Bridges are Smart
In Asterisk 12, the bridging framework is smart! It will automatically choose the best mixing technology available based on the channels in the bridge and - if needed - it will dynamically change the mixing type of the bridge based on conditions that occur. For example, a two-party core bridge may turn into a multiparty bridge if an attended transfer converges into a three-way bridge via the `atxferthreeway` DTMF option.

Two-Party

A two-party bridge shares media between two channels. Because there are only two participants in the bridge, certain optimizations can take place, depending on the type of channels in the bridge. As such, there are "sub-types" of two-party bridges that Asterisk can attempt to use to improve performance.

Core

A core bridge is the basic two-party bridge in Asterisk. Any channel of any type can communicate with any channel of any other type. A core bridge can perform media transcoding, media manipulation, call recording, DTMF feature execution, talk detection, and additional functionality because Asterisk has direct access to the media flowing between channels. Core bridges are the fallback when other types of bridging are not possible due to limiting network factors, configuration, or functionality requirements.



Native

A native bridge occurs when both participants in a two-party bridge have similar channel technologies. When this occurs, Asterisk defers the transfer of media to the channel drivers/protocol stacks themselves, and simply monitors for the channels leaving the bridge (either due to hangup, time-out, or some other condition). Since media is handled in the channel drivers/protocol stacks, no transcoding, media manipulation, recording, DTMF, or other features depending on media interpretation can be done by Asterisk. The primary advantage to native bridging is higher performance.

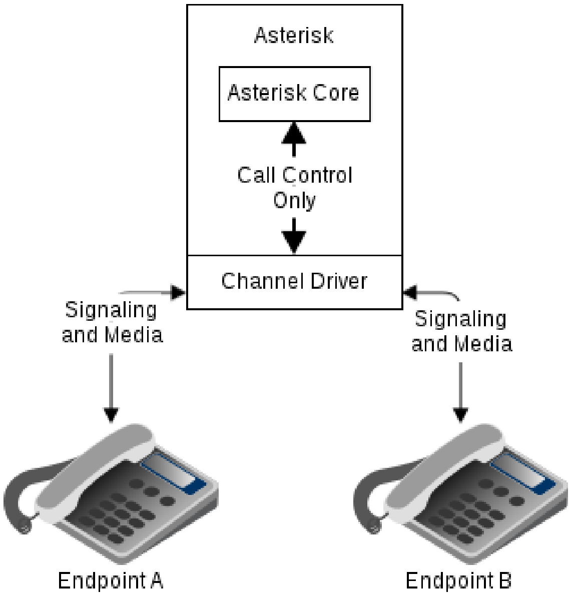
The following channel technologies support native bridging:

- RTP capable channel drivers (such as SIP channels)
- DAHDI channels
- IAX2 channels (Asterisk 11-)

> ✓ **Asterisk 12+ IAX2 Native Bridging is Gone**
> As it turned out, IAX2 native bridging was not much more efficient than a standard core bridge. In an IAX2 native bridge, the media must still be handled a good bit, i.e., placed into internal Asterisk frames. As such, when the bridging in Asterisk was converted to the new smart bridging framework, the IAX2 native bridge did not survive the transition.

Local

A local native bridge occurs when the media between two channels is handled by the channel drivers/protocol stacks themselves, but the media is still sent from each device to Asterisk. In this case, Asterisk is merely proxying the media back and forth between the two devices. Most types of native bridging in Asterisk are local.

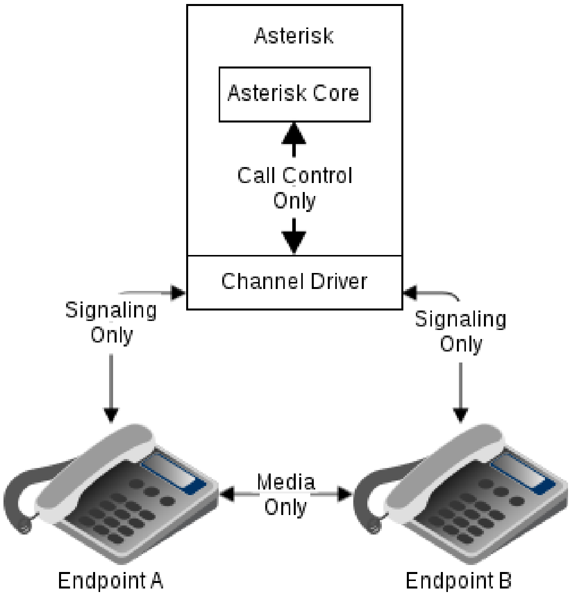


Remote

A remote native bridge occurs when the media between two channels is redirected by Asterisk to flow directly between the two devices the channels talk to. When this occurs, the media is completely outside of Asterisk. With SIP channels, this is often called "direct media". Not surprisingly, since the media is flowing outside of Asterisk, this bridge has the best performance in Asterisk. However, it can only be used in certain circumstances:

- Both channels in the native bridge must support direct media.
- The devices communicating with Asterisk cannot be behind a NAT (or otherwise obscured with a private IP address that the other device cannot resolve).

Only SIP channels support this type of native bridge.



Multiparty

Multiparty bridges interact with one or more channels and may route media among them. This can be thought of as an extension to two-party core bridging where media from multiple channels is merged or selected to be forwarded to the channels participating in the bridge. These bridges can have some, all, or none of the extended features of two-party core bridges depending on their intended use.

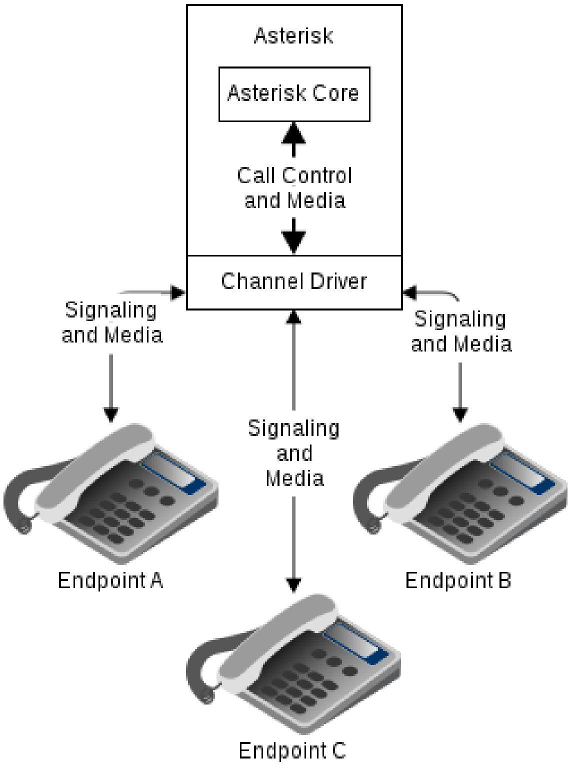
Mixing

There are several ways to access mixing multiparty bridges:

- MeetMe - This is a legacy conference bridge application and relies on DAHDI. This type of conference is limited to narrow band audio.
- ConfBridge (Asterisk 11+) - This is a conference bridge application based that supports wide band mixing.
- Ad-hoc Multiparty Bridges (Asterisk 12+) - Some DTMF features like 3-way attended transfers can create multiparty bridges as necessary.

Holding

Holding bridges are only available in Asterisk 12+ and provide a waiting area for channels which you may not yet be prepared to process or connect to other channels. This type of bridge prevents participants from exchanging media, can provide entertainment for all participants, and provides the ability for an announcer to interrupt entertainment with special messages as necessary. Entertainment for waiting channels can be MoH, silence, ringing, hold, etc.. Holding bridges can be accessed via BridgeWait or ARI.



Channels

Asterisk Channels

Almost nothing happens in Asterisk without a channel being involved. A channel is an entity inside Asterisk that acts as a channel of communication between Asterisk and another device. That is, a phone, a PBX, another Asterisk system, or even Asterisk itself (in the case of a local channel).

Our documentation and many Asterisk users speak about channels in terms of "calls". A **call** can be one or more channels creating a path of communication or activity through the Asterisk system.

To give you an idea about what channels do, here are a few facts about them:

- Channel Drivers provide channels in Asterisk.
- Channels can interface with each other through bridges.
- Applications and functions can affect the state or attributes of a channel or its media stream.
- Channels are commonly passing audio between communication endpoints, but can pass other data, such as video or text messages.
- Channels execute instructions with dialplan, but can be controlled by other APIs (AGI,AMI,ARI) or interfaces (CLI).

Common Asterisk Channels

One of the many benefits of Asterisk is the capability to interface with as many different technologies as you have channel drivers! However, most administrators will only make use of a few types at a time.

Here are a few commonly used channel types:

- A SIP channel driver such as chan_sip or chan_pjsip.
- DAHDI channels provided by chan_dahdi.
- Local channels provided by chan_local. (This was moved into the core in Asterisk 12)

SIP channels are used to interface with SIP capable VOIP devices, such as phones, channel banks, other PBXs or Internet Telephony Service Providers.

DAHDI channels are used to interface with DAHDI drivers and PRI libraries. In this case chan_dahdi allows you to use any DAHDI capable devices, such as Digium's line of T1/E1/J1 interface cards.

Local channels are used for dialing inward to the system itself, allowing any Asterisk component that can dial to call directly into dialplan. This provides a sort of "fake" call that still executes real instructions.

Configuring Channels

Text File Configuration

Most channel drivers have an associated configuration file. Some channels may require the configuration of dependent resources for optimal operation. For example, SIP channels, configured in sip.conf or pjsip.conf use RTP resources which can be configured in rtp.conf.

The Channel Drivers configuration section contains information on configuring documented channel drivers. In other cases the configuration file itself contains configuration documentation.

Database Configuration

Flat text configuration isn't the only option. A few channel drivers provide support for the ARA (Asterisk Realtime Architecture) and can therefore pull configuration from a local or remote database. Use of the ARA requires configuration of additional resources and dependencies outside the channel drivers themselves.

Using, Controlling and Routing Channels

Once you have a channel driver configured, how does it get used? When do channels get created?

Here are a few scenarios where a channel could get created:

- A device configured in the channel driver communicates to Asterisk (e.g. over a network) that it wants to make a call.
- A user executes a command (such as Originate) to create a new channel.
- An existing channel executes dialplan that calls an application (such as Dial) to create a new channel.
- Asterisk receives API calls that create a new channel or channels.

Once a channel is established, the events that occur are channel technology-dependent. That is, whether audio, video or other data communication begins over the channel will depend on signaling that occurs over SIP, ISDN, H.323 or other protocols implemented via the channel driver.

When Asterisk has established a channel, Asterisk will use a combination of channel driver configuration and dialplan instruction to determine how the channel behaves. On top of that Asterisk can communicate with external programs synchronously or asynchronously to receive API calls for channel inspection, direction or manipulation.

Once channels are established and communicating between devices and Asterisk; where that data flows to depends on the channel type itself, its overall configuration, device specific configuration, signaling sent by the originating mechanism (a device, a command, an API call) and associated bridges. One or more channels could be talking to one or more channels over various bridges. What specifically Asterisk talks to on each channel is limited only by the technology implemented by the channel driver.

Inbound and Outbound Channels

Often in our documentation, troubleshooting and development discussions you'll see mention of inbound or outbound channels. It'll be helpful to define what that means here.

Inbound channels are channels created when things outside of Asterisk call **into** Asterisk. This is typically the channel executing Dialplan.

Outbound channels are channels created when Asterisk is calling **out** to something outside Asterisk.

The primary exception is with Local Channels. In the case of local channels, you'll typically have two local channel legs, one that is treated as outbound and the other as inbound. In this case both are really inside Asterisk, but one is executing dialplan and the other is not. The leg executing dialplan is the one treated as inbound.

Below we'll diagram a few examples for clarity.

Fig 1

Two channels bridged from a typical Dial

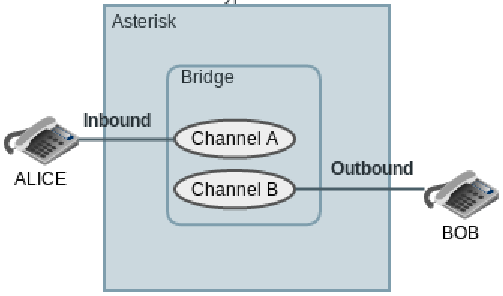


Fig 2

Originate Channel to Application

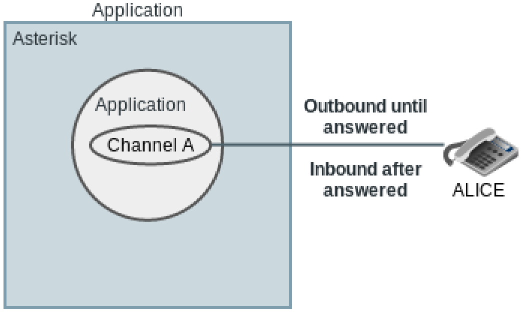
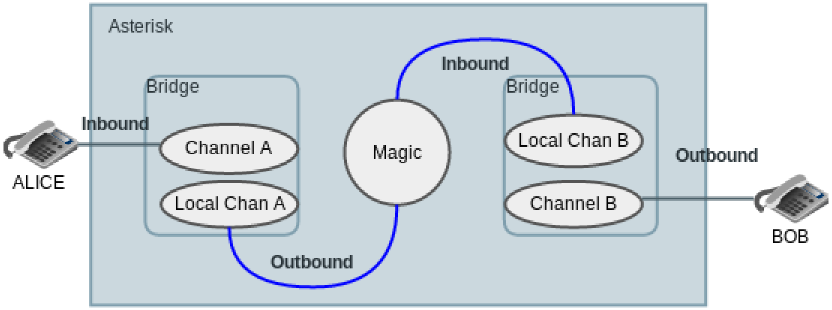


Fig 3

SIP to SIP via non-optimized Local Channels



The figures have been kept somewhat generic and should apply to most channel types. Each figure shows the final state of the call, rather than a sequence of events.

Below are explanations of the various figures.

Fig 1

One phone dials another phone; about as simple as it gets.

The **inbound** channel is created from Alice's phone calling Asterisk. Asterisk then calls the extension dialed by Alice by creating an **outbound** channel to talk to Bob. Once the call is established the two channels are put into a bridgeBridges.

Fig 2

A user runs the originate command from AMI, or maybe something like "channel originate SIP/Alice application playback demo-congrats" from the CLI.

Asterisk creates an **outbound** channel to call the device specified (SIP/Alice). When answered, Asterisk begins treating the channel like an **inbound** channel and connects it to the specified dialplan application.

Fig 3

Perhaps a user runs originate again - but this time "channel originate SIP/Alice extension dialbob@internal" from the CLI. Where dialbob@internal contains dialplan telling Asterisk to dial outbound to SIP/Bob. At first, the created **outbound** channel would look like Fig 2 where it begins to be treated as **inbound** after the device answers the call. At that point, a number of things happen:

- Asterisk creates an outbound local channel into Asterisk and bridges it with the now inbound channel to Alice.
- Asterisk creates another leg of local channel as "inbound" into Asterisk to execute the dialplan at the extension specified with the originate. This local channel is essentially bridged with some magic to the other local channel.
- In our case the dialplan executes something like Dial(SIP/Bob), so the new SIP channel is created to communicate with SIP/Bob and is then bridged with the inbound local channel. Now communication flows across the whole path.

For this example demonstrating relationships between channels and other elements we used non-optimized local channels. If the local channels are optimized, then they will optimize themselves out of this mix and Alice and Bob's channels would be bridged together directly.

Channel Variable Inheritance

When working with channels you'll almost certainly be touching channel variables. It is useful to note that upon setting a channel variable the level of inheritance between channels can be defined. This is discussed in the Channel Variables sub-section Variable Inheritance.

Frames

Under Construction

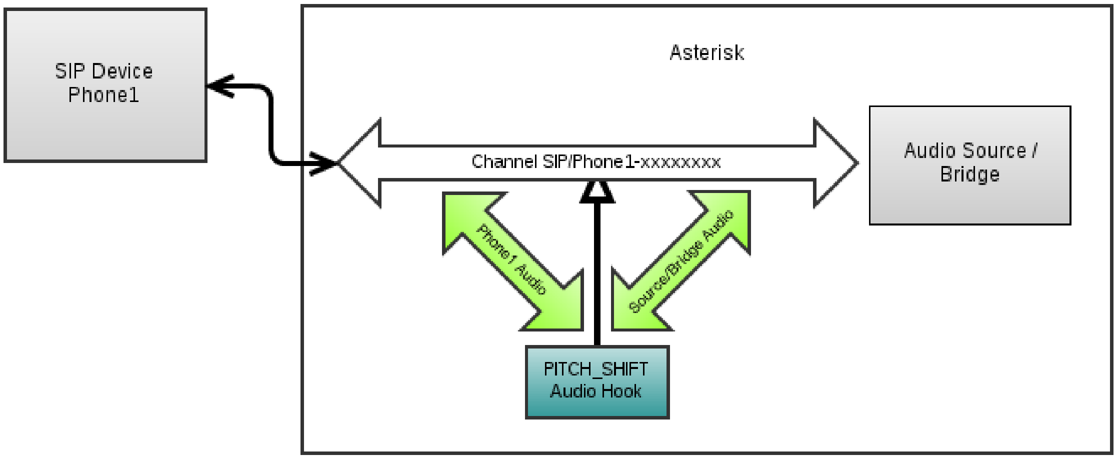
Top-level page for talking about frames, frame-hooks/audio-hooks.

Audiohooks

Overview

Certain applications and functions are capable of attaching what is known as an audiohook to a channel. In order to understand what this means and how to handle these applications and functions, it is useful to understand a little of the architecture involved with attaching them.

Introduction - A Simple Audiohook



In this simple example, a SIP phone has dialed into Asterisk and its channel has invoked a function (pitch_shift) which has been set to cause all audio sent and received to have its pitch shifted higher (i.e. if the audio is voice, the voices will sound squeaky sort of like obnoxious cartoon chipmunks). The following dialplan provides a more concrete usage:

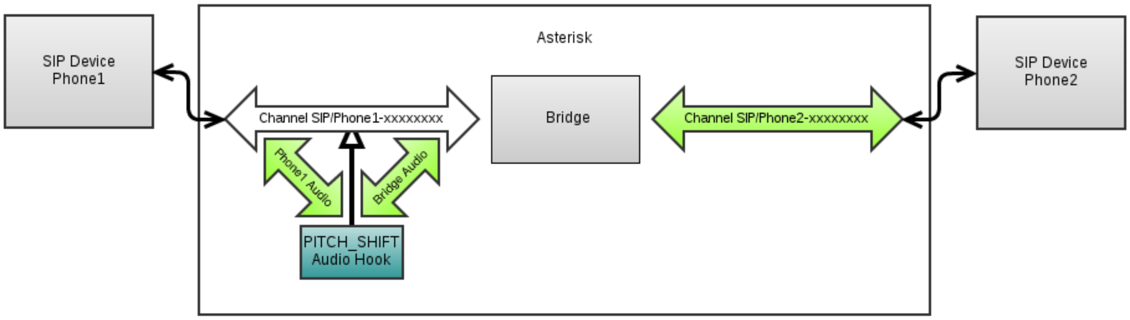
```
exten => 1,1,Answer()
exten => 1,n,Set(PITCH_SHIFT(both)=higher)
exten => 1,n,Voicemail(501)
```

When a phone calls this extension, it will be greeted by a higher pitched version of the voicemail prompt and then the speaker will leave a message for 501. The sound going from the phone to voicemail will also be higher pitched than what was actually said by the person who left the message.

Right now a serious minded Asterisk user reading this example might think something along the lines of 'So what, I don't have any use for making people using my phone system sound like squirrels." However, audiohooks provide a great deal of the functionality for other applications within Asterisk including some features that are very business minded (listening in on channels, recording phone calls, and even less spy-guy type things like adjusting volume on the fly)

It's important to note that audiohooks are bound to the channel that they were invoked on. They don't apply to a call (a call is actually a somewhat nebulous concept in general anyway) and so one shouldn't expect audiohooks to follow other channels around just because audio that those channels are involved with touches the hook. If the channel that created the audiohook ceases to be involved with an audio stream, the audiohook will also no longer be involved with that audio stream.

Attended Transfers and AUDIOHOOK_INHERIT



```
exten => 1,1,Answer()
exten => 1,n,MixMonitor(training_recording.wav)
exten => 1,n,Queue(techsupport)
```

Imagine the following scenario. An outside line calls into an Asterisk system to enter a tech support queue. When the call starts this user hears something along the lines of "Thank you for calling, all calls will be recorded for training purposes", so naturally MixMonitor will be used to record the call. The first available agent answers the call and can't quite seem to provide a working solution to the customer's problem, so he attempts to perform an attended transfer to someone with more expertise on the issue. The user gets transfered, and the rest of the call goes smoothly, but... ah nuts. The recording stopped for some reason when the agent transferred the customer to the other user. And why didn't this happen when he blind transferred a customer the other day?

The reason MixMonitor stopped is because the channel that owned it died. An Asterisk admin might think something like "That's not true, the mixmonitor was put on the customer channel and its still there, I can still see it's name is the same and everything." and it's true that it seems that way, but attended transfers in particular cause what's known as a channel masquerade. Yes, its name and everything else about it seems like the same channel, but in reality the customer's channel has been swapped for the agent's channel and died since the agent hung up. The audiohook went with it. Under normal circumstances, administrators don't need to think about masquerades at all, but this is one of the rare instances where it gets in the way of desired behavior. This doesn't affect blind transfers because they don't start the new dialog by having the person who initiated the transfer bridging to the end recipient.

Working around this problem is pretty easy though. Audiohooks are not swapped by default when a masquerade occurs, unlike most of the relevant data on the channel. This can be changed on a case by case basis though with the AUDIOHOOK_INHERIT dialplan function.

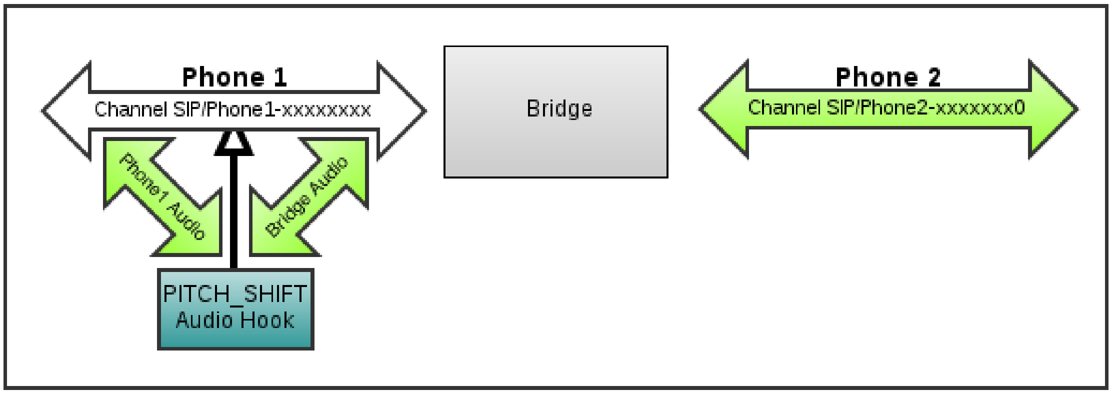
Using AUDIOHOOK_INHERT only requires that AUDIOHOOK_INHERIT(source)=yes is set where source is the name given for the source of the audiohook. For more information on the sources available, see the description of the source argument in the documentation for AUDIOHOOK_INHERIT.

So to fix the above example so that mixmonitor continues to record after the attended transfer, only one extra line is needed.

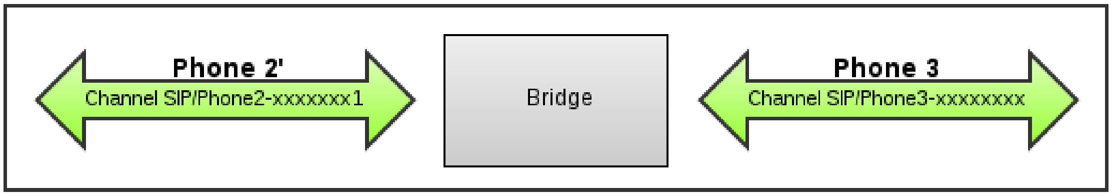
```
exten => 1,1,Answer()
exten => 1,n,MixMonitor(training_recording.wav)
exten => 1,n,Set(AUDIOHOOK_INHERIT(MixMonitor)=yes)
exten => 1,n,Queue(techsupport)
```

Below is an illustrated example of how the masquerade process impacts an audiohook (in the case of the example, PITCH_SHIFT)

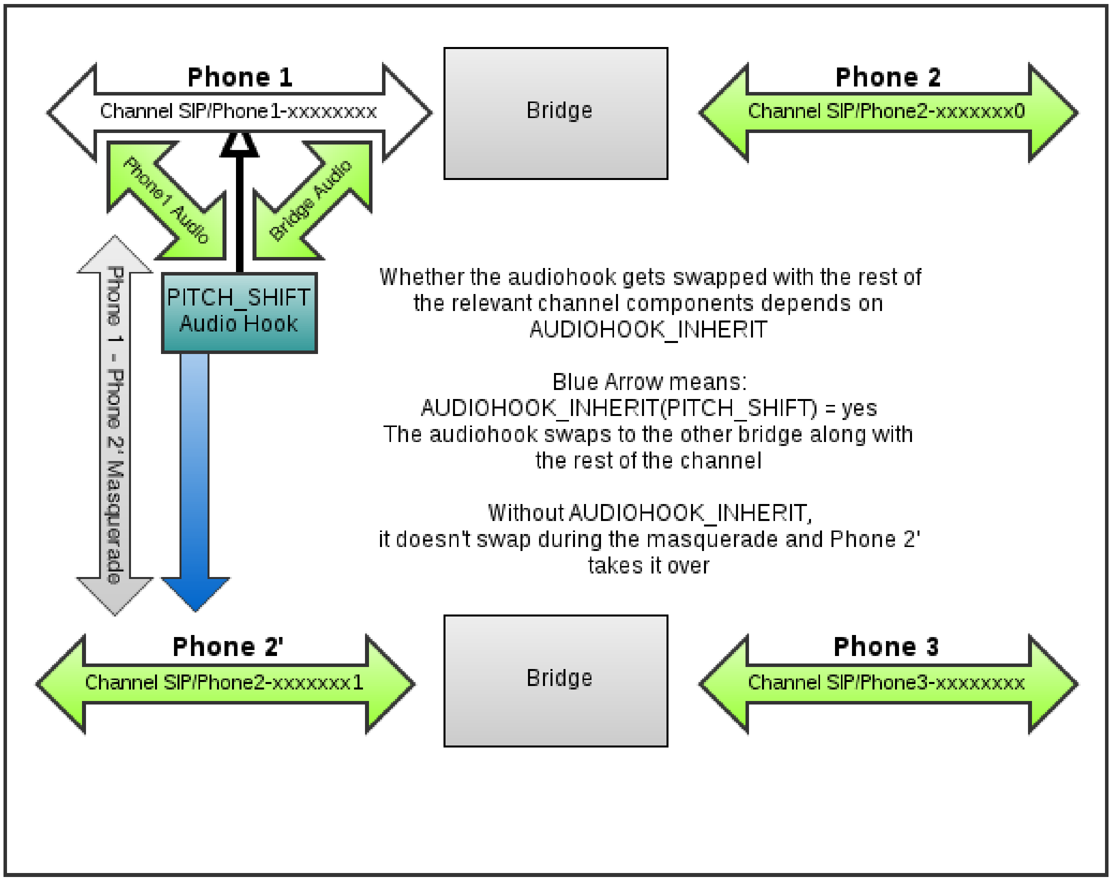
Initial Call Setup



SIP/Phone2 attempts starts to attended transfer SIP/Phone1 to SIP/Phone3



Phone 2 hangs up on Phone 3, initiating the transfer. This requires a masquerade.

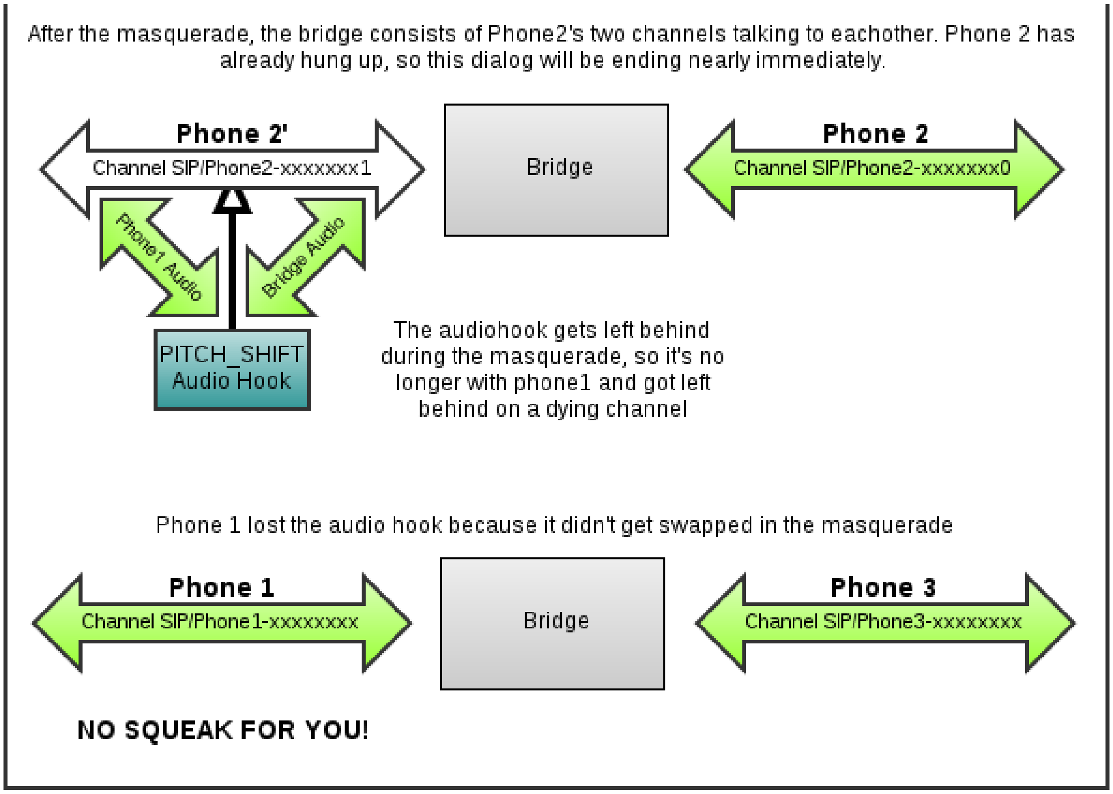

Phone 1
Channel SIP/Phone1-xxxxxxxx
Phone1 Audio
Bridge Audio
PITCH_SHIFT
Audio Hook
Phone 1 - Phone 2' Masquerade
Bridge
Phone 2
Channel SIP/Phone2-xxxxxxx0
Phone 2'
Channel SIP/Phone2-xxxxxxx1
Bridge
Phone 3
Channel SIP/Phone3-xxxxxxxx

Whether the audiohook gets swapped with the rest of the relevant channel components depends on AUDIOHOOK_INHERIT

Blue Arrow means:
AUDIOHOOK_INHERIT(PITCH_SHIFT) = yes
The audiohook swaps to the other bridge along with the rest of the channel

Without AUDIOHOOK_INHERIT,
it doesn't swap during the masquerade and Phone 2' takes it over

Bridges after Transfer: Without AUDIOHOOK_INHERIT(PITCH_SHIFT)



After the masquerade, the bridge consists of Phone2's two channels talking to eachother. Phone 2 has already hung up, so this dialog will be ending nearly immediately.

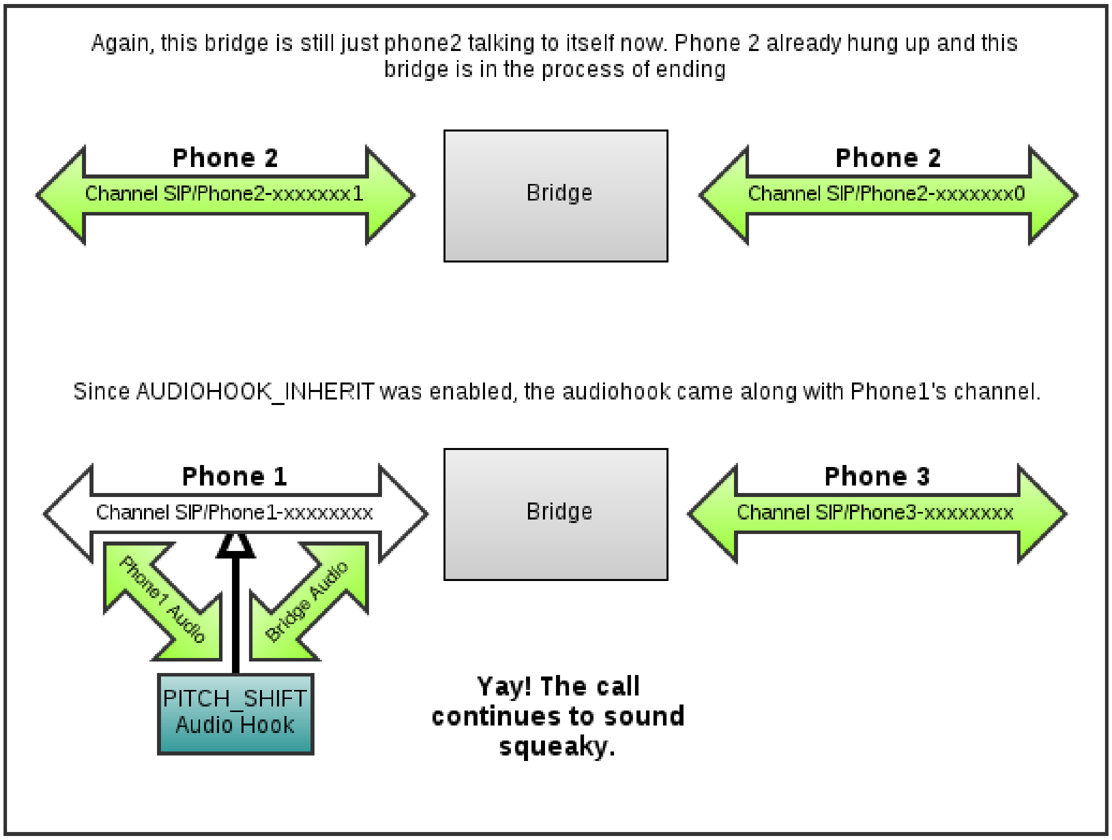
The audiohook gets left behind during the masquerade, so it's no longer with phone1 and got left behind on a dying channel

Phone 1 lost the audio hook because it didn't get swapped in the masquerade

NO SQUEAK FOR YOU!

Phone 2'
Channel SIP/Phone2-xxxxxxx1
Phone1 Audio
Bridge Audio
PITCH_SHIFT Audio Hook
Bridge
Phone 2
Channel SIP/Phone2-xxxxxxx0
Phone 1
Channel SIP/Phone1-xxxxxxxx
Bridge
Phone 3
Channel SIP/Phone3-xxxxxxxx

Bridges after Transfer: With AUDIOHOOK_INHERIT(PITCH_SHIFT)



Again, this bridge is still just phone2 talking to itself now. Phone 2 already hung up and this bridge is in the process of ending

Since AUDIOHOOK_INHERIT was enabled, the audiohook came along with Phone1's channel.

Yay! The call continues to sound squeaky.

Phone 2
Channel SIP/Phone2-xxxxxxx1
Bridge
Phone 2
Channel SIP/Phone2-xxxxxxx0
Phone 1
Channel SIP/Phone1-xxxxxxxx
Phone1 Audio
Bridge Audio
PITCH_SHIFT Audio Hook
Bridge
Phone 3
Channel SIP/Phone3-xxxxxxxx

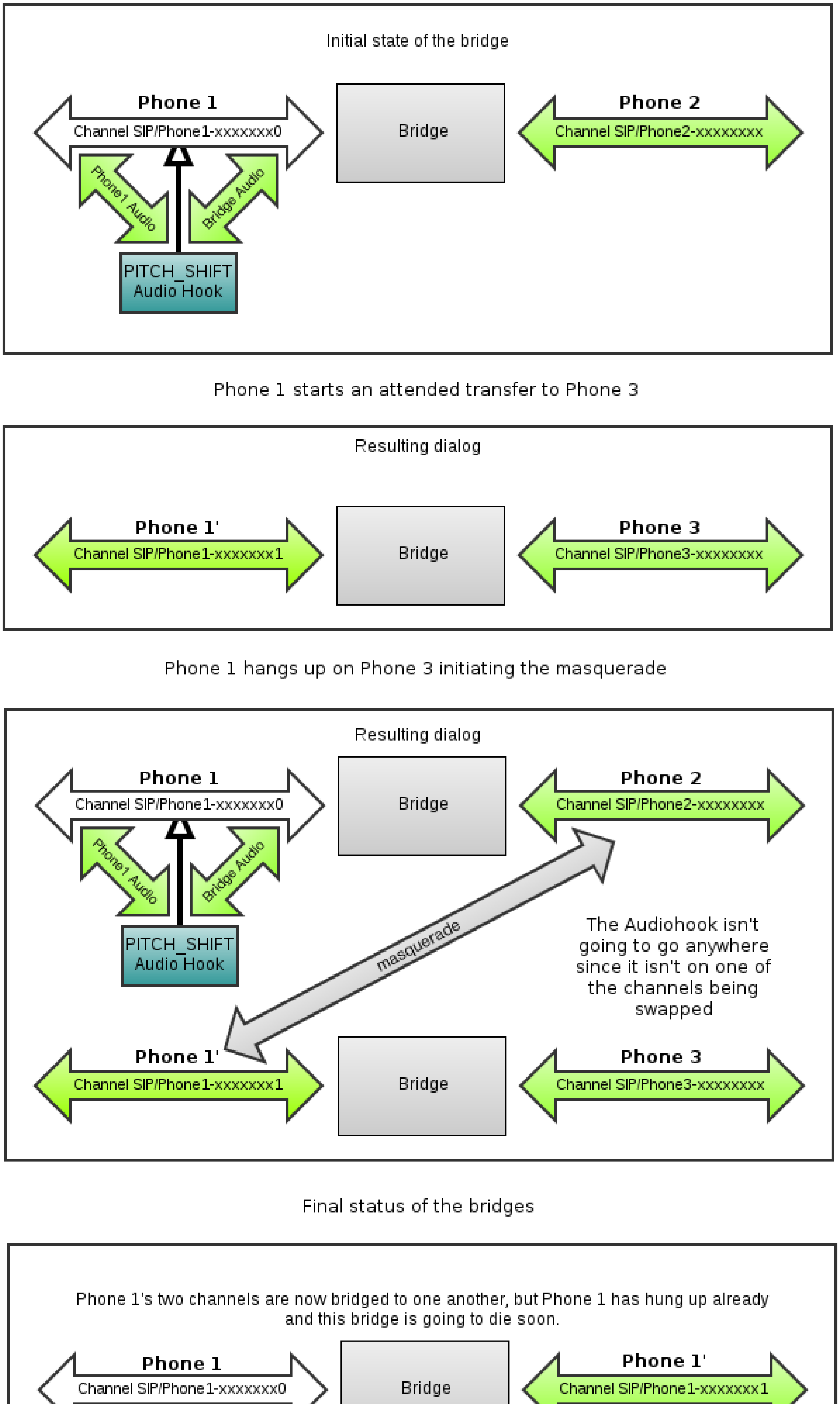
Inheritance of audiohooks can be turned off in the same way by setting AUDIOHOOK_INHERIT(source)=no.

Audiohook Sources

Audiohooks have a source name and can come from a number of sources. An up to date list of possible sources should always be available from the documentation for AUDIOHOOK_INHERIT.

Limitations for transferring Audiohooks

Even with audiohook inheritance set, the MixMonitor is still bound to the channel that invoked it. The only difference in this case is that with this option set, the audiohook won't be left on the discarded channel through the masquerade. This option doesn't enable a channel running mixmonitor to transfer the MixMonitor to another channel or anything like that. The dialog below illustrates why.


Initial state of the bridge
Phone 1
Channel SIP/Phone1-xxxxxxx0
Phone1 Audio
Bridge Audio
PITCH_SHIFT
Audio Hook
Bridge
Phone 2
Channel SIP/Phone2-xxxxxxxx

Phone 1 starts an attended transfer to Phone 3

Resulting dialog
Phone 1'
Channel SIP/Phone1-xxxxxxx1
Bridge
Phone 3
Channel SIP/Phone3-xxxxxxxx

Phone 1 hangs up on Phone 3 initiating the masquerade

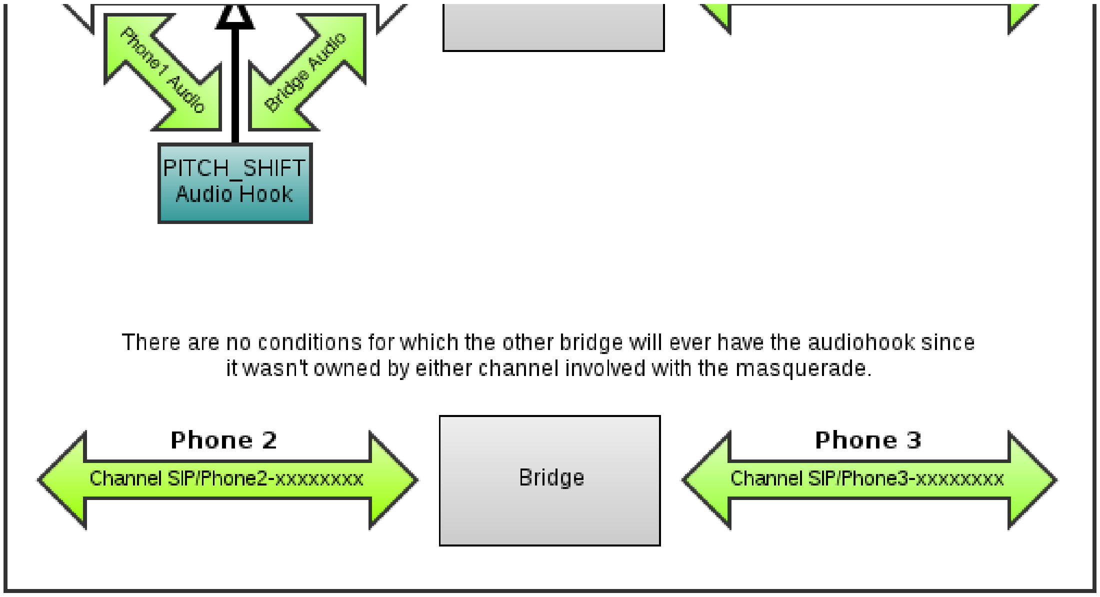
Resulting dialog
Phone 1
Channel SIP/Phone1-xxxxxxx0
Phone1 Audio
Bridge Audio
PITCH_SHIFT
Audio Hook
Bridge
Phone 2
Channel SIP/Phone2-xxxxxxxx
masquerade
Phone 1'
Channel SIP/Phone1-xxxxxxx1
Bridge
Phone 3
Channel SIP/Phone3-xxxxxxxx

The Audiohook isn't going to go anywhere since it isn't on one of the channels being swapped

Final status of the bridges

Phone 1's two channels are now bridged to one another, but Phone 1 has hung up already and this bridge is going to die soon.

Phone 1
Channel SIP/Phone1-xxxxxxx0
Bridge
Phone 1'
Channel SIP/Phone1-xxxxxxx1



There are no conditions for which the other bridge will ever have the audiohook since it wasn't owned by either channel involved with the masquerade.

States and Presence

Asterisk includes the concepts of Device State , Extension State and Presence State which together allow Asterisk applications and interfaces to receive information about the state of devices, extensions and the users of the devices.

As an example, channel drivers like chan_sip or res_pjsip/chan_pjsip may both provide devices with device state, plus allow devices to subscribe to hints to receive notifications of state change. Other examples would be app_queue which takes into consideration the device state of queue members to influence queue logic or the Asterisk Manager Interface which provides actions for querying extension state and presence state.

In This Section

See Also

Distributed Device State

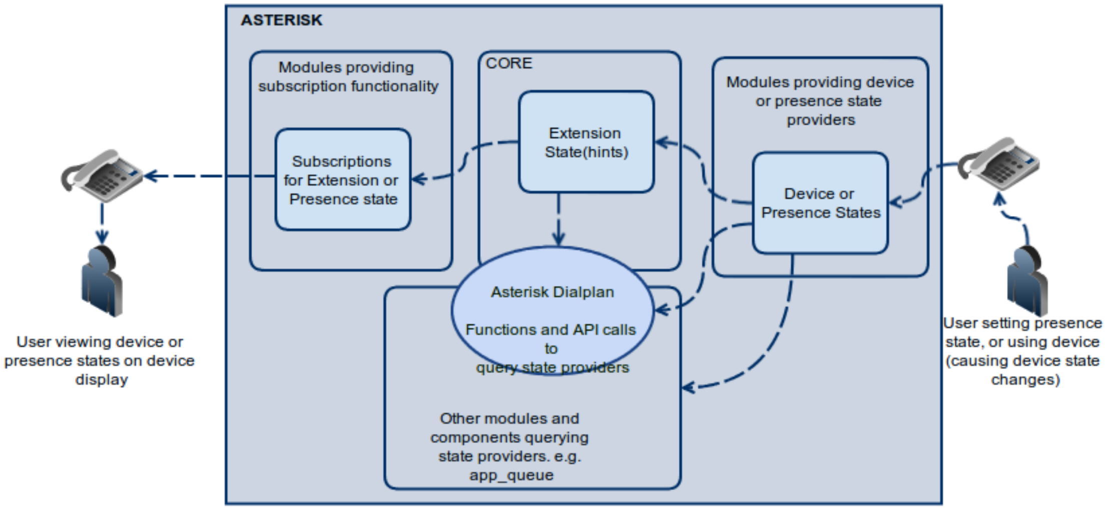
Publishing Extension State

Exchanging Device and Mailbox State Using PJSIP

Additionally, modules exist for Corosync and XMPP PubSub support to allow device state to be shared and distributed across multiple systems.

The sub-sections here describe these concepts, point to related module specific configuration sections and discuss Querying and Manipulating State information.

The figure below may help you get an idea of the overall use of states and presence with the Asterisk system. It has been simplified to focus on the flow and usage of state and presence. In reality, the architecture can be a bit more confusing. For example a module could both provide subscription functionality for a subscriber and be the same module providing the devices and device state on the other end.



ASTERISK
Modules providing subscription functionality
Subscriptions for Extension or Presence state
CORE
Extension State(hints)
Modules providing device or presence state providers
Device or Presence States
Asterisk Dialplan
Functions and API calls to query state providers
Other modules and components querying state providers. e.g. app_queue
User viewing device or presence states on device display
User setting presence state, or using device (causing device state changes)

Device State

Devices

Devices are discrete components of functionality within Asterisk that serve a particular task. A device may be a channel technology resource, such as SIP/<name> in the case of chan_sip.so or a feature resource of another module such as app_confbridge.so which provides devices like confbridge:<name>.

State information

Asterisk devices make state information available to the Asterisk user, such that a user might make use of the information to affect call flow or behavior of the Asterisk system. The device state identifier for a particular device is typically very similar to the device name. For example the device state identifier for SIP/6001 would be SIP/6001, for confbridge 7777 it would be confbridge:7777. Device states have a one-to-one mapping to the device they represent. That is opposed to other state providers in Asterisk which may have one-to-many relationships, such as Extension State.

The Querying and Manipulating State section covers how to access or manipulate device state as well as other states.

Common Device State Providers

Device state providers are the components of Asterisk that provide some state information for their resources. The device state providers available in Asterisk will depend on the version of Asterisk you are using, what modules you have installed and how those modules are configured. Here is a list of the common *device state identifiers* you will see and what Asterisk component provides the resources and state.

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Device State Identifier	Device State Provider
PJSIP/<resource>	PJSIP SIP stack, res_pjsip.so, chan_pjsip.so.
SIP/<resource>	The older SIP channel driver, chan_sip.so.
DAHDI/<resource>	The popular telephony hardware interface driver, chan_dahdi.so.
IAX2/<resource>	Inter-Asterisk Exchange protocol! chan_iax2.so.
ConfBridge:<resource>	The conference bridge application, app_confbridge.so.
MeetMe:<resource>	The older conference bridging app, app_meetme.so.
Park:<resource>	The Asterisk core in versions up to 11. res_parking.so in versions 12 or greater.
Calendar:<resource>	res_calendar.so and related calendaring modules.
Custom:<resource>	Custom device state provided by the asterisk core.

Note that we are not differentiating any device state providers based on what is on the far end. Depending on device state provider, the far end of signaling for state could be a physical device, or just a discrete feature resource inside of Asterisk. In terms of understanding device state for use in Asterisk, it doesn't really matter. The device state represents the state of the Asterisk device as long as it is able to provide it regardless of what is on the far end of the communication path.

Custom Device States

The Asterisk core provides a *Custom* device state provider (custom:<resource>) that allows you to define arbitrary device state resources. See the Querying and Manipulating State section for more on using custom device state.

Possible Device States

Here are the possible states that a device state may have.

- UNKNOWN
- NOT_INUSE
- INUSE
- BUSY
- INVALID

- UNAVAILABLE
- RINGING
- RINGINUSE
- ONHOLD

Though the label for each state carries a certain connotation, the actual meaning of each state is really up to the device state provider. That is, any particular state may mean something different across device state providers.

Module Specific Device State

There is module specific configuration that you must be aware of to get optimal behavior with certain state providers.

For **chan_sip** see the chan_sip State and Presence Options section.

For **res_pjsip** see the Configuring res_pjsip for Presence Subscriptions section.

Extension State and Hints

Overview

Extension state is the state of an Asterisk extension, as opposed to the direct state of a device or a user. It is the aggregate of Device state from devices mapped to the extension through a **hint** directive. See the States and Presence section for a diagram showing the relationship of all the various states.

Asterisk's SIP channel drivers provide facilities to allow SIP presence subscriptions (RFC3856) to extensions with a defined hint. With an active subscription, devices can receive notification of state changes for the subscribed to extension. That notification will take the form of a SIP NOTIFY with PIDF content (RFC3863) containing the presence/state information.

Defining Hints

For Asterisk to store and provide state for an extension, you must first define a **hint** for that extension. Hints are defined in the Asterisk dialplan, i.e. extensions.conf.

When Asterisk loads the configuration file it will create hints in memory for each hint defined in the dialplan. Those hints can then be queried or manipulated by functions and CLI commands. The state of each hint will regularly be updated based on state changes for any devices mapped to a hint.

The full syntax for a hint is

```
exten = <extension>,hint,<device state id>[& <more dev state id],<presence state id>
```

Here is what you might see for a few configured hints.

```
[internal]

exten = 6001,hint,SIP/Alice&SIP/Alice-mobile
exten = 6002,hint,SIP/Bob
exten = 6003,hint,SIP/Charlie&DAHDI/3
exten = 6004,hint,SIP/Diane,CustomPresence:Diane
exten = 6005,hint,,CustomPresence:Ellen
```

Things of note:

- You may notice that the syntax for a hint is similar to a regular extension, except you use the **hint** keyword in place of the priority. Remember these special hint directives are used at load-time and not during run-time, so there is no need for a priority.
- Multiple devices can be mapped to an extension by providing an ampersand delimited list.
- A presence state ID is set after the device state IDs. If set with only a presence state provider you must be sure to include a blank field after the hint as in the example for extension 6005.
- Hints can be anywhere in the dialplan. Though, remember that dialplan referencing the extension and devices subscribing to it will need use the extension number/name and context. The hints shown above would be 6001@internal, 6002@internal, etc, just like regular extensions.

Querying Extension State

The Querying and Manipulating State section covers accessing and affecting the various types of state.

For a quick CLI example, once you have defined some hints, you can easily check from the CLI to verify they get loaded correctly.

```
*CLI> core show hints
    -= Registered Asterisk Dial Plan Hints =-
                   6003@internal            : SIP/Charlie&DAHDI/3   State:Unavailable     Watchers  0
                   6002@internal            : SIP/Bob               State:Unavailable     Watchers  0
                   6001@internal            : SIP/Alice&SIP/Alice-  State:Unavailable     Watchers  0
                   6005@internal            : ,CustomPresence:Elle  State:Unavailable     Watchers  0
                   6004@internal            : SIP/Diane,CustomPres  State:Unavailable     Watchers  0
----------------
- 5 hints registered
```

In this example I was lazy, so they don't have real providers mapped otherwise you would see various states represented.

SIP Subscription to Asterisk hints

Once a hint is configured, Asterisk's SIP drivers can be configured to allow SIP User Agents to subscribe to the hints. A subscription will result in state change notifications being sent to the subscriber.

Configuration for **chan_sip** is discussed in Configuring chan_sip for Presence Subscriptions

Configuration for **res_pjsip** is discussed in Configuring res_pjsip for Presence Subscriptions

Presence State

Overview

Asterisk 11 has been outfitted with support for presence states. An easy way to understand this is to compare presence state support to the device state support Asterisk has always had. Like with device state support, Asterisk has a core API so that modules can register themselves as presence state providers, alert others to changes in presence state, and query the presence state of others. The difference between the device and presence state concepts is made clear by understanding the subject of state for each concept.

- Device state reflects the current **state of a physical device** connected to Asterisk
- Presence state reflects the current **state of the user** of the device

For example, a **device** may currently be `not in use` but the **person** is `away`. This can be a critical detail when determining the availability of the **person**.

While the architectures of presence state and device state support in Asterisk are similar, there are some key differences between the two.

- Asterisk cannot infer presence state changes the same way it can device state changes. For instance, when a SIP endpoint is on a call, Asterisk can infer that the device is being used and report the device state as `in use`. Asterisk cannot infer whether a user of such a device does not wish to be disturbed or would rather chat, though. Thus, all presence state changes have to be manually enacted.
- Asterisk does not take presence into consideration when determining availability of a device. For instance, members of a queue whose device state is `busy` will not be called; however, if that member's device is `not in use` but his presence is `away` then Asterisk will still attempt to call the queue member.
- Asterisk cannot aggregate multiple presence states into a single combined state. Multiple device states can be listed in an extension's hint priority to have a combined state reported. Presence state support in Asterisk lacks this concept.

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Presence States

- `not_set`: No presence state has been set for this entity.
- `unavailable`: This entity is present but currently not available for communications.
- `available`: This entity is available for communication.
- `away`: This entity is not present and is unable to communicate.
- `xa`: This entity is not present and is not expected to return for a while.
- `chat`: This entity is available to communicate but would rather use instant messaging than speak.
- `dnd`: This entity does not wish to be disturbed.

Subtype and Message

In addition to the basic presence states provided, presence also has the concept of a **subtype** and a **message**.

The subtype is a brief method of describing the nature of the state. For instance, a subtype for the `away` status might be "at home".

The message is a longer explanation of the current presence state. Using the same `away` example from before, the message may be "Sick with the flu. Out until the 18th".

`func_presencestate` And The `CustomPresence` Provider

The only provider of presence state in Asterisk 11 is the `CustomPresence` provider. This provider is supplied by the `func_presencestate.so` module, which grants access to the `PRESENCE_STATE` dialplan function. The documentation for `PRESENCE_STATE` can be found here. `CustomPresence` is device-agnostic within the core and can be a handy way to set and query presence from dialplan, or APIs such as the AMI.

A simple use case for `CustomPresence` in dialplan is demonstrated below.

```
[default]
exten => 2000,1,Answer()
same => n,Set(CURRENT_PRESENCE=${PRESENCE_STATE(CustomPresence:Bob,value)})
same => n,GotoIf($[${CURRENT_PRESENCE}=available]?set_unavailable:set_available)
same => n(set_available),Set(PRESENCE_STATE(CustomPresence:Bob)=available,,)
same => n,Goto(finished)
same => n(set_unavailable),Set(PRESENCE_STATE(CustomPresence:Bob)=unavailable,,)
same => n(finished),Playback(queue-thankyou)
same => n,Hangup

exten => 2001,1,GotoIf($[${PRESENCE_STATE(CustomPresence:Bob,value)}!=available]?voicemail)
same => n,Dial(SIP/Bob)
same => n(voicemail)VoiceMail(Bob@default)
```

With this dialplan, a user can dial `2000@default` to toggle Bob's presence between `available` and `unavailable`. When a user attempts to call Bob using `2001@default`, if Bob's presence is currently not `available` then the call will go directly to voicemail.

> One thing to keep in mind with the `PRESENCE_STATE` dialplan function is that, like with `DEVICE_STATE`, state may be queried from any presence provider, but `PRESENCE_STATE` is only capable of setting presence state for the `CustomPresence` presence state provider.

Configuring Presence Subscription with Hints

As is mentioned in the phone support section, at the time of writing this will only work with a Digium phone.

Like with device state, presence state is associated to a dialplan extension with a hint. Presence state hints come after device state in the hint extension and are separated by a comma (`,`). As an example:

```
[default]
exten => 2000,hint,SIP/2000,CustomPresence:2000
exten => 2000,1,Dial(SIP/2000)
same => n,Hangup()
```

Or alternatively, you could define the presence state provider without a device.

```
exten => 2000,hint,,CustomPresence:2000
```

The **first** example would allow for someone subscribing to the extension state of `2000@default` to be notified of device state changes for device `SIP/2000` as well as presence state changes for the presence provider `CustomPresence:2000`.

The **second** example would allow for the subscriber to receive notification of state changes for only the presence provider CustomPresence:2000.

The `CustomPresence` presence state provider will be discussed further on this page.

Also like with device state, there is an Asterisk Manager Interface command for querying presence state. Documentation for the AMI `PresenceState` command can be found here.

Example Presence Notification

When a SIP device is subscribed to a hint you have configured in Asterisk and that hint references a presence state provider, then upon change of that state Asterisk will generate a notification. That notification will take the form of a SIP NOTIFY including XML content. In the expanding panel below I've included an example of a presence notification sent to a Digium phone. This particular presence notification happened when we changed presence state for CustomPresence:6002 via the CLI command 'presencestate change'.
Click here to see the NOTIFY example

```
myserver*CLI> presencestate change CustomPresence:6002 UNAVAILABLE
Changing 6002 to UNAVAILABLE
set_destination: Parsing <sip:6002@10.24.18.138:5060;ob> for address/port to send to
set_destination: set destination to 10.24.18.138:5060
Reliably Transmitting (no NAT) to 10.24.18.138:5060:
NOTIFY sip:6002@10.24.18.138:5060;ob SIP/2.0
Via: SIP/2.0/UDP 10.24.18.124:5060;branch=z9hG4bK68008251;rport
Max-Forwards: 70
From: sip:6002@10.24.18.124;tag=as722c69ec
To: "Bob" <sip:6002@10.24.18.124>;tag=4DpRZfRIlaKU9iQcaME2APx85TgFOEN7
Contact: <sip:6002@10.24.18.124:5060>
Call-ID: JVoQfeZe1cWTdPI5aTWkRpdqkjs8zmME
CSeq: 104 NOTIFY
User-Agent: Asterisk PBX SVN-branch-12-r413487
Subscription-State: active
Event: presence
Content-Type: application/pidf+xml
Content-Length: 602

<?xml version="1.0" encoding="ISO-8859-1"?>
<presence xmlns="urn:ietf:params:xml:ns:pidf"
xmlns:pp="urn:ietf:params:xml:ns:pidf:person"
xmlns:es="urn:ietf:params:xml:ns:pidf:rpid:status:rpid-status"
xmlns:ep="urn:ietf:params:xml:ns:pidf:rpid:rpid-person"
entity="sip:6002@10.24.18.124">
<pp:person><status>
</status></pp:person>
<note>Ready</note>
<tuple id="6002">
<contact priority="1">sip:6002@10.24.18.124</contact>
<status><basic>open</basic></status>
</tuple>
<tuple id="digium-presence">
<status>
<digium_presence type="unavailable" subtype=""></digium_presence>
</status>
</tuple>
</presence>

---
  == Extension Changed 6002[from-internal] new state Idle for Notify User 6002

<--- SIP read from UDP:10.24.18.138:5060 --->
SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.24.18.124:5060;rport=5060;received=10.24.18.124;branch=z9hG4bK68008251
Call-ID: JVoQfeZe1cWTdPI5aTWkRpdqkjs8zmME
From: <sip:6002@10.24.18.124>;tag=as722c69ec
To: "Bob" <sip:6002@10.24.18.124>;tag=4DpRZfRIlaKU9iQcaME2APx85TgFOEN7
CSeq: 104 NOTIFY
Contact: "Bob" <sip:6002@10.24.18.138:5060;ob>
Allow: PRACK, INVITE, ACK, BYE, CANCEL, UPDATE, SUBSCRIBE, NOTIFY, REFER, MESSAGE, OPTIONS
Supported: replaces, 100rel, timer, norefersub
Content-Length: 0

<------------->
```

Phone Support for Presence State via SIP presence notifications

At the time of writing, only Digium phones have built-in support for interpreting Asterisk's Presence State notifications (as opposed to SIP presence notifications for extension/device state). The CustomPresence provider itself is device-agnostic and support for other devices could be added in. Or devices themselves (soft-phone or hardphone) could be modified to interpret the XML send out in the Presence State notification.

Digium Phones

This Video provides more insight on how presence can be set and viewed on Digium phones.

When using Digium phones with the Digium Phone Module for Asterisk, you can set hints in Asterisk so that when one Digium phone's presence is updated, other Digium phones can be notified of the presence change. The DPMA automatically creates provisions such that when a Digium Phone updates its presence, `CustomPresence:<line name>` is updated, where `<line name>` is the value set for the `line=` option in a `type=phone` category. Using the example dialplan from the Overview section, Digium phones that are subscribed to `2000@default` will automatically be updated about line 2000's presence whenever line 2000's presence changes.

Digium phones support only the available, away, dnd, xa, and chat states. The unavailable and not_set states are not supported.

Querying and Manipulating State

Overview

This section will enumerate and briefly describe the ways in which you can query or manipulate the various Asterisk state resources. Device State, Extension State and Presence State. Where mentioned, the various functions and commands will be linked to further available documentation.

Device State

The **DEVICE_STATE function** will return the Device State for a specified device state identifier and allow you to set Custom device states.

On the command line, the **devstate** command will allow you to list or modify Custom device states specifically.

```
devstate change                -- Change a custom device state
devstate list                  -- List currently known custom device states
```

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Extension State

The **EXTENSION_STATE function** will return the Extension State for any specified extension that has a defined hint.

The CLI command **core show hints** will show extension state for all defined hints, as well as display a truncated list of the mapped Device State or Presence State identifiers.

```
myserver*CLI> core show hints
    -= Registered Asterisk Dial Plan Hints =-
                   6002@from-internal       : SIP/6002              State:Unavailable     Watchers  0
                   7777@from-internal       : SIP/6003,CustomPrese  State:Unavailable     Watchers  0
----------------
- 2 hints registered
```

Presence State

Added in Asterisk 11, the **PRESENCE_STATE function** will return Presence State for any specified Presence State identifier, or set the Presence State for specifically for a CustomPresence identifier.

The **presencestate** CLI command will list or modify any currently defined Presence State resources provided by the CustomPresence provider.

```
myserver*CLI> core show help presencestate
presencestate change           -- Change a custom presence state
presencestate list             -- List currently know custom presence states
```

Asterisk Manager Interface actions

Any of the previously mentioned functions could be called via AMI with the Setvar and Getvar actions.

Then there are two more specific actions called ExtensionState and PresenceState. See the linked documentation for more info.

The Stasis Message Bus

Overview

✓ **Asterisk 12 and Later**
This content only applies to Asterisk 12 and later.

In Asterisk 12, a new core component was added to Asterisk: the Stasis Message Bus. As the name suggests, Stasis is an internal publish/subscribe message bus that lets the real-time core of Asterisk inform other modules or components – who subscribe for specific information topic – about events that occurred that they were interested in.

While the Stasis Message Bus is mostly of interest to those developing Asterisk, its existence is a useful piece of information in understanding how the Asterisk architecture works.

Key Concepts

The Stasis Message Bus has many concepts that work in concert together. Some of the most important are:

Publisher

A Publisher is some core component that wants to inform other components in Asterisk about some event that took place. More rarely, this can be a dynamically loadable module; most publishers however are real-time components in the Asterisk core (such as the Channel Core or the Bridging Framework).

Topic

A Topic is a high level, abstract concept that provides a way to group events together. For example, a topic may be all changes to a single channel, or all changes to all bridges in Asterisk.

Message

A Message contains the information about the event that just occurred. A Publisher publishes a Message under a specific Topic to the Stasis Message Bus.

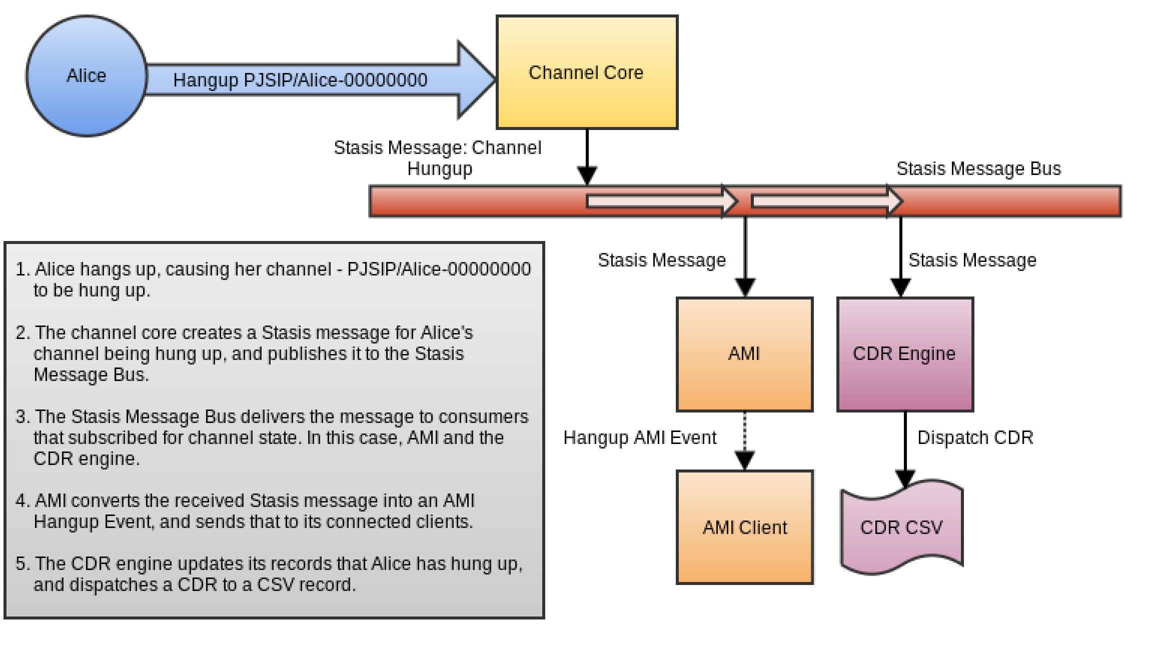
Subscriber

A Subscriber subscribes to a particular topic, and chooses which messages it is interested in. When the Stasis Message Bus receives a Message from a Publisher, it delivers the Message to each subscribed Subscriber.

Cache

Some Messages - particularly those that affect core communications primitives in Asterisk (such as channels or bridges) are stored in a special cache in Stasis. Subscribers have the option to query the cache for the last known state of those primitives.

Example: Channel Hangup



1. Alice hangs up, causing her channel - PJSIP/Alice-00000000 to be hung up.
2. The channel core creates a Stasis message for Alice's channel being hung up, and publishes it to the Stasis Message Bus.
3. The Stasis Message Bus delivers the message to consumers that subscribed for channel state. In this case, AMI and the CDR engine.
4. AMI converts the received Stasis message into an AMI Hangup Event, and sends that to its connected clients.
5. The CDR engine updates its records that Alice has hung up, and dispatches a CDR to a CSV record.

Benefits

Prior to Asterisk 12, various parts of the real-time core of Asterisk itself would have been responsible for updating AMI, the CDR Engine, and other modules/components during key operations. By decoupling the consumers of state (such as AMI or the CDR Engine) from the producer (such as the Channel Core), we have the following benefits:

- Improved Modularity: the logic for AMI, CDRs, and other consumers of state is no longer tightly coupled with the real-time components. This simplifies both the producers and the consumers.
- Insulation: because the APIs are now based on the Stasis Message Bus, changes to other parts of the Asterisk core do not immediately affect the APIs. The APIs have the ability to transform, buffer, or even discard messages from the message bus, and can choose how to represent Asterisk to their consumers. This provides increased stability for Asterisk users.
- Extensibility: because real-time state is now readily available over the message bus, adding additional consumers of state becomes much easier. New interfaces and APIs can be added to Asterisk without modifying the Asterisk core.

Configuration

This section contains many sub-sections on configuring every aspect of Asterisk. Other than what is covered under Core Configuration, most features and functionality are provided by modules that you may or may not have installed in your Asterisk system. Built-in configuration documentation for each module (that has documentation) can be accessed through the Asterisk CLI. The CLI Syntax and Help Commands section has more information on accessing the module configuration help.

Topics

Core Configuration

The sub-pages here cover any possible configuration of Asterisk's core. That is, functionality which is not separated out into modules.

If you are unfamiliar with the core and modules concepts, take a look at the Asterisk Architecture section.

Topics

Asterisk Main Configuration File

The asterisk.conf file

asterisk.conf is used to configure the locations of directories and files used by Asterisk, as well as options relevant to the core of Asterisk.

Link to the asterisk.conf.sample file in the Asterisk trunk subversion repo. The information below could become out of date, so always check the relevant sample file in our version control system.

asterisk.conf has two primary contexts, shown below with some descriptions about their content.

A Note on Includes

Includes in this file will only work with absolute paths, as the configuration in this file is setting the relative paths that would be used in includes set in other files.

Directories Context

```
[directories](!)
astetcdir => /etc/asterisk
astmoddir => /usr/lib/asterisk/modules
astvarlibdir => /var/lib/asterisk
astdbdir => /var/lib/asterisk
astkeydir => /var/lib/asterisk
astdatadir => /var/lib/asterisk
astagidir => /var/lib/asterisk/agi-bin
astspooldir => /var/spool/asterisk
astrundir => /var/run/asterisk
astlogdir => /var/log/asterisk
astsbindir => /usr/sbin
```

The directories listed above are explained in detail in the Directory and File Structure page.

Options Context

Some additional annotation for each configuration option is included inline.

TODO: Match this up with what is current in the sample, and update both.

```
 [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

; Language Options
documentation_language = en | es | ru

; 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
```

Timing Interfaces

Asterisk Timing Interfaces

In the past, if internal timing were desired for an Asterisk system, then the only source acceptable was from DAHDI. Beginning with Asterisk 1.6.1, a new timing API was introduced which allows for various timing modules to be used.
Asterisk includes the following timing modules:

- `res_timing_pthread.so`
- `res_timing_dahdi.so`
- `res_timing_timerfd.so` – *as of Asterisk 1.6.2*
- `res_timing_kqueue.so` – *as of Asterisk 11*

`res_timing_pthread` uses the POSIX pthreads library in order to provide timing. Since the code uses a commonly-implemented set of functions, `res_timing_pthread` is portable to many types of systems. In fact, this is the only timing source currently usable on a non-Linux system. Due to the fact that a single userspace thread is used to provide timing for all users of the timer, `res_timing_pthread` is also the least efficient of the timing sources and has been known to lose its effectiveness in a heavily-loaded environment.

`res_timing_dahdi` uses timing mechanisms provided by DAHDI. This method of timing was previously the only means by which Asterisk could receive timing. It has the benefit of being efficient, and if a system is already going to use DAHDI hardware, then it makes good sense to use this timing source. If, however, there is no need for DAHDI other than as a timing source, this timing source may seem unattractive. For users who are upgrading from Asterisk 1.4 and are used to the `ztdummy` timing interface, `res_timing_dahdi` provides the interface to DAHDI via the `dahdi` kernel module.

> **Historical Note**
> At the time of Asterisk 1.4's release, Zaptel (now DAHDI) was used to provide timing to Asterisk, either by utilizing telephony hardware installed in the computer or via `ztdummy` (a kernel module) when no hardware was available.
>
> When DAHDI was first released, the `ztdummy` kernel module was renamed to `dahdi_dummy`. As of DAHDI Linux 2.3.0 the `dahdi_dummy` module has been removed and its functionality moved into the main `dahdi` kernel module. As long as the `dahdi` module is loaded, it will provide timing to Asterisk either through installed telephony hardware or utilizing the kernel timing facilities when separate hardware is not available.

`res_timing_timerfd` uses a timing mechanism provided directly by the Linux kernel. This timing interface is only available on Linux systems using a kernel version at least 2.6.25 and a glibc version at least 2.8. This interface has the benefit of being very efficient, but at the time this is being written, it is a relatively new feature on Linux, meaning that its availability is not widespread.

`res_timing_kqueue` uses the Kqueue event notification system introduced with FreeBSD 4.1. It can be used on operating systems that support Kqueue, such as OpenBSD and Mac OS X. Because Kqueue is not available on Linux, this module will not compile or be available there.

What Asterisk does with the Timing Interfaces

By default, Asterisk will build and load all of the timing interfaces. These timing interfaces are "ordered" based on a hard-coded priority number defined in each of the modules. As of the time of this writing, the preferences for the modules is the following: `res_timing_timerfd.so`, `res_timing_kqueue.so` (where available), `res_timing_dahdi.so`, `res_timing_pthread.so`.

The only functionality that requires internal timing is IAX2 trunking. It may also be used when generating audio for playback, such as from a file. Even though internal timing is not a requirement for most Asterisk functionality, it may be advantageous to use it since the alternative is to use timing based on incoming frames of audio. If there are no incoming frames or if the incoming frames of audio are from an unreliable or jittery source, then the corresponding outgoing audio will also be unreliable, or even worse, nonexistent. Using internal timing prevents such unreliability.

Customizations/Troubleshooting

Now that you know Asterisk's default preferences for timing modules, you may decide that you have a different preference. Maybe you're on a timerfd-capable system but you would prefer to get your timing from DAHDI since you already are using DAHDI to drive your hardware.

Alternatively, you may have been directed to this document due to an error you are currently experiencing with Asterisk. If you receive an error message regarding timing not working correctly, then you can use one of the following suggestions to disable a faulty timing module.

1. Don't build the timing modules you know you will not use. You can disable the compilation of any of the timing modules using `menuselect`. The modules are listed in the "Resource Modules" section. Note that if you have already built Asterisk and have received an error about a timing module not working properly, it is not sufficient to disable it from being built. You will need to remove the module from your modules directory (by default, `/usr/lib/asterisk/modules`) to make sure that it does not get loaded again.
2. Build, but do not load the timing modules you know you will not use. You can edit `modules.conf` using `noload` directives to disable the loading of specific timing modules by default. Based on the note in the section above, you may realize that your Asterisk setup does not require internal timing at all. If this is the case, you can safely `noload` all timing modules.

> Some confusion has arisen regarding the fact that non-DAHDI timing interfaces are available now. One common misconception which has arisen is that since timing can be provided elsewhere, DAHDI is no longer required for using the MeetMe application. Unfortunately, this is not the case. In addition to providing timing, DAHDI also provides a conferencing engine which the MeetMe application requires.
>
> Starting with Asterisk 1.6.2, however, there is a new application, ConfBridge, which is capable of conference bridging without the use of DAHDI's

built-in mixing engine.

Asterisk Builtin mini-HTTP Server

Overview

The core of Asterisk provides a basic HTTP/HTTPS server.

Certain Asterisk modules may make use of the HTTP service, such as the Asterisk Manager Interface over HTTP, the Asterisk Restful Interface or WebSocket transports for modules that support that, like chan_sip or chan_pjsip.

Configuration

The configuration sample file is by default located at /etc/asterisk/http.conf

A very basic configuration of http.conf could be as follows:

```
[general]
enabled=yes
bindaddr=0.0.0.0
bindport=8088
```

That configuration would enable the HTTP server and have it bind to all available network interfaces on port 8088.

Configuration Options

See the sample file in your version of Asterisk for detail on the various configuration options, as this information is not yet automatically pushed to the wiki.

Logging Configuration

Asterisk Log File Configuration

General purpose logging facilities in Asterisk can be configured in the *logger.conf* file. Within this file one is able to configure Asterisk to log messages to files and/or a syslog and even to the Asterisk console. Note, the sections and descriptions listed below are meant to be informational and act as a guide (a "how to") when configuring logging in Asterisk. Options with stated defaults don't have to be explicitly set as they will simply default to a designated value.

General Section:

```
[general]
; Customize the display of debug message time stamps
; this example is the ISO 8601 date format (yyyy-mm-dd HH:MM:SS)
;
; see strftime(3) Linux manual for format specifiers.  Note that there is
; also a fractional second parameter which may be used in this field.  Use
; %1q for tenths, %2q for hundredths, etc.
;
dateformat = %F %T.%3q   ; ISO 8601 date format with milliseconds

; Write callids to log messages (defaults to yes)
use_callids = yes

; Append the hostname to the name of the log files (defaults to no)
appendhostname = no

; Log queue events to a file (defaults to yes)
queue_log = yes

; Always log queue events to a file, even when a realtime backend is
; present (defaults to no).
queue_log_to_file = no

; Set the queue_log filename (defaults to queue_log)
queue_log_name = queue_log

; When using realtime for the queue log, use GMT for the timestamp
; instead of localtime.  (defaults to no)
queue_log_realtime_use_gmt = no

; Log rotation strategy (defaults to sequential):
; none:  Do not perform any log rotation at all.  You should make
;        very sure to set up some external log rotate mechanism
;        as the asterisk logs can get very large, very quickly.
; sequential:  Rename archived logs in order, such that the newest
;              has the highest sequence number.  When
;              exec_after_rotate is set, ${filename} will specify
;              the new archived logfile.
; rotate:  Rotate all the old files, such that the oldest has the
;          highest sequence number (this is the expected behavior
;          for Unix administrators).  When exec_after_rotate is
;          set, ${filename} will specify the original root filename.
; timestamp:  Rename the logfiles using a timestamp instead of a
;             sequence number when "logger rotate" is executed.
;             When exec_after_rotate is set, ${filename} will
;             specify the new archived logfile.
rotatestrategy = rotate

; Run a system command after rotating the files.  This is mainly
; useful for rotatestrategy=rotate. The example allows the last
; two archive files to remain uncompressed, but after that point,
; they are compressed on disk.
exec_after_rotate=gzip -9 ${filename}.2
```

Log Files Section:

```
[logfiles]
; File names can either be relative to the standard Asterisk log directory (see "astlogdir" in
; asterisk.conf), or absolute paths that begin with '/'.
;
; A few file names have been reserved and are considered special, thus cannot be used and will
; not be considered as a regular file name.  These include the following:
;
;    syslog - logs to syslog facility
;    console - logs messages to the Asterisk root console.
;
; For each file name given a comma separated list of logging "level" types should be specified
; and include at least one of the following (in no particular order):
;    debug
;    notice
;    warning
;    error
;    dtmf
;    fax
;    security
;    verbose(<level>)
;
; The "verbose" value can take an optional integer argument that indicates the maximum level
; of verbosity to log at.  Verbose messages with higher levels than the indicated level will
; not be logged to the file.  If a verbose level is not given, verbose messages are logged
; based upon the current level set for the root console.
;
; The special character "*" can also be specified and represents all levels, even dynamic
; levels registered by modules after the logger has been initialized.  This means that loading
; and unloading modules that create and remove dynamic logging levels will result in these
; levels being included on filenames that have a level name of "*", without any need to
; perform a "logger reload" or similar operation.
;
; Note, there is no value in specifying both "*" and specific level types for a file name.
; The "*" level means ALL levels.  The only exception is if you need to specify a specific
; verbose level. e.g, "verbose(3),*".
;
; It is highly recommended that you DO NOT turn on debug mode when running a production system
; unless you are in the process of debugging a specific issue.  Debug mode outputs a LOT of
; extra messages and information that can and do fill up log files quickly. Most of these
; messages are hard to interpret without an understanding of the underlying code.  Do NOT report
; debug messages as code issues, unless you have a specific issue that you are attempting to debug.
; They are messages for just that -- debugging -- and do not rise to the level of something that
; merit your attention as an Asterisk administrator.

; output notices, warnings and errors to the console
console => notice,warning,error

; output security messages to the file named "security"
security => security

; output notices, warnings and errors to the the file named "messages"
messages => notice,warning,error

; output notices, warnings, errors, verbose, dtmf, and fax to file name "full"
full => notice,warning,error,verbose,dtmf,fax

; output notices, warning, and errors to the syslog facility
syslog.local0 => notice,warning,error
```

Asterisk CLI Configuration

With the exception of the functionality provided by the res_clialiases.so module, Asterisk's Command Line Interface is provided by the core. There are a few configuration files relevant to the CLI that you'll see in a default Asterisk installation. All of these should be found in the typical /etc/asterisk/ directory in a default install. The configuration of these files is trivial and examples exist in the sample files included in the source and tarballs.

cli.conf

This file allows a listing of CLI commands to be automatically executed upon startup of Asterisk.

cli_permissions.conf

Allows you to configure specific restrictions or allowances on commands for users connecting to an Asterisk console. Read through the sample file carefully before making use of it, as you could create security issues.

cli_aliases.conf

This file allows configuration of aliases for existing commands. For example, the 'help' command is really an alias to 'core show help'. This functionality is provided by the res_clialiases.so module.

CLI related commands

There are a few commands relevant to the CLI configuration itself.

- **cli check permissions** - allows you to try running a command through the permissions of a specified user
- **cli reload permissions** - reloads the cli_permissions.conf file
- **cli show permissions** - shows configured CLI permissions
- **cli show aliases** - shows configured CLI command aliases

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

Configuring the Asterisk Module Loader

Overview

As you may have learned from the Asterisk Architecture section, the majority of Asterisk's features and functionality are separated outside of the core into various **modules**. Each module has distinct functionality, but sometimes relies on another module or modules.

Asterisk provides capability to automatically and manually load modules. Module load order can be configured before load-time, or modules may be loaded and unloaded during run-time.

Configuration

The configuration file for Asterisk's module loader is **modules.conf**. It is read from the typical Asterisk configuration directory. You can also view the sample of modules.conf file in your source directory at configs/modules.conf.sample or on SVN at this link.

The configuration consist of one large section called "modules" with possible directives configured within.

There are several directives that can be used.

- autoload - When enabled, Asterisk will automatically load any modules found in the Asterisk modules directory.
- preload - Used to specify individual modules to load before the Asterisk core has been initialized. Often used for realtime modules so that config files can be pushed to a backend before the dependent modules are loaded.
- require - Set a required module. If a required module does not load, then Asterisk exits with status code 2.
- preload-require - A combination of preload and require.
- noload - Do not load the specified module.
- load - Load the specified module. Typically used when autoload is set to 'no'.

Let's show a few arbitrary examples below.

```
[modules]
;autoload = yes
;preload = res_odbc.so
;preload = res_config_odbc.so
;preload-require = res_odbc.so
;require = res_pjsip.so
;noload = pbx_gtkconsole.so
;load = res_musiconhold.so
```

CLI Commands

Asterisk provides a few commands for managing modules at run-time. Be sure to check the current usage using the CLI help with "core show help <command>".

- module show

```
Usage: module show [like keyword]
       Shows Asterisk modules currently in use, and usage statistics.
```

- module load

```
Usage: module load <module name>
       Loads the specified module into Asterisk.
```

- module unload

```
Usage: module unload [-f|-h] <module_1> [<module_2> ... ]
       Unloads the specified module from Asterisk. The -f
       option causes the module to be unloaded even if it is
       in use (may cause a crash) and the -h module causes the
       module to be unloaded even if the module says it cannot,
       which almost always will cause a crash.
```

- module reload

```
Usage: module reload [module ...]
       Reloads configuration files for all listed modules which support
       reloading, or for all supported modules if none are listed.
```

Configuring Localized Tone Indications

Overview

In certain cases Asterisk will generate tones to be used in call signaling. It may be during the use of a specific application, or with certain channel drivers. The tones used are configurable and may be defined by location.

Note that the tones configured here are only used when Asterisk is directly generating the tones.

Configuration

The configuration file for location specific tone indications is **indications.conf**. It is read from the typical Asterisk configuration directory. You can also view the sample of indications.conf file in your source directory at configs/modules.conf.sample or on SVN at this link.

The configuration itself consists of a 'general' section and then one or more country specific sections. (e.g. '[au]' for Australia)

Within the general section, only the **country** option can be set. This option sets the default location tone set to be used.

```
[general]
country=us
```

As an example, the above set the default country to the tone set for the USA.

Within any location specific configuration, several tone types may be configured.

- description = string ;   The full name of your country, in English.
- ringcadence = num[,num]* ;   List of durations the physical bell rings.
- dial = tonelist   ;   Set of tones to be played when one picks up the hook.
- busy = tonelist ;   Set of tones played when the receiving end is busy.
- congestion = tonelist   ;   Set of tones played when there is some congestion (on the network?)
- callwaiting = tonelist   ;   Set of tones played when there is a call waiting in the background.
- dialrecall = tonelist   ;   Not well defined; many phone systems play a recall dial tone after hook flash
- record = tonelist ;   Set of tones played when call recording is in progress.
- info = tonelist ;   Set of tones played with special information messages (e.g., "number is out of service")
- 'name' = tonelist ;   Every other variable will be available as a shortcut for the "PlayList" command but will not be used automatically by Asterisk.

Explanation of the 'tonelist' usage:

```
; The tonelist itself is defined by a comma-separated sequence of elements.
; Each element consist of a frequency (f) with an optional duration (in ms)
; attached to it (f/duration). The frequency component may be a mixture of two
; frequencies (f1+f2) or a frequency modulated by another frequency (f1*f2).
; The implicit modulation depth is fixed at 90%, though.
; If the list element starts with a !, that element is NOT repeated,
; therefore, only if all elements start with !, the tonelist is time-limited,
; all others will repeat indefinitely.
;
; concisely:
;   element = [!]freq[+|*freq2][/duration]
;   tonelist = element[,element]*
```

Example of a location specific tone configuration:

```
[br]
description = Brazil
ringcadence = 1000,4000
dial = 425
busy = 425/250,0/250
ring = 425/1000,0/4000
congestion = 425/250,0/250,425/750,0/250
callwaiting = 425/50,0/1000
; Dialrecall not used in Brazil standard (using UK standard)
dialrecall = 350+440
; Record tone is not used in Brazil, use busy tone
record = 425/250,0/250
; Info not used in Brazil standard (using UK standard)
info = 950/330,1400/330,1800/330
stutter = 350+440
```

Video Telephony

Asterisk and Video telephony

Asterisk supports video telephony in the core infrastructure. Internally, it's one audio stream and one video stream in the same call. Some channel drivers and applications has video support, but not all.

Codecs and formats

Asterisk supports the following video codecs and file formats. There's no video transcoding so you have to make sure that both ends support the same video format.

Codec	Format	Notes
H.263	read/write	
H.264	read/write	
H.261	-	Passthrough only

Note that the file produced by Asterisk video format drivers is in no generic video format. Gstreamer has support for producing these files and converting from various video files to Asterisk video+audio files.

Note that H.264 is not enabled by default. You need to add that in the channel configuration file.

Channel Driver Support

Channel Driver	Module	Notes
SIP	`chan_sip.so`	The SIP channel driver (chan_sip.so) has support for video
IAX2	`chan_iax2.so`	Supports video calls (over trunks too)
Local	`chan_local.so`	Forwards video calls as a proxy channel
Agent	`chan_agent.so`	Forwards video calls as a proxy channel
oss	`chan_oss.so`	Has support for video display/decoding, see video_console.txt

Applications

This is not yet a complete list. These dialplan applications are known to handle video:

- Voicemail - Video voicemail storage (does not attach video to e-mail)
- Record - Records audio and video files (give audio format as argument)
- Playback - Plays a video while being instructed to play audio
- Echo - Echos audio and video back to the user

There is a development group working on enhancing video support for Asterisk.

If you want to participate, join the asterisk-video mailing list on http://lists.digium.com

Updates to this file are more than welcome!

Video Console

Video Console Support in Asterisk

Some console drivers (at the moment chan_oss.so) can be built with support for sending and receiving video. In order to have this working you need to perform the following steps:

Enable building the video_console support

The simplest way to do it is add this one line to channels/Makefile:

```
chan_oss.so: _ASTCFLAGS+=-DHAVE_VIDEO_CONSOLE
```

Install prerequisite packages

The video_console support relies on the presence of SDL, SDL_image and ffmpeg libraries, and of course on the availability of X11

On Linux, these are supplied by

- libncurses-dev
- libsdl1.2-dev
- libsdl-image1.2-dev
- libavcodec-dev
- libswcale-dev

On FreeBSD, you need the following ports:

- multimedia/ffmpeg (2007.10.04)
- devel/sdl12 graphics/sdl_image

Build and install asterisk with all the above

Make sure you do a 'make clean' and run configure again after you have installed the required packages, to make sure that the required pieces are found.

Check that chan_oss.so is generated and correctly installed.

Update configuration files

Video support requires explicit configuration as described below:

oss.conf
You need to set various parameters for video console, the easiest way is to uncomment the following line in oss.conf by removing the leading ';'

```
;[general](+,my_video,skin2)
```

You also need to manually copy the two files

- images/kpad2.jpg
- images/font.png

into the places specified in oss.conf, which in the sample are set to

```
keypad = /tmp/kpad2.jpg
keypad_font = /tmp/font.png
```

other configuration parameters are described in oss.conf.sample

sip.conf

To actually run a call using SIP (the same probably applies to iax.conf) you need to enable video support as following

```
[general](+)
videosupport=yes
allow=h263      ; this or other video formats
allow=h263p     ; this or other video formats
```

You can add other video formats e.g. h261, h264, mpeg if they are supported by your version of libavcodec.

Run the Program

Run asterisk in console mode e.g. asterisk -vdc

If video console support has been successfully compiled in, then you will see the "console startgui" command available on the CLI interface. Run the command, and you should see a window like this http://info.iet.unipi.it/~luigi/asterisk_video_console.jpg

To exit from this window, in the console run "console stopgui".

If you want to start a video call, you need to configure your dialplan so that you can reach (or be reachable) by a peer who can support video. Once done, a video call is the same as an ordinary call:

"console dial ...", "console answer", "console hangup" all work the same.

To use the GUI, and also configure video sources, see the next section.

Video Sources

Video sources are declared with the "videodevice=..." lines in oss.conf where the ... is the name of a device (e.g. /dev/video0 ...) or a string starting with X11 which identifies one instance of an X11 grabber.

You can have up to 9 sources, displayed in thumbnails in the gui, and select which one to transmit, possibly using Picture-in-Picture.

For webcams, the only control you have is the image size and frame rate (which at the moment is the same for all video sources). X11 grabbers capture a region of the X11 screen (it can contain anything, even a live video) and use it as the source. The position of the grab region can be configured using the GUI below independently for each video source.

The actual video sent to the remote side is the device selected as "primary" (with the mouse, see below), possibly with a small 'Picture-in-Picture' of the "secondary" device (all selectable with the mouse).

GUI Commands and Video Sources

(most of the text below is taken from channels/console_gui.c)

The GUI is made of 4 areas: remote video on the left, local video on the right, keypad with all controls and text windows in the center, and source device thumbnails on the top. The top row is not displayed if no devices are specified in the config file.

```
________________________________________________________________
    |  ______   ______   ______   ______   ______   ______   ______  |
    | | tn.1 | | tn.2 | | tn.3 | | tn.4 | | tn.5 | | tn.6 | | tn.7 | |
    | |______| |______| |______| |______| |______| |______| |______| |
    |  ______   ______   ______   ______   ______   ______   ______  |
    | |______| |______| |______| |______| |______| |______| |______| |
    |  _________________    __________________    _________________  |
    | |                 |  |                  |  |                 | |
    | |                 |  |                  |  |                 | |
    | |                 |  |                  |  |                 | |
    | |   remote video  |  |                  |  |   local video   | |
    | |                 |  |                  |  |          ______ | |
    | |                 |  |      keypad      |  |         |  PIP || |
    | |                 |  |                  |  |         |______|| |
    | |_________________|  |                  |  |_________________| |
    |                      |                  |                      |
    |                      |                  |                      |
    |                      |__________________|                      |
    |________________________________________________________________|
```

The central section is built using an image (jpg, png, maybe gif too) for the skin and other GUI elements. Comments embedded in the image indicate to what function each area is mapped to.

Another image (png with transparency) is used for the font.

Mouse and keyboard events are detected on the whole surface, and handled differently according to their location:

- Center/right click on the local/remote window are used to resize the corresponding window
- Clicks on the thumbnail start/stop sources and select them as primary or secondary video sources
- Drag on the local video window are used to move the captured area (in the case of X11 grabber) or the picture-in-picture position
- Keystrokes on the keypad are mapped to the corresponding key; keystrokes are used as keypad functions, or as text input

if we are in text-input mode.

- Drag on some keypad areas (sliders etc.) are mapped to the corresponding functions (mute/unmute audio and video, enable/disable Picture-in-Picture, freeze the incoming video, dial numbers, pick up or hang up a call, ...)

Named ACLs

Overview

Named ACLs introduce a new way to define Access Control Lists (ACLs) in Asterisk. Unlike traditional ACLs defined in specific module configuration files, Named ACLs can be shared across multiple modules. Named ACLs can also be accessed via the Asterisk Realtime Architecture (ARA), allowing for run-time updates of ACL information that can be retrieved by multiple consumers of ACL information.

Configuration

Static Configuration

Named ACLs can be defined statically in *acl.conf*. Each context in *acl.conf* defines a specific Named ACL, where the name of the context is the name of the ACL. The syntax for each context follows the permit/deny nomenclature used in traditional ACLs defined in a consumer module's configuration file.

Option	Value	Description
deny	IP address [/Mask]	An IP address to deny, with an optional subnet mask to apply
permit	IP address [/Mask]	An IP address to allow, with an optional subnet mask to apply

Examples

```
; within acl.conf

[name_of_acl1]
deny=0.0.0.0/0.0.0.0
permit=127.0.0.1
```

Multiple rules can be specified in an ACL as well by chaining deny/permit specifiers.

```
[name_of_acl2]
deny=10.24.0.0/255.255.0.0
deny=10.25.0.0/255.255.0.0
permit=10.24.11.0/255.255.255.0
permit=10.24.12.0/255.255.255.0
```

Named ACLs support common modifiers like templates and additions within configuration as well.

```
[template_deny_all](!)
deny=0.0.0.0/0.0.0.0

[deny_all_whitelist_these](template_deny_all)
permit=10.24.20.1
permit=10.24.20.2
permit=10.24.20.3
```

Configuring for IPv6

Named ACLs can use ipv6 addresses just like normal ACLs.

```
[ipv6_example_1]
deny = ::
permit = ::1/128

[ipv6_example_2]
permit = fe80::21d:bad:fad:2323
```

ARA Configuration

The ARA supports Named ACLs using the '**acls**' keyword in *extconfig.conf*.

Example Configuration

```
;in extconfig.conf
acls => odbc,asterisk,acltable
```

Schema

Column Name	Type	Description
name	varchar(80)	Name of the ACL
rule_order	integer	Order to apply the ACL rule. Rules are applied in ascending order. Rule numbers do not have to be sequential
sense	varchar(6)	Either 'permit' or 'deny'
rule	varchar(95)	The IP address/Mask pair to apply

Examples

Table Creation Script (PostgreSQL)

```
CREATE TABLE acltable
(
   "name" character varying(80) NOT NULL,
   rule_order integer NOT NULL,
   sense character varying(6) NOT NULL,
   "rule" character varying(95) NOT NULL,
   CONSTRAINT aclrulekey PRIMARY KEY (name, rule_order, rule, sense)
)
WITH (
   OIDS=FALSE
);
ALTER TABLE acltable OWNER TO asterisk;
GRANT ALL ON TABLE acltable TO asterisk;
)
```

Table Creation Script (SQLite3)

```
BEGIN TRANSACTION;
CREATE TABLE acltable (rule TEXT, sense TEXT, rule_order NUMERIC, name TEXT);
COMMIT;
```

⚠ These scripts were generated by pgadmin III and SQLite Database Browser. They might not necessarily apply for your own setup.

⚠ Since ACLs are obtained by consumer modules when they are loaded, an ACL updated in an ARA backend will not be propagated automatically to consumers using static configuration. Consumer modules also using ARA for their configuration (such as SIP/IAX2 peers) will similarly be up to date if and only if they have built the peer in question since the changes to the realtime ACL have taken place.

Named ACL Consumers

Named ACLs are supported by the following Asterisk components:

- Manager (IPv4 and IPv6)
- chan_sip (IPv4 and IPv6)
- chan_pjsip (IPv4 and IPv6)
- chan_iax2 (IPv4 and IPv6)

Configuration

A consumer of Named ACLs can be configured to use a named ACL using the *acl* option in their ACL access rules. This can be in addition to the ACL rules traditionally defined in those configuration files.

Example 1: referencing a Named ACL

```
; within sip.conf

[peer1]
;stuff
;deny=0.0.0.0/0.0.0.0
;permit=127.0.0.1
acl=name_of_acl_1 ; an ACL included from acl.conf that matches peer1's commented out permits/denies
```

Multiple named ACLs can be referenced as well by specifying a comma delineated list of Named ACLs to apply.

Example 2: multiple Named ACL references

```
; within sip.conf

[peer1]
;stuff
acl=named_acl_1,named_acl_2
```

Similarly, a SIP or IAX2 peer defined in ARA can include an '*acl*' column and list the Named ACLs to apply in that column.

> ⚠ **NOTE**
> Named ACLs can also be defined using multiple instances of the *acl* keyword. This is discouraged, however, as the order in which ACLs are applied can be less obvious then the comma delineated list format.
>
> ```
> acl=named_acl_1
> acl=named_acl_2
> ```

ACL Rule Application

Each module consumer of ACL information maintains, for each object that uses the information, a list of the defined ACL rule sets that apply to that object. When an address is evaluated for the particular object, the address is evaluated against each rule. For an address to pass the ACL rules, it must pass each ACL rule set that was defined for that object. Failure of any ACL rule set will result in a rejection of the address.

Module Reloads

ACL information is static once a consumer module references that information. Hence, changes in ACL information in an ARA backend will not automatically update consumers of that information. In order for consumers to receive updated ACL information, the Named ACL component must be reloaded.

The Named ACL component supports module reloads, in the same way as other Asterisk components. When the Named ACL component is reloaded, it will issue a request to all consumers of Named ACLs. Those consumer modules will also be automatically reloaded.

> ⓘ **WARNING**
> This implies that reloading the Named ACL component will force a reload of manager, chan_sip, etc. Only reload the Named ACL component if you want all consumers of that information to be reloaded as well.

Channel Drivers

All about Asterisk and its Channel Drivers

Topics

SIP

Under Construction

Section to hold information on configuring the SIP channel drivers, chan_sip and chan_pjsip

Configuring chan_sip

Under Construction - This is a stub

Currently the documentation resides in the sip.conf.sample file included with the source. We are in the process of updating the wiki!

chan_sip State and Presence Options

Device State

There are a few configuration options for chan_sip that affect Device State behavior.

callcounter

The **callcounter** option in sip.conf **must be enabled** for SIP devices (e.g. SIP/Alice) to provide advanced device state. Without it you may see some state, such as unavailable or idle, but not much more.

The option can be set in the general context, or on a per-peer basis.

Default: no

```
[general]
callcounter=yes
```

busylevel

The **busylevel** option only works if call counters are enabled via the above option. If call counters are enabled, then busylevel allows you to set a threshold for when to consider this device busy. If busylevel is set to 2, then only at 2 or more calls will the device state report BUSY. The busylevel option can only be set for peers.

Default: 0

```
[6001]
type=friend
busylevel=2
```

notifyhold

The **notifyhold** option, when enabled, adds the ONHOLD device state to the range of possible device states that chan_sip will use.

This option can only be set in the general section.

Default: yes

```
[general]
notifyhold=no
```

Extension State, Hints, Subscriptions

Extension State and subscriptions tend to go hand in hand. That is, if you are using Extension State, you probably have SIP user agents subscribing to those extensions/hints. These options all affect that behavior.

allowsubscribe

The **allowsubscribe** option enables or disables support for any kind of subscriptions. You can set allowsubscribe per-peer or in the general section.

Default: yes

```
[6001]
type=friend
allowsubscribe=no
```

subscribecontext

subscribecontext sets a specific context to be used for subscriptions. That means, if SIP user agent subscribes to this peer, Asterisk will search for an associated hint mapping in the context specified.

This option can be set per-peer or in the general section.

Default: null (by default Asterisk will use the context specified with the "context" option)

```
[6001]
type=friend
context=internal
subscribecontext=myhints
```

notifyringing

notifyringing enables or disables notifications for the RINGING state when an extension is already INUSE. Only affects subscriptions using the **dialog-info** event package. Option can be configured in the general section only. It cannot be set per-peer.

Default: yes

```
[general]
notifyringing=no
```

notifycid

notifycid some nuance and may only be relevant to SNOM phones or others that support dialog-info+xml notifications. Below are the notes from the sample sip.conf.

Default: no

```
;notifycid = yes                ; Control whether caller ID information is sent along with
                                ; dialog-info+xml notifications (supported by snom phones).
                                ; Note that this feature will only work properly when the
                                ; incoming call is using the same extension and context that
                                ; is being used as the hint for the called extension.  This means
                                ; that it won't work when using subscribecontext for your sip
                                ; user or peer (if subscribecontext is different than context).
                                ; This is also limited to a single caller, meaning that if an
                                ; extension is ringing because multiple calls are incoming,
                                ; only one will be used as the source of caller ID.  Specify
                                ; 'ignore-context' to ignore the called context when looking
                                ; for the caller's channel.  The default value is 'no.' Setting
                                ; notifycid to 'ignore-context' also causes call-pickups attempted
                                ; via SNOM's NOTIFY mechanism to set the context for the call pickup
                                ; to PICKUPMARK.
```

Configuring chan_sip for IPv6

Configuring chan_sip for IPv6

Mostly you can use IPv6 addresses where you would have otherwise used IPv4 addresses within sip.conf. The sip.conf.sample provides several examples of how to use the various options with IPv6 addresses. We'll provide a few examples here as well.

Examples

Binding to a specific IPv6 interface

```
[general]
bindaddr=2001:db8::1
```

Binding to all available IPv6 interfaces (wildcard)

```
[general]
bindaddr=::
```

You can specify a port number by wrapping the address in square brackets and using a colon delimiter.

```
[general]
bindaddr=[::]:5062
```

✓ You can choose independently for UDP, TCP, and TLS, by specifying different values for "udpbindaddr", "tcpbindaddr", and "tlsbindaddr".

Note that using bindaddr=:: will show only a single IPv6 socket in netstat. IPv4 is supported at the same time using IPv4-mapped IPv6 addresses.)

Other Options

Other options such as "outboundproxy" or "permit" can use IPv6 addresses the same as in the above examples.

```
permit=2001:db8::/32
```

Configuring chan_sip for Presence Subscriptions

Overview

This page is a rough guide to get you configuring chan_sip and Asterisk to accept subscriptions for presence (in this case, Extension State) and notify the subscribers of state changes.

Requirements

You should understand the basics of

- Device State and Extension State and Hints
- Configuring SIP peers in sip.conf

General Process

Overview

It is best to consider this configuration in the context of a very simplified use case. It should illustrate the overall concept, as well as the ability for Extension State to aggregate Device States.

The case is that our administrator wants the user device of SIP/Alice to display the presence of Bob. Bob has two devices, SIP/Bob-mobile and SIP/Bob-desk. He could be on either device at any one time, so we want to map them both to the same Hint. That way, when Alice subscribes to the Hint, she'll get the aggregated Extension State of Bob's devices. That means if either of Bobs phones are busy, then the extension state will be busy. Then Alice knows that Bob is busy without having to have a separate light for each of Bob's phones.

Figure 1 should illustrate the overall relationships of the different elements involved.

Then following down the page you can find detail on configuring the three major elements, SIP configuration options, hints in dialplan, and configuring a phone to subscribe.

Configure SIP options

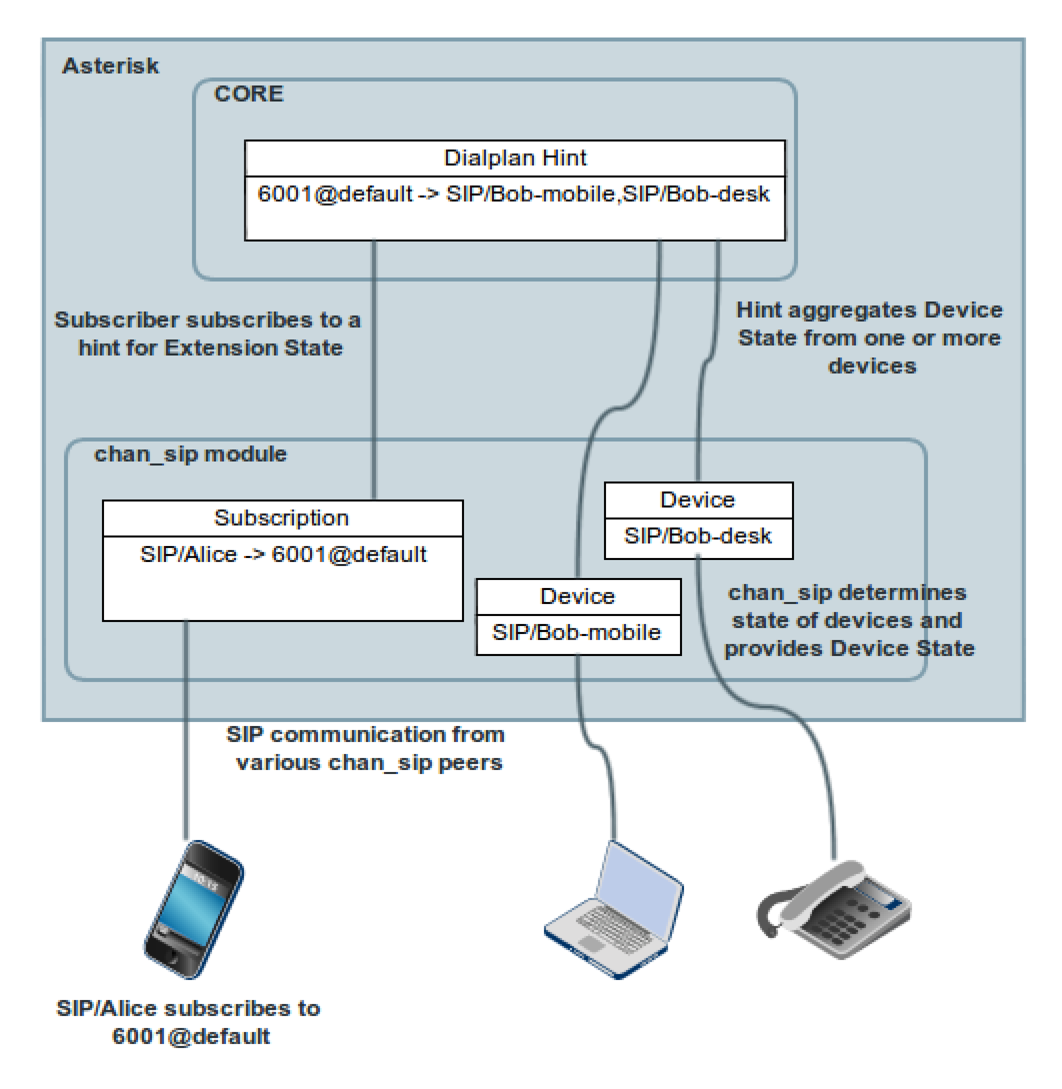
Since this is not a guide on configuring SIP peers, we'll show a very simple **sip.conf** with only enough configuration to point out where you might set specific chan_sip State and Presence Options .

```
[general]
callcounter=yes

[Alice]
type=friend
subscribecontext=default
allowsubscribe=yes

[Bob-mobile]
type=friend
busylevel=1

[Bob-desk]
type=friend
busylevel=1
```

We are setting one option in the general section, and then a few options across the three SIP peers involved.

callcounter and **busylevel** are the most essential options. **callcounter** needs to be enabled for chan_sip to provide accurate device. **busylevel**=1 says we want the device states of those peers to show busy if they have at least one call in progress. The **subscribecontext** option tells Asterisk which dialplan context to look for the hint. **allowsubscribe** says that we will allow subscriptions for that peer. It is really set to yes by default, but we are defining it here to demonstrate that you could allow and disallow subscriptions on a per-peer basis if you wanted.

Figure 1


Asterisk
CORE
Dialplan Hint
6001@default -> SIP/Bob-mobile,SIP/Bob-desk
Subscriber subscribes to a hint for Extension State
Hint aggregates Device State from one or more devices
chan_sip module
Subscription
SIP/Alice -> 6001@default
Device
SIP/Bob-desk
Device
SIP/Bob-mobile
chan_sip determines state of devices and provides Device State
SIP communication from various chan_sip peers
SIP/Alice subscribes to 6001@default

This diagram is purposefully simplified to only show the relationships between the elements involved in this configuration.

Configure Hints

Hints are configured in Asterisk dialplan (extensions.conf). This is where you map Device State identifiers or Presence State identifiers to a hint, which will then be subscribed to by one or more SIP User Agents.

For our example we need to define a hint mapping 6001 to Bob's two devices.

```
[default]
exten = 6001,hint,SIP/Bob-mobile&SIP/Bob-desk
```

Defining the hint is pretty straightforward and follows the syntax discussed in the Extension State and Hints section.

Notice that we put it in the context we set in **subscribecontext** in sip.conf earlier. Otherwise we would need to make sure it is in the same context that the SIP peer uses (defined with "context").

If you have restarted Asterisk to load the hints, then you can check to make sure they are configured with "core show hints"

```
*CLI> core show hints
    -= Registered Asterisk Dial Plan Hints =-
                   6001@default             : SIP/Bob-mobile&SIP/B  State:Unavailable     Watchers  0
```

You'll see the state changes to Idle or something else if you have your sip.conf configured properly and the two SIP devices are at least available.

Configure Subscriber

You should configure your SIP User Agent (soft-phone, hard-phone, another phone application like Asterisk) to subscribe to the hint. In this case that is SIP/Alice and we want her phone to subscribe to 6001.

The process will be different for every phone, and keep in mind that some phones may not support Asterisk's state notification. With most phones it'll be a matter of adding a "contact" to a contact list, buddy list, or address book and then making sure that SIP presence is enabled in the options.

If you want to submit a guide for a specific phone, feel free to comment on this page or submit it to the Asterisk issue tracker.

Operation

Typically as soon as you add the contact or subscription on the phone then it will attempt to SUBSCRIBE to Asterisk.

If you haven't done so, restart Asterisk and then restart the SIP User Agent client doing the subscribing.

The flow of SIP messaging can differ based on configuration, but typically looks like this for a peer that requires authentication:

```
SIP/Alice                    Asterisk
----------------------------------------
SUBSCRIBE          --->
                   <---      401 Unauthorized
SUBSCRIBE(w/ Auth) --->
                   <---      200 OK
                   <---      NOTIFY
200 OK             --->
```

In the expanding frame below is a SIP trace of a successful subscription for reference. You could see this on your own system by running "sip set debug on" and then watching for the subscription. You might have to restart your phone again or re-add a contact to see it.

Click to see the subscription trace...

```
<--- SIP read from UDP:10.24.17.254:37509 --->
SUBSCRIBE sip:6001@10.24.18.124;transport=UDP SIP/2.0
Via: SIP/2.0/UDP 10.24.17.254:37509;branch=z9hG4bK-d8754z-e5ecfde1f337b690-1---d8754z-
Max-Forwards: 70
Contact: <sip:Alice@10.24.17.254:37509;transport=UDP>
To: <sip:6001@10.24.18.124;transport=UDP>
From: <sip:Alice@10.24.18.124;transport=UDP>;tag=f51e9632
Call-ID: ZjE2ZDAwYThiOTA2MzYxOWEwNTEwMjc1ZGIxNTk3NDU.
CSeq: 1 SUBSCRIBE
Expires: 1800
Accept: application/pidf+xml
Allow: INVITE, ACK, CANCEL, BYE, NOTIFY, REFER, MESSAGE, OPTIONS, INFO, SUBSCRIBE
Supported: replaces, norefersub, extended-refer, timer, X-cisco-serviceuri
User-Agent: Z 3.2.21357 r21103
Event: presence
Allow-Events: presence, kpml
Content-Length: 0

<------------->
--- (16 headers 0 lines) ---
Sending to 10.24.17.254:37509 (no NAT)
Creating new subscription
Sending to 10.24.17.254:37509 (no NAT)
list_route: route/path hop: <sip:Alice@10.24.17.254:37509;transport=UDP>
Found peer 'Alice' for 'Alice' from 10.24.17.254:37509

<--- Transmitting (no NAT) to 10.24.17.254:37509 --->
SIP/2.0 401 Unauthorized
Via: SIP/2.0/UDP 10.24.17.254:37509;branch=z9hG4bK-d8754z-e5ecfde1f337b690-1---d8754z-;received=10.24.17.254
From: <sip:Alice@10.24.18.124;transport=UDP>;tag=f51e9632
To: <sip:6001@10.24.18.124;transport=UDP>;tag=as46a6e039
Call-ID: ZjE2ZDAwYThiOTA2MzYxOWEwNTEwMjc1ZGIxNTk3NDU.
CSeq: 1 SUBSCRIBE
Server: Asterisk PBX SVN-branch-12-r413487
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH, MESSAGE
Supported: replaces, timer
WWW-Authenticate: Digest algorithm=MD5, realm="asterisk", nonce="522456f4"
Content-Length: 0


<------------>
Scheduling destruction of SIP dialog 'ZjE2ZDAwYThiOTA2MzYxOWEwNTEwMjc1ZGIxNTk3NDU.' in 32000 ms (Method: SUBSCRIBE)

<--- SIP read from UDP:10.24.17.254:37509 --->
SUBSCRIBE sip:6001@10.24.18.124;transport=UDP SIP/2.0
Via: SIP/2.0/UDP 10.24.17.254:37509;branch=z9hG4bK-d8754z-c6908de6f0126edf-1---d8754z-
Max-Forwards: 70
Contact: <sip:Alice@10.24.17.254:37509;transport=UDP>
To: <sip:6001@10.24.18.124;transport=UDP>
From: <sip:Alice@10.24.18.124;transport=UDP>;tag=f51e9632
Call-ID: ZjE2ZDAwYThiOTA2MzYxOWEwNTEwMjc1ZGIxNTk3NDU.
CSeq: 2 SUBSCRIBE
Expires: 1800
```

```
Accept: application/pidf+xml
Allow: INVITE, ACK, CANCEL, BYE, NOTIFY, REFER, MESSAGE, OPTIONS, INFO, SUBSCRIBE
Supported: replaces, norefersub, extended-refer, timer, X-cisco-serviceuri
User-Agent: Z 3.2.21357 r21103
Authorization: Digest
username="Alice",realm="asterisk",nonce="522456f4",uri="sip:6001@10.24.18.124;transport=UDP",response="6d66dcad8c176aa3ef7bae
c7680d2445",algorithm=MD5
Event: presence
Allow-Events: presence, kpml
Content-Length: 0

<------------->
--- (17 headers 0 lines) ---
Creating new subscription
Sending to 10.24.17.254:37509 (no NAT)
Found peer 'Alice' for 'Alice' from 10.24.17.254:37509
Looking for 6001 in default (domain 10.24.18.124)
Scheduling destruction of SIP dialog 'ZjE2ZDAwYThiOTA2MzYxOWEwNTEwMjc1ZGIxNTk3NDU.' in 1810000 ms (Method: SUBSCRIBE)

<--- Transmitting (no NAT) to 10.24.17.254:37509 --->
SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.24.17.254:37509;branch=z9hG4bK-d8754z-c6908de6f0126edf-1---d8754z-;received=10.24.17.254
From: <sip:Alice@10.24.18.124;transport=UDP>;tag=f51e9632
To: <sip:6001@10.24.18.124;transport=UDP>;tag=as46a6e039
Call-ID: ZjE2ZDAwYThiOTA2MzYxOWEwNTEwMjc1ZGIxNTk3NDU.
CSeq: 2 SUBSCRIBE
Server: Asterisk PBX SVN-branch-12-r413487
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, REFER, SUBSCRIBE, NOTIFY, INFO, PUBLISH, MESSAGE
Supported: replaces, timer
Expires: 1800
Contact: <sip:6001@10.24.18.124:5060>;expires=1800
Content-Length: 0


<------------>
set_destination: Parsing <sip:Alice@10.24.17.254:37509;transport=UDP> for address/port to send to
set_destination: set destination to 10.24.17.254:37509
Reliably Transmitting (no NAT) to 10.24.17.254:37509:
NOTIFY sip:Alice@10.24.17.254:37509;transport=UDP SIP/2.0
Via: SIP/2.0/UDP 10.24.18.124:5060;branch=z9hG4bK14aacddc
Max-Forwards: 70
From: <sip:6001@10.24.18.124;transport=UDP>;tag=as46a6e039
To: <sip:Alice@10.24.18.124;transport=UDP>;tag=f51e9632
Contact: <sip:6001@10.24.18.124:5060>
Call-ID: ZjE2ZDAwYThiOTA2MzYxOWEwNTEwMjc1ZGIxNTk3NDU.
CSeq: 102 NOTIFY
User-Agent: Asterisk PBX SVN-branch-12-r413487
Subscription-State: active
Event: presence
Content-Type: application/pidf+xml
Content-Length: 530

<?xml version="1.0" encoding="ISO-8859-1"?>
<presence xmlns="urn:ietf:params:xml:ns:pidf"
xmlns:pp="urn:ietf:params:xml:ns:pidf:person"
xmlns:es="urn:ietf:params:xml:ns:pidf:rpid:status:rpid-status"
xmlns:ep="urn:ietf:params:xml:ns:pidf:rpid:rpid-person"
entity="sip:Alice@10.24.18.124">
<pp:person><status>
<ep:activities><ep:away/></ep:activities>
</status></pp:person>
<note>Unavailable</note>
<tuple id="6001">
<contact priority="1">sip:6001@10.24.18.124</contact>
<status><basic>closed</basic></status>
</tuple>
</presence>

---

<--- SIP read from UDP:10.24.17.254:37509 --->
SIP/2.0 200 OK
Via: SIP/2.0/UDP 10.24.18.124:5060;branch=z9hG4bK14aacddc
Contact: <sip:Alice@10.24.17.254:37509;transport=UDP>
To: <sip:Alice@10.24.18.124;transport=UDP>;tag=f51e9632
From: <sip:6001@10.24.18.124;transport=UDP>;tag=as46a6e039
Call-ID: ZjE2ZDAwYThiOTA2MzYxOWEwNTEwMjc1ZGIxNTk3NDU.
CSeq: 102 NOTIFY
User-Agent: Z 3.2.21357 r21103
Content-Length: 0

<------------->
--- (9 headers 0 lines) ---
```

Once the subscription has taken place, there is a command to list them. "sip show subscriptions"

```
*CLI> sip show subscriptions
Peer             User             Call ID          Extension        Last state     Type            Mailbox    Expiry
10.24.17.254     Alice            ZjE2ZDAwYThiOTA  6001@default     Unavailable    pidf+xml        <none>     001800
1 active SIP subscription
```

From this point onward, Asterisk should send out a SIP NOTIFY to the Alice peer whenever state changes for any of the devices mapped to the hint 6001. Alice's phone should then reflect that state on its display.

Configuring res_pjsip

Overview

This page and its sub-pages are intended to help an administrator configure the new SIP resources and channel driver included with Asterisk 12. The channel driver itself being chan_pjsip which depends on res_pjsip and its many associated modules. The **res_pjsip** module handles configuration, so we'll mostly speak in terms of configuring res_pjsip.

A variety of reference content is provided in the following sub-pages.

- If you are moving from the old channel driver, then look at Migrating from chan_sip to res_pjsip.
- For basic config examples look at res_pjsip Configuration Examples.
- For detailed explanation of the res_pjsip config file go to PJSIP Configuration Sections and Relationships.
- You can also find info on Dialing PJSIP Channels.
- Maybe you're migrating to IPv6 and need to learn about Configuring res_pjsip for IPv6

Before You Configure

This page assumes certain knowledge, or that you have completed a few prerequisites.

- You have installed pjproject, a dependency for res_pjsip.
- You have Installed Asterisk including the res_pjsip and chan_pjsip modules and their dependencies.
- You understand basic Asterisk concepts. Including the role of extensions.conf (dialplan) in your overall Asterisk configuration.

Quick Start

If you like to figure out things as you go; here's a few quick steps to get you started.

- Understand that res_pjsip is configured through pjsip.conf. This is where you'll be configuring everything related to your inbound or outbound SIP accounts and endpoints.
- Look at the res_pjsip Configuration Examples section. Grab the example most appropriate to your goal and use that to replace your pjsip.conf.
- Reference documentation for all configuration parameters is available on the wiki:
  - Core res_pjsip configuration options
  - Configuration options for ACLs in res_pjsip_acl
  - Configuration options for outbound registration, provided by res_pjsip_outbound_registration
  - Configuration options for endpoint identification by IP address, provided by res_pjsip_endpoint_identifier_ip
- You'll need to tweak details in pjsip.conf and on your SIP device (for example IP addresses and authentication credentials) to get it working with Asterisk.
  Refer back to the config documentation on the wiki or the sample pjsip.conf if you get confused.

PJSIP Configuration Sections and Relationships

Configuration Section Format

pjsip.conf is a flat text file composed of **sections** like most configuration files used with Asterisk. Each **section** defines configuration for a **configuration object** within res_pjsip or an associated module.

Sections are identified by **names in square brackets**. (see SectionName below)

Each section has one or more **configuration options** that can be assigned a value by using an **equal sign** followed by a value. (see ConfigOption and Value below)These options and values are the configuration for a particular component of functionality provided by the configuration object's respective Asterisk modules.

Every section will have a **type** option that defines what kind of section is being configured. You'll see that in every example config section below.

Syntax for res_sip config objects

```
[ SectionName ]
ConfigOption = Value
ConfigOption = Value
```

On this Page

Config Section Help and Defaults

Reference documentation for all configuration parameters is available on the wiki:

- Core res_pjsip configuration options
- Configuration options for ACLs in res_pjsip_acl
- Configuration options for outbound registration, provided by res_pjsip_outbound_registration
- Configuration options for endpoint identification by IP address, provided by res_pjsip_endpoint_identifier_ip

The same documentation is available at the Asterisk CLI as well. You can use "config show help <res_pjsip module name> <configobject> <configoption>" to get help on a particular option. That help will typically describe the default value for an option as well.

Defaults: For many config options, it's very helpful to understand their default behavior. For example, for the endpoint section "transport=" option, if no value is assigned then Asterisk will *DEFAULT* to the first configured transport in pjsip.conf which is valid for the URI we are trying to contact.

Section Names

In most cases, you can name a section whatever makes sense to you. For example you might name a transport [transport-udp-nat] to help you remember how that section is being used.

However, in some cases, (endpoint and aor types) the section name has a relationship to its function. In the case of endpoint and aor their names must match the user portion of the SIP URI in the "From" header for inbound SIP requests. The exception to that rule is if you have an identify section configured for that endpoint. In that case the inbound request would be matched by IP instead of against the user in the "From" header.

Section Types

Below is a brief description of each section type and an example showing configuration of that section only. The module providing the configuration object related to the section is listed in parentheses next to each section name.

There are dozens of config options for some of the sections, but the examples below are very minimal for the sake of simplicity.

ENDPOINT

(provided by module: res_pjsip)

Endpoint configuration provides numerous options relating to core SIP functionality and ties to other sections such as auth, aor and transport. You can't contact an endpoint without associating one or more AoR sections. An endpoint is essentially a profile for the configuration of a SIP endpoint such as a phone or remote server.

EXAMPLE BASIC CONFIGURATION

```
[6001]
type=endpoint
context=default
disallow=all
allow=ulaw
transport=simpletrans
auth=auth6001
aors=6001
```

If you want to define the Caller Id this endpoint should use, then add something like the following:

```
trust_id_outbound=yes
callerid=Spaceman Spiff <6001>
```

TRANSPORT

(provided by module: res_pjsip)

Configure how res_pjsip will operate at the transport layer. For example, it supports configuration options for protocols such as TCP, UDP or WebSockets and encryption methods like TLS/SSL.

You can setup multiple transport sections and other sections (such as endpoints) could each use the same transport, or a unique one. However, there are a couple caveats for creating multiple transports:

- They cannot share the same IP+port or IP+protocol combination. That is, each transport that binds to the same IP as another must use a different port or protocol.
- PJSIP does not allow multiple TCP or TLS transports of the same IP version (IPv4 or IPv6).

> **Reloading Config:** Configuration for transport type sections can't be reloaded during run-time without a full module unload and load. You'll effectively need to restart Asterisk completely for your transport changes to take effect.

EXAMPLE BASIC CONFIGURATION

A basic UDP transport bound to all interfaces

```
[simpletrans]
type=transport
protocol=udp
bind=0.0.0.0
```

Or a TLS transport, with many possible options and parameters:

```
[simpletrans]
type=transport
protocol=tls
bind=0.0.0.0
;various TLS specific options below:
cert_file=
priv_key_file=
ca_list_file=
cipher=
method=
```

AUTH

(provided by module: res_pjsip)

Authentication sections hold the options and credentials related to inbound or outbound authentication. You'll associate other sections such as endpoints or registrations to this one. Multiple endpoints or registrations can use a single auth config if needed.

EXAMPLE BASIC CONFIGURATION

An example with username and password authentication

```
[auth6001]
type=auth
auth_type=userpass
password=6001
username=6001
```

And then an example with MD5 authentication

```
[auth6001]
type=auth
auth_type=md5
md5_cred=51e63a3da6425a39aecc045ec45f1ae8
username=6001
```

AOR

(provided by module: res_pjsip)

A primary feature of AOR objects (Address of Record) is to tell Asterisk where an endpoint can be contacted. Without an associated AOR section, an endpoint cannot be contacted. AOR objects also store associations to mailboxes for MWI requests and other data that might relate to the whole group of contacts such as expiration and qualify settings.

When Asterisk receives an inbound registration, it'll look to match against available AORs.

Registrations: The name of the AOR section must match the user portion of the SIP URI in the "To:" header of the inbound SIP registration. That will usually be the "user name" set in your hard or soft phones configuration.

EXAMPLE BASIC CONFIGURATION

First, we have a configuration where you are expecting the SIP User Agent (likely a phone) to register against the AOR. In this case, the contact objects will be created automatically. We limit the maximum contact creation to 1. We could do 10 if we wanted up to 10 SIP User Agents to be able to register against it.

```
[6001]
type=aor
max_contacts=1
```

Second, we have a configuration where you are **not** expecting the SIP User Agent to register against the AOR. In this case, you can assign contacts manually as follows. We don't have to worry about max_contacts since that option only affects the maximum allowed contacts to be created through external interaction, like registration.

```
[6001]
type=aor
contact=sip:6001@192.0.2.1:5060
```

Third, it's useful to note that you could define only the domain and omit the user portion of the SIP URI if you wanted. Then you could define the **user** portion dynamically in your dialplan when calling the Dial application. You'll likely do this when building an AOR/Endpoint combo to use for dialing out to an ITSP. For example: "Dial(PJSIP/${EXTEN}@mytrunk)"

```
[mytrunk]
type=aor
contact=sip:203.0.113.1:5060
```

REGISTRATION

(provided by module: res_pjsip_outbound_registration)

The registration section contains information about an outbound registration. You'll use this when setting up a registration to another system whether it's local or a trunk from your ITSP.

EXAMPLE BASIC CONFIGURATION

This example shows you how you might configure registration and outbound authentication against another Asterisk system, where the other system is using the older chan_sip peer setup.

This example is just the registration itself. You'll of course need the associated transport and auth sections. Plus, if you want to receive calls from the far end (who now knows where to send calls, thanks to your registration!) then you'll need endpoint, AOR and possibly identify sections setup to match inbound calls to a context in your dialplan.

```
[mytrunk]
type=registration
transport=simpletrans
outbound_auth=mytrunk
server_uri=sip:myaccountname@203.0.113.1:5060
client_uri=sip:myaccountname@192.0.2.1:5060
retry_interval=60
```

And an example that may work with a SIP trunking provider

```
[mytrunk]
type=registration
transport=simpletrans
outbound_auth=mytrunk
server_uri=sip:sip.example.com
client_uri=sip:1234567890@sip.example.com
retry_interval=60
```

What if you don't need to authenticate? You can simply omit the **outbound_auth** option.

DOMAIN_ALIAS

(provided by module: res_pjsip)

Allows you to specify an alias for a domain. If the domain on a session is not found to match an AoR then this object is used to see if we have an alias for the AoR to which the endpoint is binding. This sections name as defined in configuration should be the domain alias and a config option (domain=) is provided to specify the domain to be aliased.

EXAMPLE BASIC CONFIGURATION

```
[example2.com]
type=domain_alias
domain=example.com
```

ACL

(provided by module: res_pjsip_acl)

The ACL module used by 'res_pjsip'. This module is independent of 'endpoints' and operates on all inbound SIP communication using res_pjsip. Features such as an Access Control List, as defined in the configuration section itself, or as defined in **acl.conf**. ACL's can be defined specifically for source IP addresses, or IP addresses within the contact header of SIP traffic.

EXAMPLE BASIC CONFIGURATION

A configuration pulling from the acl.conf file:

```
[acl]
type=acl
acl=example_named_acl1
```

A configuration defined in the object itself:

```
[acl]
type=acl
deny=0.0.0.0/0.0.0.0
permit=209.16.236.0
permit=209.16.236.1
```

A configuration where we are restricting based on contact headers instead of IP addresses.

```
[acl]
type=acl
contactdeny=0.0.0.0/0.0.0.0
contactpermit=209.16.236.0
contactpermit=209.16.236.1
```

All of these configurations can be combined.

IDENTIFY

(provided by module: res_pjsip_endpoint_identifier_ip)

Controls how the res_pjsip_endpoint_identifier_ip module determines what endpoint an incoming packet is from. If you don't have an identify section defined, or else you have res_pjsip_endpoint_**identifier_ip** loading **after** res_pjsip_endpoint_**identifier_user**, then res_pjsip_endpoint_**identifier_user** will identify inbound traffic by pulling the user from the "From:" SIP header in the packet. Basically the module load order, and your configuration will both determine whether you identify by IP or by user.

EXAMPLE BASIC CONFIGURATION

Its use is quite straightforward. With this configuration if Asterisk sees inbound traffic from 203.0.113.1 then it will match that to Endpoint 6001.

```
[6001]
type=identify
endpoint=6001
match=203.0.113.1
```

CONTACT

(provided by module: res_pjsip)

The contact config object effectively acts as an alias for a SIP URIs and holds information about an inbound registrations. Contact objects can be associated with an individual SIP User Agent and contain a few config options related to the connection. Contacts are created automatically upon registration to an AOR, or can be created manually by using the "contact=" config option in an AOR section. Manually configuring a CONTACT config object itself is outside the scope of this "getting started" style document.

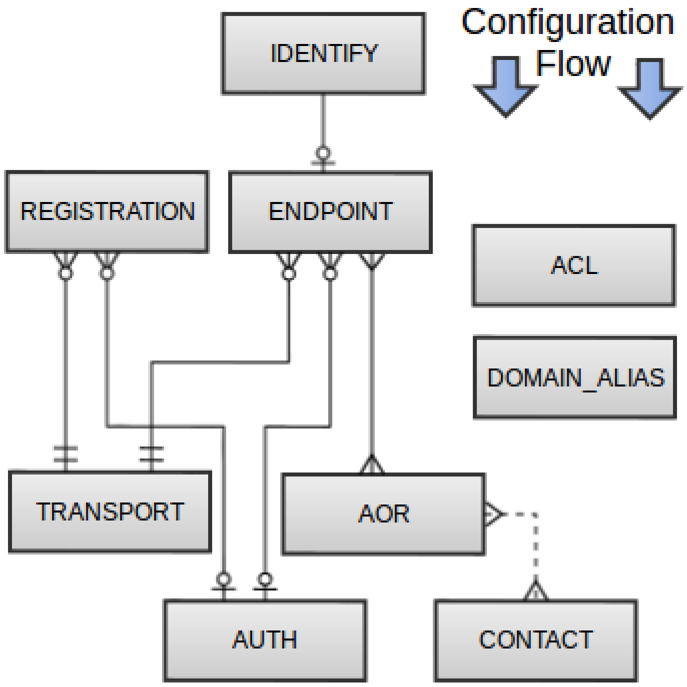
Relationships of Configuration Objects in pjsip.conf

Now that you understand the various configuration sections related to each config object, lets look at how they interrelate.

You'll see that the new SIP implementation within Asterisk is extremely flexible due to its modular design. A diagram will help you to visualize the relationships between the various configuration objects. The following entity relationship diagram covers only the configuration relationships between the objects. For example if an **endpoint** object requires authorization for registration of a SIP device, then you may associate a single **auth** object with the endpoint object. Though many endpoints could use the same or different auth objects.

Configuration Flow: This lets you know which direction the objects are associated to other objects. e.g. The identify config section has an option "endpoint=" which allows you to associate it with an endpoint object.

Entity Relationships	Relationship Descriptions



ENDPOINT

- Many ENDPOINTs can be associated with many AORs
- Zero to many ENDPOINTs can be associated with zero to one AUTHs
- Zero to many ENDPOINTs can be associated with at least one TRANSPORT
- Zero to one ENDPOINTs can be associated with an IDENTIFY

REGISTRATION

- Zero to many REGISTRATIONs can be associated with zero to one AUTHs
- Zero to many REGISTRATIONs can be associated with at least one TRANSPORT

AOR

- Many ENDPOINTs can be associated with many AORs
- Many AORs can be associated with many CONTACTs

CONTACT

- Many CONTACTs can be associated with many AORs

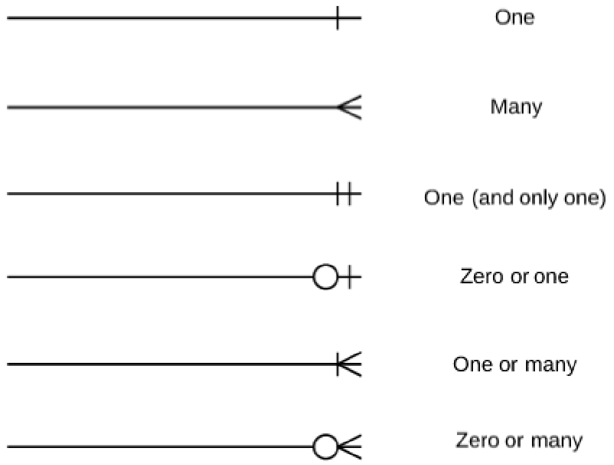
IDENTIFY

- Zero to One ENDPOINTs can be associated with an IDENTIFY object

ACL, DOMAIN_ALIAS

- These objects don't have a direct configuration relationship to the other objects.

Unfamiliar with ERD? Click here to see a key...



res_pjsip Configuration Examples

Below are some sample configurations to demonstrate various scenarios with complete pjsip.conf files. To see examples side by side with old chan_sip config head to Migrating from chan_sip to res_pjsip. Explanations of the config sections found in each example can be found in PJSIP Configuration Sections and Relationships.

A tutorial on secure and encrypted calling is located in the Secure Calling section of the wiki.

An endpoint with a single SIP phone with inbound registration to Asterisk

EXAMPLE CONFIGURATION

```
;===============TRANSPORT

[simpletrans]
type=transport
protocol=udp
bind=0.0.0.0

;===============EXTENSION 6001

[6001]
type=endpoint
context=internal
disallow=all
allow=ulaw
auth=auth6001
aors=6001

[auth6001]
type=auth
auth_type=userpass
password=6001
username=6001

[6001]
type=aor
max_contacts=1
```

- auth= is used for the endpoint as opposed to outbound_auth= since we want to allow inbound registration for this endpoint
- max_contacts= is set to something non-zero as we want to allow contacts to be created through registration

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A SIP trunk to your service provider, including outbound registration

EXAMPLE CONFIGURATION

Trunks are a little tricky since many providers have unique requirements. Your final configuration may differ from what you see here.

```
;==============TRANSPORTS

[simpletrans]
type=transport
protocol=udp
bind=0.0.0.0

;===============TRUNK

[mytrunk]
type=registration
outbound_auth=mytrunk
server_uri=sip:sip.example.com
client_uri=sip:1234567890@sip.example.com
retry_interval=60

[mytrunk]
type=auth
auth_type=userpass
password=1234567890
username=1234567890

[mytrunk]
type=aor
contact=sip:sip.example.com:5060

[mytrunk]
type=endpoint
context=from-external
disallow=all
allow=ulaw
outbound_auth=mytrunk
aors=mytrunk

[mytrunk]
type=identify
endpoint=mytrunk
match=sip.example.com
```

- "contact=sip:203.0.113.1:5060", we don't define the user portion statically since we'll set that dynamically in dialplan when we call the Dial application.
  See the dialing examples in the section "Dialing using chan_pjsip" for more.
- "outbound_auth=mytrunk", we use "outbound_auth" instead of "auth" since the provider isn't typically going to authenticate with us when calling, but we will probably
  have to authenticate when calling through them.
- We use an identify object to map all traffic from the provider's IP as traffic to that endpoint since the user portion of their From: header may vary with each call.
- This example assumes that sip.example.com resolves to 203.0.113.1

You can specify the transport type by appending it to the server_uri and client_uri parameters. e.g.:

```
[mytrunk]
type=registration
outbound_auth=mytrunk
server_uri=sip:sip.example.com\;transport=tcp
client_uri=sip:1234567890@sip.example.com\;transport=tcp
retry_interval=60
```

Multiple endpoints with phones registering to Asterisk, using templates

EXAMPLE CONFIGURATION

We want to show here that generally, with a large configuration you'll end up using templates to make configuration easier to handle when scaling. This avoids having redundant code in every similar section that you create.

```
 ;===============TRANSPORT

[simpletrans]
type=transport
protocol=udp
bind=0.0.0.0

;===============ENDPOINT TEMPLATES

[endpoint-basic](!)
type=endpoint
context=internal
disallow=all
allow=ulaw

[auth-userpass](!)
type=auth
auth_type=userpass

[aor-single-reg](!)
type=aor
max_contacts=1

;===============EXTENSION 6001

[6001](endpoint-basic)
auth=auth6001
aors=6001

[auth6001](auth-userpass)
password=6001
username=6001

[6001](aor-single-reg)

;===============EXTENSION 6002

[6002](endpoint-basic)
auth=auth6002
aors=6002

[auth6002](auth-userpass)
password=6002
username=6002

[6002](aor-single-reg)

;===============EXTENSION 6003

[6003](endpoint-basic)
auth=auth6003
aors=6003

[auth6003](auth-userpass)
password=6003
username=6003

[6003](aor-single-reg)
```

Obviously the larger your configuration is, the more templates will benefit you. Here we just break apart the endpoints with templates, but you could do that with any config section that needs instances with variation, but where each may share common settings with their peers.

Migrating from chan_sip to res_pjsip

Overview

This page documents any useful tools, tips or examples on moving from the old chan_sip channel driver to the new chan_pjsip/res_pjsip added in Asterisk 12.

Configuration Conversion Script

Contained within a download of Asterisk, there is a Python script, sip_to_pjsip.py, found within the contrib/scripts/sip_to_pjsip subdirectory, that provides a basic conversion of a sip.conf config to a pjsip.conf config. It is not intended to work for every scenario or configuration; for basic configurations it should provide a good example of how to convert it over to pjsip.conf style config.

To insure that the script can read any #include'd files, run it from the /etc/asterisk directory or in another location with a copy of the sip.conf and any included files. The default input file is sip.conf, and the default output file is pjsip.conf. Any included files will also be converted, and written out with a pjsip_ prefix, unless changed with the --prefix=*xxx* option.

Command line usage

```
# /path/to/asterisk/source/contrib/scripts/sip_to_pjsip/sip_to_pjsip.py --help
Usage: sip_to_pjsip.py [options] [input-file [output-file]]
input-file defaults to 'sip.conf'
output-file defaults to 'pjsip.conf'
Options:
  -h, --help            show this help message and exit
  -p PREFIX, --prefix=PREFIX
                        output prefix for include files
```

Example of Use

```
# cd /etc/asterisk
# /path/to/asterisk/source/contrib/scripts/sip_to_pjsip/sip_to_pjsip.py
Reading sip.conf
Converting to PJSIP...
Writing pjsip.conf
```

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Side by Side Examples of sip.conf and pjsip.conf Configuration

These examples contain only the configuration required for sip.conf/pjsip.conf as the configuration for other files should be the same, excepting the Dial statements in your extensions.conf. Dialing with PJSIP is discussed in Dialing PJSIP Channels.

> It is important to know that PJSIP syntax and configuration format is stricter than the older chan_sip driver. When in doubt, try to follow the documentation exactly, avoid extra spaces or strange capitalization. Always check your logs for warnings or errors if you suspect something is wrong.

Example Endpoint Configuration

This examples shows the configuration required for:

- two SIP phones need to make calls to or through Asterisk, we also want to be able to call them from Asterisk
- for them to be identified as users (in the old chan_sip) or endpoints (in the new res_sip/chan_pjsip)
- both devices need to use username and password authentication
- 6001 is setup to allow registration to Asterisk, and 6002 is setup with a static host/contact

sip.conf	pjsip.conf

```
[general]
udpbindaddr=0.0.0.0

[6001]
type=friend
host=dynamic
disallow=all
allow=ulaw
context=internal
secret=1234

[6002]
type=friend
host=192.0.2.1
disallow=all
allow=ulaw
context=internal
secret=1234
```

```
[simpletrans]
type=transport
protocol=udp
bind=0.0.0.0

[6001]
type = endpoint
context = internal
disallow = all
allow = ulaw
aors = 6001
auth = auth6001

[6001]
type = aor
max_contacts = 1

[auth6001]
type=auth
auth_type=userpass
password=1234
username=6001

[6002]
type = endpoint
context = internal
disallow = all
allow = ulaw
aors = 6002
auth = auth6002

[6002]
type = aor
contact = sip:6002@192.0.2.1:5060

[auth6002]
type=auth
auth_type=userpass
password=1234
username=6001
```

Example SIP Trunk Configuration

This shows configuration for a SIP trunk as would typically be provided by an ITSP. That is registration to a remote server, authentication to it and a peer/endpoint setup to allow inbound calls from the provider.

- SIP provider requires registration to their server with a username of "myaccountname" and a password of "1234567890"
- SIP provider requires registration to their server at the address of 203.0.113.1:5060
- SIP provider requires outbound calls to their server at the same address of registration, plus using same authentication details.
- SIP provider will call your server with a user name of "mytrunk". Their traffic will only be coming from 203.0.113.1

sip.conf	pjsip.conf

```
[general]
udpbindaddr=0.0.0.0

register => myaccountname:1234567890@203.0.113.1:5060

[mytrunk]
type=friend
secret=1234567890
username=myaccountname
host=203.0.113.1
disallow=all
allow=ulaw
context=from-external
```

```
[simpletrans]
type=transport
protocol=udp
bind=0.0.0.0

[mytrunk]
type=registration
outbound_auth=mytrunk
server_uri=sip:myaccountname@20
client_uri=sip:myaccountname@20

[mytrunk]
type=auth
auth_type=userpass
password=1234567890
username=myaccountname

[mytrunk]
type=aor
contact=sip:203.0.113.1:5060

[mytrunk]
type=endpoint
context=from-external
disallow=all
allow=ulaw
outbound_auth=mytrunk
aors=mytrunk

[mytrunk]
type=identify
endpoint=mytrunk
match=203.0.113.1
```

Disabling res_pjsip and chan_pjsip

You may want to keep using chan_sip for a short time in Asterisk 12+ while you migrate to res_pjsip. In that case, it is best to disable res_pjsip unless you understand how to configure them both together.

There are several methods to disable or remove modules in Asterisk. Which method is best depends on your intent.

If you have built Asterisk with the PJSIP modules, but don't intend to use them at this moment, you might consider the following:

1. Edit the file **modules.conf** in your Asterisk configuration directory. (typically /etc/asterisk/)

   ```
   noload => res_pjsip.so
   noload => res_pjsip_pubsub.so
   noload => res_pjsip_session.so
   noload => chan_pjsip.so
   noload => res_pjsip_exten_state.so
   noload => res_pjsip_log_forwarder.so
   ```

   Having a noload for the above modules should (at the moment of writing this) prevent any PJSIP related modules from loading.
2. Restart Asterisk!

Other possibilities would be:

- Remove all PJSIP modules from the modules directory (often, /usr/lib/asterisk/modules)
- Remove the configuration file (pjsip.conf)
- Un-install and re-install Asterisk with no PJSIP related modules.

- If you are wanting to use chan_pjsip alongside chan_sip, you could change the port or bind interface of your chan_pjsip transport in pjsip.conf

Network Address Translation (NAT)

When configured with **chan_sip**, peers that are, relative to Asterisk, located behind a NAT are configured using the **nat** parameter. In versions 1.8 and greater of Asterisk, the following nat parameter options are available:

Value	Description
no	Do not perform NAT handling other than RFC 3581.
force_rport	When the rport parameter is not present, send responses to the source IP address and port anyway, as though the rport parameter was present
comedia	Send media to the address and port from which Asterisk received it, regardless of where SDP indicates that it should be sent
auto_force_rport	Automatically enable the sending of responses to the source IP address and port, as though rport were present, if Asterisk detects NAT. Default.
auto_comedia	Automatically send media to the port from which Asterisk received it, regardless of where SDP indicates that it should be sent, if Asterisk detects NAT.

Versions of Asterisk prior to 1.8 had less granularity for the nat parameter:

Value	Description
no	Do not perform NAT handling other than RFC 3581
yes	Send media to the port from which Asterisk received it, regardless of where SDP indicates that it should be sent; send responses to the source IP address and port as though rport were present; and rewrite the SIP Contact to the source address and port of the request so that subsequent requests go to that address and port.
never	Do not perform any NAT handling
route	Send media to the port from which Asterisk received it, regardless of where SDP indicates that it should be sent and rewrite the SIP Contact to the source address and port of the request so that subsequent requests go to that address and port.

In **chan_pjsip**, the **endpoint** options that control NAT behavior are:

- rtp_symmetric - Send media to the address and port from which Asterisk receives it, regardless of where SDP indicates that it should be sent
- force_rport - Send responses to the source IP address and port as though port were present, even if it's not
- rewrite_contact - Rewrite SIP Contact to the source address and port of the request so that subsequent requests go to that address and port.

Thus, the following are equivalent:

chan_sip (sip.conf)	chan_pjsip (pjsip.conf)

[mypeer1]
type=peer
nat=yes
;...
 	
[mypeer2]
type=peer
nat=no
;...
 	
[mypeer3]
type=peer
nat=never
;...
 	
[mypeer4]
type=peer
nat=route
;...
	[mypeer1]
	type=endpoint
	rtp_symmetric=yes
	force_rport=yes
	rewrite_contact=yes
	;...
	
	[mypeer2]
	type=endpoint
	rtp_symmetric=no
	force_rport=no
	rewrite_contact=no
	;...
	
	[mypeer3]
	type=endpoint
	rtp_symmetric=no
	force_rport=no
	rewrite_contact=no
	;...
	
	[mypeer4]
	type=endpoint
	rtp_symmetric=no
	force_rport=yes
	rewrite_contact=yes
	;...

Dialing PJSIP Channels

Dialing from dialplan

We are assuming you already know a little bit about the Dial application here. To see the full help for it, see "core show help application dial" on the Asterisk CLI, or see Application_Dial

Below we'll simply dial an endpoint using the chan_pjsip channel driver. This is really going to look at the AOR of the same name as the endpoint and start dialing the first contact associated.

```
exten => _6XXX,1,Dial(PJSIP/${EXTEN})
```

To dial all the contacts associated with the endpoint, use the PJSIP_DIAL_CONTACTS function. It evaluates to a list of contacts separated by &, which causes the Dial application to call them simultaneously.

```
exten => _6XXX,1,Dial(${PJSIP_DIAL_CONTACTS(${EXTEN})})
```

Heres how you would dial with an explicit SIP URI, user and domain, via an endpoint (in this case dialing out a trunk), but not using its associated AOR/contact objects.

```
exten => _9NXXNXXXXXX,1,Dial(PJSIP/mytrunk/sip:${EXTEN:1}@203.0.113.1:5060)
```

This uses a contact(and its domain) set in the AOR associated with the **mytrunk** endpoint, but still explicitly sets the user portion of the URI in the dial string. For the AOR's contact, you would define it in the AOR config without the user name.

```
exten => _9NXXNXXXXXX,1,Dial(PJSIP/${EXTEN:1}@mytrunk)
```

Configuring res_pjsip to work through NAT

Here we can show some examples of working configuration for Asterisk's SIP channel driver when Asterisk is behind NAT (Network Address Translation).

If you are migrating from chan_sip to chan_pjsip, then also read the NAT section in Migrating from chan_sip to res_pjsip for helpful tips.

Asterisk and Phones Connecting Through NAT to an ITSP

This example should apply for most simple NAT scenarios that meet the following criteria:

- Asterisk and the phones are on a private network.
- There is a router interfacing the private and public networks. Where the public network is the Internet.
- The router is performing Network Address Translation and Firewall functions.
- The router is configured for port-forwarding, where it is mapping the necessary ranges of SIP and RTP traffic to your internal Asterisk server.
  In this example the router is port-forwarding WAN inbound TCP/UDP 5060 and UDP 10000-20000 to LAN 192.0.2.10

This example was based on a configuration for the ITSP SIP.US and assuming you swap out the addresses and credentials for real ones, it should work for a SIP.US SIP account.

Devices Involved in the Example

Using RFC5737 documentation addresses

Device	IP in example
VOIP Phone(6001)	192.0.2.20
PC/Asterisk	192.0.2.10
Router	LAN: 192.0.2.1 WAN: 198.51.100.5

ITSP SIP gateway	203.0.113.1(gw1.example.com)
	203.0.113.2(gw2.example.com)

For the sake of a complete example and clarity, in this example we use the following fake details:

ITSP Account number: 1112223333

DID number provided by ITSP: 19998887777

pjsip.conf Configuration

We are assuming you have already read the Configuring res_pjsip page and have a basic understanding of Asterisk. For this NAT example, the important config options to note are **local_net**, **external_media_address** and **external_signaling_address** in the transport type section and **direct_media** in the endpoint section. The rest of the options may depend on your particular configuration, phone model, network settings, ITSP, etc. The key is to make sure you have those three options set appropriately.
local_net

This is the IP network that we want to consider our local network. For communication to addresses within this range, we won't apply any NAT-related settings, such as the external* options below.
external_media_address

This is the external IP address to use in RTP handling. When a request or response is sent out from Asterisk, if the destination of the message is outside the IP network defined in the option 'local_net', and the media address in the SDP is within the localnet network, then the media address in the SDP will be rewritten to the value defined for 'external_media_address'.
external_signaling_address

This is much like the external_media_address setting, but for SIP signaling instead of RTP media. The two external* options mentioned here should be set to the same address unless you separate your signaling and media to different addresses or servers.
direct_media

Determines whether media may flow directly between endpoints

Together these options make sure the far end knows where to send back SIP and RTP packets, and direct_media ensures Asterisk stays in the media path. This is important, because our Asterisk system has a private IP address that the ITSP cannot route to. We want to make sure the SIP and RTP traffic comes back to the WAN/Public internet address of our router. The sections prefixed with "sipus" are all configuration needed for inbound and outbound connectivity of the SIP trunk, and the sections named 6001 are all for the VOIP phone.

```
[transport-udp-nat]
type=transport
protocol=udp
bind=0.0.0.0
local_net=192.0.2.0/24
local_net=127.0.0.1/32
external_media_address=198.51.100.5
external_signaling_address=198.51.100.5

[sipus_reg]
type=registration
transport=transport-udp-nat
outbound_auth=sipus_auth
server_uri=sip:gw1.example.com
client_uri=sip:1112223333@gw1.example.com
contact_user=19998887777
retry_interval=60

[sipus_auth]
type=auth
auth_type=userpass
password=************
username=1112223333
realm=gw1.example.com

[sipus_endpoint]
type=endpoint
transport=transport-udp-nat
context=from-external
disallow=all
allow=ulaw
outbound_auth=sipus_auth
aors=sipus_aor
direct_media=no
from_domain=gw1.example.com

[sipus_aor]
type=aor
contact=sip:gw1.example.com
contact=sip:gw2.example.com

[sipus_identify]
type=identify
endpoint=sipus_endpoint
match=203.0.113.1
match=203.0.113.2

[6001]
type=endpoint
context=from-internal
disallow=all
allow=ulaw
transport=transport-udp-nat
auth=6001
aors=6001
direct_media=no

[6001]
type=auth
```

```
auth_type=userpass
password=*********
username=6001
```

```
[6001]
type=aor
max_contacts=2
```

For Remote Phones Behind NAT

In the above example we assumed the phone was on the same local network as Asterisk. Now, perhaps Asterisk is exposed on a public address, and instead your phones are remote and behind NAT, or maybe you have a double NAT scenario?

In these cases you will want to consider the below settings for the remote endpoints.
media_address

IP address used in SDP for media handling

At the time of SDP creation, the IP address defined here will be used as
the media address for individual streams in the SDP.
NOTE: Be aware that the 'external_media_address' option, set in Transport
configuration, can also affect the final media address used in the SDP.
rtp_symmetric

Enforce that RTP must be symmetric. Send RTP back to the same address/port we received it from.
force_rport

Force RFC3581 compliant behavior even when no rport parameter exists. Basically always send SIP responses back to the same port we received SIP requests from.
direct_media

Determines whether media may flow directly between endpoints.
rewrite_contact

Determine whether SIP requests will be sent to the source IP address and port, instead of the address provided by the endpoint.

Clients Supporting ICE,STUN,TURN

This is really relevant to media, so look to the section here for basic information on enabling this support and we'll add relevant examples later.

Setting up PJSIP Realtime

Overview

This tutorial describes the configuration of Asterisk's PJSIP channel driver with the "realtime" database storage backend. The realtime interface allows storing much of the configuration of PJSIP, such as endpoints, auths, aors and more, in a database, as opposed to the normal flat-file storage of pjsip.conf.

Installing Dependencies

For the purposes of this tutorial, we will assume a base Ubuntu 12.0.4.3 x86_64 server installation, with the OpenSSH server and LAMP server options, and that Asterisk will use its ODBC connector to reach a back-end MySQL database.

Beyond the normal packages needed to install Asterisk 12 on such a server (build-essential, libncurses5-dev, uuid-dev, libjansson-dev, libxml2-dev, libsqlite3-dev) as well as the Installation of pjproject, you will need to install the following packages:

- unixodbc and unixodbc-dev
  - ODBC and the development packages for building against ODBC
- libmyodbc
  - The ODBC to MySQL interface package
- python-dev and python-pip
  - The Python development package and the pip package to allow installation of Alembic
- python-mysqldb
  - The Python interface to MySQL, which will be used by Alembic to generate the database tables

So, from the CLI, perform:

```
# apt-get install unixodbc unixodbc-dev libmyodbc python-dev python-pip python-mysqldb
```

Once these packages are installed, check your Asterisk installation's **make menuconfig** tool to make sure that the **res_config_odbc** and **res_odbc** resource modules, as well as the **res_pjsip_xxx** modules are selected for installation. If they are, then go through the normal Asterisk installation process: **./configure; make; make install**

And, if this is your first installation of Asterisk, be sure to install the sample files: **make samples**

Creating the MySQL Database

Use the **mysqladmin** tool to create the database that we'll use to store the configuration. From the Linux CLI, perform:

```
# mysqladmin -u root -p create asterisk
```

This will prompt you for your MySQL database password and then create a database named **asterisk** that we'll use to store our PJSIP configuration.

Installing and Using Alembic

Alembic is a full database migration tool, with support for upgrading the schemas of existing databases, versioning of schemas, creation of new tables and databases, and a whole lot more. A good guide on using Alembic with Asterisk can be found on the Managing Realtime Databases with Alembic wiki page. A shorter discussion of the steps necessary to prep your database will follow.

First, install Alembic:

```
# pip install alembic
```

Then, move to the Asterisk source directory containing the Alembic scripts:

```
# cd contrib/ast-db-manage/
```

Next, edit the **config.ini.sample** file and change the **sqlalchemy.url** option, e.g.

```
sqlalchemy.url = mysql://root:password@localhost/asterisk
```

such that the URL matches the username and password required to access your database.

Then rename the config.ini.sample file to config.ini

```
# cp config.ini.sample config.ini
```

Finally, use Alembic to setup the database tables:

```
# alembic -c config.ini upgrade head
```

You'll see something similar to:

```
# alembic -c config.ini upgrade head
INFO  [alembic.migration] Context impl MySQLImpl.
INFO  [alembic.migration] Will assume non-transactional DDL.
INFO  [alembic.migration] Running upgrade None -> 4da0c5f79a9c, Create tables
INFO  [alembic.migration] Running upgrade 4da0c5f79a9c -> 43956d550a44, Add tables for pjsip
#
```

You can then connect to MySQL to see that the tables were created:

```
# mysql -u root -p -D asterisk

mysql> show tables;
+--------------------+
| Tables_in_asterisk |
+--------------------+
| alembic_version    |
| iaxfriends         |
| meetme             |
| musiconhold        |
| ps_aors            |
| ps_auths           |
| ps_contacts        |
| ps_domain_aliases  |
| ps_endpoint_id_ips |
| ps_endpoints       |
| sippeers           |
| voicemail          |
+--------------------+
12 rows in set (0.00 sec)
mysql> quit
```

Configuring ODBC

Now that we have our MySQL database created and populated, we'll need to setup ODBC and Asterisk's ODBC resource to access the database. First, we'll tell ODBC how to connect to MySQL. To do this, we'll edit the **/etc/odbcinst.ini** configuration file. Your file should look something like:

/etc/odbcinst.ini

```
[MySQL]
Description = ODBC for MySQL
Driver = /usr/lib/x86_64-linux-gnu/odbc/libmyodbc.so
Setup = /usr/lib/x86_64-linux-gnu/odbc/libodbcmyS.so
UsageCount = 2
```

Next, we'll tell ODBC **which** MySQL database to use. To do this, we'll edit the **/etc/odbc.ini** configuration file and create a database handle called **asterisk** . Your file should look something like:

/etc/odbc.ini

```
[asterisk]
Driver = MySQL
Description = MySQL connection to ‘asterisk’ database
Server = localhost
Port = 3306
Database = asterisk
UserName = root
Password = password
Socket = /var/run/mysqld/mysqld.sock
```

Take care to use your database access UserName and Password, and not necessarily what's defined in this example.

Now, we need to configure Asterisk's ODBC resource, res_odbc, to connect to the ODBC **asterisk** database handle that we just created. res_odbc is configured using the **/etc/asterisk/res_odbc.conf** configuration file. There, you'll want:

/etc/asterisk/res_odbc.conf

```
[asterisk]
enabled => yes
dsn => asterisk
username => root
password => password
pre-connect => yes
```

Again, take care to use the proper username and password.

Now, you can start Asterisk and you can check its connection to your "asterisk" MySQL database using the "asterisk" res_odbc connector to ODBC. You can do this by executing "odbc show" from the Asterisk CLI. If everything went well, you'll see:

```
# asterisk -vvvvc
*CLI> odbc show

ODBC DSN Settings
-----------------
 Name: asterisk
 DSN:    asterisk
 Last connection attempt: 1969-12-31 18:00:00
 Pooled: No
 Connected: Yes
*CLI>
```

Connecting PJSIP Sorcery to the Realtime Database

The PJSIP stack uses a new data abstraction layer in Asterisk called **sorcery**. Sorcery lets a user build a hierarchical layer of data sources for Asterisk to use when it retrieves, updates, creates, or destroys data that it interacts with. This tutorial focuses on getting PJSIP's configuration stored in a realtime back-end; the rest of the details of sorcery are beyond the scope of this page.

PJSIP bases its configuration on types of objects. For more information about these types of objects, please refer to the Configuring res_pjsip wiki page. In this case, we have a total of five objects we need to configure in Sorcery:

- endpoint
- auth
- aor
- domain
- identify

We'll also configure the contact object, though we don't need it for this example.

Sorcery is configured using the **/etc/asterisk/sorcery.conf** configuration file. So, we need to add the following lines to the file:

/etc/asterisk/sorcery.conf

```
[res_pjsip] ; Realtime PJSIP configuration wizard
endpoint=realtime,ps_endpoints
auth=realtime,ps_auths
aor=realtime,ps_aors
domain_alias=realtime,ps_domain_aliases
contact=realtime,ps_contacts

[res_pjsip_endpoint_identifier_ip]
identify=realtime,ps_endpoint_id_ips
```

The items use the following nomenclature:

```
{object_type} = {sorcery_wizard_name},{wizard_arguments}
```

In our case, the `sorcery_wizard_name` is **realtime**, and the **wizard_arguments** are the name of the database connector ("asterisk") to associate with our object types. Note that the "identify" object is separated from the rest of the configuration objects. This is because this object type is provided by an optional module (res_pjsip_endpoint_idenfifier_ip.so) and not the main PJSIP module (res_pjsip.so).

Optionally configuring sorcery for realtime and non-realtime data sources

If you want to configure **both realtime and static configuration** file lookups for PJSIP then you need to add additional lines to the sorcery config.

For example if you want to read **endpoints** from both realtime and static configuration:

```
endpoint=realtime,ps_endpoints
endpoint=config,pjsip.conf,criteria=type=endpoint
```

You can swap the order to control which data source is read first.

Realtime Configuration

Since we've associated the PJSIP objects with database connector types, we now need to tell Asterisk to use a database backend with the object types, and not just the flat pjsip.conf file. To do this, we modify the **/etc/asterisk/extconfig.conf** configuration file to provide these connections.

Open extconfig.conf (/etc/asterisk/extconfig.conf) and add the following lines to the 'settings' configuration section

/etc/asterisk/extconfig.conf

```
[settings]
ps_endpoints => odbc,asterisk
ps_auths => odbc,asterisk
ps_aors => odbc,asterisk
ps_domain_aliases => odbc,asterisk
ps_endpoint_id_ips => odbc,asterisk
ps_contacts => odbc,asterisk
```

ⓘ Other tables allowed but not demonstrated in this tutorial: ps_systems, ps_globals, ps_transports, and ps_registrations.

At this point, Asterisk is nearly ready to use the tables created by alembic with PJSIP to configure endpoints, authorization, AORs, domain aliases, and endpoint identifiers.

> **A warning for adventurous types:**
> Sorcery.conf allows you to try to configure other PJSIP objects such as transport using realtime and it currently won't stop you from doing so. However, some of these object types should not be used with realtime and this can lead to errant behavior.

Asterisk Startup Configuration

Now, we need to configure Asterisk to load its ODBC driver at an early stage of startup, so that it's available when any other modules might need to take advantage of it. Also, we're going to prevent the old chan_sip channel driver from loading, since we're only worried about PJSIP.

To do this, edit the **/etc/asterisk/modules.conf** configuration file. In the **[modules]** section, add the following lines:

/etc/asterisk/modules.conf

```
preload => res_odbc.so
preload => res_config_odbc.so
noload => chan_sip.so
```

Asterisk PJSIP configuration

Next, we need to configure a transport in **/etc/asterisk/pjsip.conf**. PJSIP transport object types are not stored in realtime as unexpected results can occur. So, edit it and add the following lines:

/etc/asterisk/pjsip.conf

```
[transport-udp]
type=transport
protocol=udp
bind=0.0.0.0
```

Here, we created a transport called **transport-udp** that we'll reference in the next section.

Endpoint Population

Now, we need to create our endpoints inside of the database. For this example, we'll create two peers, 101 and 102, that register using the totally insecure passwords "101" and "102" respectively. Here, we'll be populating data directly into the database using the MySQL interactive tool.

```
# mysql -u root -p -D asterisk;
mysql> insert into ps_aors (id, max_contacts) values (101, 1);
mysql> insert into ps_aors (id, max_contacts) values (102, 1);
mysql> insert into ps_auths (id, auth_type, password, username) values (101, 'userpass', 101, 101);
mysql> insert into ps_auths (id, auth_type, password, username) values (102, 'userpass', 102, 102);
mysql> insert into ps_endpoints (id, transport, aors, auth, context, disallow, allow, direct_media) values (101, 'transport-udp',
'101', '101', 'testing', 'all', 'g722', 'no');
mysql> insert into ps_endpoints (id, transport, aors, auth, context, disallow, allow, direct_media) values (102, 'transport-udp',
'102', '102', 'testing', 'all', 'g722', 'no');
mysql> quit;
```

In this example, we first created an **aor** for each peer, one called **101** and the other **102**.

Next, we created an **auth** for each peer with a userpass of **101** and **102**, respectively.

Then, we created two endpoints, **101** and **102**, each referencing the appropriate **auth** and **aor**, we selected the G.722 codec and we forced media to route inside of Asterisk (not the default behavior of Asterisk).

Now, you can start Asterisk and you can check to see if it's finding your PJSIP endpoints in the database. You can do this by executing "pjsip show endpoints" from the Asterisk CLI. If everything went well, you'll see:

```
# asterisk -vvvvc
*CLI> pjsip show endpoints
Endpoints:
101
102
*CLI>
```

A Little Dialplan

Now that we have our PJSIP endpoints stored in our MySQL database, let's add a little dialplan so that they can call each other. To do this, edit Asterisk's **/etc/asterisk/extensions.conf** file and add the following lines to the end:

/etc/asterisk/extensions.conf

```
[testing]
exten => _1XX,1,NoOp()
same => n,Dial(PJSIP/${EXTEN})
```

Or to dial multiple AOR contacts at the same time, use the PJSIP_DIAL_CONTACTS function:

/etc/asterisk/extensions.conf

```
[testing]
exten => _1XX,1,NoOp()
same => n,Dial(${PJSIP_DIAL_CONTACTS(${EXTEN})})
```

Reserved Characters

Realtime uses the semicolon (;) as a delimiter for multiple entries. It must be replaced with "^3B" to prevent the data from being interpreted as multiple entries.

> "^3B" is the corresponding byte value of the semicolon character in ASCII, represented as a pair of hexadecimal digits, preceded by a caret (^) acting as the escape character.

For example, this outbound_proxy parameter

```
sip:10.30.100.28:5060;lr
```

should be stored in the database as

```
sip:10.30.100.28:5060^3Blr
```

Conclusion

Now, start Asterisk back up, or reload it using **core reload** from the Asterisk CLI, register your two SIP phones using the 101/101 and 102/102 credentials, and make a call.

Exchanging Device and Mailbox State Using PJSIP

Background

Asterisk has permitted the exchange of device and mailbox state for many versions. This has normally been accomplished using the res_xmpp module for instances across networks or using res_corosync for instances on the same network. This has required, in some cases, an extreme amount of work to setup. In the case of res_xmpp this also adds another point of failure for the exchange in the form of the XMPP server itself. The res_pjsip_publish_asterisk module on the other hand does not suffer from this.

Operation

The res_pjsip_publish_asterisk module establishes an optionally bidirectional or unidirectional relationship between Asterisk instances. When the device or mailbox state on one Asterisk changes it is sent to the other Asterisk instance using a PUBLISH message containing an Asterisk specific body. This body is comprised of JSON and contains the information required to reflect the remote state change. For situations where you may not want to expose all states or you may not want to allow all states to be received you can optionally filter using a regular expression. This limits the scope of traffic.

Configuration

Configuring things to exchange state requires a few different objects: endpoint, publish, asterisk-publication, and optionally auth. These all configure a specific part in the exchange. An endpoint must be configured as a fundamental part of PJSIP is that **all** incoming requests are associated with an endpoint. A publish object tells the res_pjsip_outbound_publish where to send the PUBLISH and what type of PUBLISH message to send. An asterisk-publication object configures handling of PUBLISH messages, including whether they are permitted and from whom. Last you can optionally use authentication so that PUBLISH messages are challenged for credentials.

Example Configuration

The below configuration is for two Asterisk instances sharing all device and mailbox state between them.

Instance #1 (IP Address: 172.16.10.1):

```
[instance2]
type=endpoint

[instance2-devicestate]
type=outbound-publish
server_uri=sip:instance1@172.16.10.2
event=asterisk-devicestate

[instance2-mwi]
type=outbound-publish
server_uri=sip:instance1@172.16.10.2
event=asterisk-mwi

[instance2]
type=inbound-publication
event_asterisk-devicestate=instance2
event_asterisk-mwi=instance2

[instance2]
type=asterisk-publication
devicestate_publish=instance2-devicestate
mailboxstate_publish=instance2-mwi
device_state=yes
mailbox_state=yes
```

This configures the first instance to publish device and mailbox state to 'instance 2' located at 172.16.10.2 using a resource name of 'instance1' without authentication. As no filters exist all state will be published. It also configures the first instance to accept all device and mailbox state messages published to a resource named 'instance2' from 'instance2'.

Instance #2 (IP Address: 172.16.10.2):

```
[instance1]
type=endpoint

[instance1-devicestate]
type=outbound-publish
server_uri=sip:instance2@172.16.10.1
event=asterisk-devicestate

[instance1-mwi]
type=outbound-publish
server_uri=sip:instance2@172.16.10.1
event=asterisk-mwi

[instance1]
type=inbound-publication
event_asterisk-devicestate=instance1
event_asterisk-mwi=instance1

[instance1]
type=asterisk-publication
devicestate_publish=instance1-devicestate
mailboxstate_publish=instance1-mwi
device_state=yes
mailbox_state=yes
```

This configures the second instance to publish device and mailbox state to 'instance 1' located at 172.16.10.1 using a resource name of 'instance2' without authentication. As no filters exist all state will be published. It also configures the second instance to accept all device and mailbox state messages published to a resource named 'instance1' from 'instance1'.

Filtering

As previously mentioned state events can be filtered by the device or mailbox they relate to using a regular expression. This is configured on 'publish' types using '@device_state_filter' and '@mailbox_state_filter' and on 'asterisk-publication' types using 'device_state_filter' and 'mailbox_state_filter'. As each event is sent or received the device or mailbox is given to the regular expression and if it does not match the event is stopped.

Example

```
[instance1]
type=endpoint

[instance1-devicestate]
type=outbound-publish
server_uri=sip:instance2@172.16.10.1
event=asterisk-devicestate

[instance1-mwi]
type=outbound-publish
server_uri=sip:instance2@172.16.10.1
event=asterisk-mwi

[instance1]
type=inbound-publication
event_asterisk-devicestate=instance1
event_asterisk-mwi=instance1

[instance1]
type=asterisk-publication
devicestate_publish=instance1-devicestate
mailboxstate_publish=instance1-mwi
device_state=yes
device_state_filter=^PJSIP/
mailbox_state=yes
mailbox_state_filter=^1000
```

This builds upon the initial configuration for instance #2 but adds filtering of received events. Only device state events relating to PJSIP endpoints will be accepted. As well only mailbox state events for mailboxes starting with 1000 will be accepted.

> ⚠ This configuration is not ideal as the publishing instance (instance #1) will still send state changes for devices and mailboxes that instance #2 does not care about, thus wasting bandwidth.

Fresh Startup

When the res_pjsip_publish_asterisk module is loaded it will send its own current states for all applicable devices and mailboxes to all configured 'publish' types. Instances may optionally be configured to send a refresh request to 'publish' types as well by setting the 'devicestate_publish' and/or 'mailboxstate_publish' option in the 'asterisk-publication' type. This refresh request causes the remote instances to send current states for all applicable devices and mailboxes back, bringing the potentially newly started Asterisk up to date with its peers.

Configuring res_pjsip for Presence Subscriptions

Under Construction - This page is a stub!

Capabilities

Asterisk's PJSIP channel driver provides the same presence subscription capabilities as `chan_sip` does. This means that RFC 3856 presence and RFC 4235 dialog info are supported. Presence subscriptions support RFC 3863 PIDF+XML bodies as well as XPIDF+XML. Beyond that, Asterisk also supports subscribing to RFC 4662 lists of presence resources.

Configuration

If you are familiar with configuring subscriptions in `chan_sip` then this should be familiar to you. Configuration of presence is performed using the "hint" priority for an extension in `extensions.conf`.

On this Page

extensions.conf

```
[default]
exten => 1000,hint,PJSIP/alice
```

The line shown here is similar to any normal line in a dialplan, except that instead of a priority number or label, the word "hint" is specified. The hint is used to associate the state of individual devices with the state of a dialplan extension. An English translation of the dialplan line would be "Use the state of device PJSIP/alice as the basis for the state of extension 1000". When PJSIP endpoints subscribe to presence, they are subscribing to the state of an extension in the dialplan. By providing the dialplan hint, you are creating the necessary association in order to know which device (or devices) are relevant. For the example given above, this means that if someone subscribes to the state of extension 1000, then they will be told the state of PJSIP/alice. For more information about device state, see this page.

There are two endpoint options that affect presence subscriptions in `pjsip.conf`. The `allow_subscribe` option determines whether SUBSCRIBE requests from the endpoint are permitted to be received by Asterisk. By default, `allow_subscribe` is enabled. The other setting that affects presence subscriptions is the `context` option. This is used to determine the dialplan context in which the extension being subscribed to should be searched for. Given the dialplan snippet above, if the intent of an endpoint that subscribes to extension 1000 is to subscribe to the hint at 1000@default, then the context of the subscribing endpoint would need to be set to "default". Note that if the `context` option is set to something other than "default", then Asterisk will search that context for the hint instead.

In order for presence subscriptions to work properly, some modules need to be loaded. Here is a list of the required modules:

- `res_pjsip.so`: Core of PJSIP code in Asterisk.
- `res_pjsip_pubsub.so`: The code that implements SUBSCRIBE/NOTIFY logic, on which individual event handlers are built.
- `res_pjsip_exten_state.so`: Handles the "presence" and "dialog" events.
- `res_pjsip_pidf_body_generator.so`: This module generates application/pidf+xml message bodies. Required for most subscriptions to the "presence" event.
- `res_pjsip_xpidf_body_generator.so`: This module generates application/xpidf+xml message bodies. Required for some subscriptions to the "presence" event.
- `res_pjsip_dialog_info_body_generator.so`: Required for subscriptions to the "dialog" event. This module generates application/dialog-info message bodies.

If you are unsure of what event or what body type your device uses for presence subscriptions, consult the device manufacturer's manual for more information.

Presence Customisations

Digium Presence

Digium phones are outfitted with a custom supplement to the base PIDF+XML presence format that allows for XMPP-like presence to be understood. To add this, the hint can be modified to include an additional presence state, like so:

extensions.conf

```
[default]
exten => 1000,hint,PJSIP/alice,CustomPresence:alice
```

This means that updates to the presence state of CustomPresence:alice will also be conveyed to subscribers to extension 1000. For more information on presence state in Asterisk, see this page.

The `res_pjsip_pidf_digium_body_supplement.so` module must be loaded in order for additional presence details to be reported.

Rich Presence (limited)

Some rich presence supplements that were in `chan_sip` have been migrated to the PJSIP channel driver as well. This is an extremely limited implementation of the "activities" element of a person. The `res_pjsip_pidf_eyebeam_body_supplement.so` module is required to add this functionality.

Resource List Subscriptions (RLS)

Overview

Beginning in Asterisk 13, Asterisk supports RFC 4662 resource list subscriptions in its PJSIP-based SIP implementation.

In a PBX environment, it is common for SIP devices to subscribe to many resources offered by the PBX. This holds especially true for presence resources. Consider a small office where an Asterisk server acts as a PBX for 20 people, each with a SIP desk phone. Each of those 20 phones subscribes to the state of the others in the office. In this case, each of the phones would create 19 subscriptions (since the phone does not subscribe to its own state). When totalled, the Asterisk server would maintain 20 * 19 = 380 subscriptions. For an office with 30 people, the total number of subscriptions becomes 30 * 29 = 870 subscriptions. For an office with 40 people, the total number of subscriptions becomes 40 * 39 = 1560. That is about four times the number of subscriptions for the 20-person office, despite only having twice the number of people. The number of subscriptions follows a geometric progression, leading to a situation commonly called an *N-squared* problem. In other words, the amount of traffic generated and amount of server resources required are proportional to the square of the number of users (N) on the system. The N-squared problem with subscriptions can be a limiting factor for PBX deployers for several reasons:

- In a situation where all phones boot up simultaneously, each of the phones will be sending out their SIP SUBSCRIBEs nearly simultaneously, placing a larger-than-average burden on the Asterisk server's CPU.
- In the SIP stack, N-squared long-term SIP dialogs have to be maintained, tying up more system resources (e.g. memory).

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These limitations can drastically limit the number of devices a PBX administrator can use with an Asterisk system. Even if the hardware is capable of handling the mean traffic of, say, 200 users, it may be required to limit the number of users to 50 or fewer because of the N-squared subscriptions generating so much simultaneous traffic.

Resource list subscriptions provide relief for this problem by allowing for resources to be grouped into lists. A single subscription to a list will result in multiple back-end subscriptions to the resources in that list. Notifications of state changes can also be batched so that multiple state changes may be conveyed in a single message. This can help to significantly decrease the amount of subscription-related traffic and processing being performed.

Configuring Resource List Subscriptions

RLS is configured in `pjsip.conf` using a special configuration section type called "resource_list". Here is an example of a simple resource list:

pjsip.conf

```
[sales]
type = resource_list
event = presence
list_item = alice
list_item = bob
list_item = carol
```

It should be simple to glean the intent of this list. We have created a list called "sales" that provides the presence of the sales team of alice, bob, and carol. Let's go over each of the options in more detail.

- `type`: This must be set to "resource_list" so that the configuration parser knows that it is looking at a resource list.
- `event`: The SIP event package provided by this resource list. Asterisk, as provided by Digium, provides support for the following event packages:
    - presence: Provides ability to determine when devices are in use or not. Commonly known as BLF.
    - dialog: An alternate method of providing BLF. Used by certain SIP equipment instead of the presence event package.
    - message-summary: Provides the ability to determine the number of voice mail messages that a mailbox contains. Commonly known as MWI.
- `list_item`: This is the name of a resource that belongs to the list. The formatting of list items is dependent on the event package provided by the list.
    - presence: This is the name of an extension in the dialplan. In the example, the extensions "alice", "bob", and "carol" exist in `extensions.conf`.
    - dialog: The same as the presence event package.
    - message-summary: This is the name of a mailbox. If you are using `app_voicemail`, then mailboxes will be in the form of "mailbox@context". If you are using an external voicemail system, then the name of the mailbox will be in whatever format the external voicemail system uses for mailbox names.

  The list items in the example were placed on separate lines, but it is also valid to place multiple list items on a single line: `list_item =`

alice,bob,carol. Note also that list items can also be other resource lists of the same event type.

There is one further option that is not listed here, but deserves some mention: full_state. RFC 4662 defines "full" and "partial" state notifications. When the states of a subset of resources on a resource list changes, a server has the option of sending a notification that only contains the resources whose states have changed. This is a partial state notification. A full state notification would include the states of all resources in the list, even if only some of the resources' states have changed. The full_state option allows for full state notifications to be transmitted unconditionally. By default, full_state is disabled on resource list subscriptions in order to keep the size of notifications small. It is **highly recommended** that you use the default value for this option unless you are using a client that does not understand partial state notifications.

Batching Notifications

In addition to the basic options listed above, there is another option, notification_batch_interval that can be used to change Asterisk's behavior when sending notifications of resource state changes on a list. By default, whenever the state of any resource on a list changes, Asterisk will immediately send out a notification of the state change. If, however, a notification_batch_interval is specified, then when a resource state changes, Asterisk will start a timer for the specified interval. While the timer is running, any further state changes of resources in the list are batched along with the original state change that started the timer. When the timer expires, then all batched state changes are sent in a single NOTIFY.

Let's modify the previous configuration to use a batching interval:

pjsip.conf

```
[sales]
type = resource_list
event = presence
list_item = alice
list_item = bob
list_item = carol
notification_batch_interval = 2000
```

The units for the notification_batch_interval are milliseconds. With this configuration, Asterisk will collect resource state changes for 2000 milliseconds before sending notifications on this resource list.

The biggest advantage of notification batching is that it can decrease the number of NOTIFY requests that Asterisk sends. If two SIP phones on a PBX are having a conversation with one another, when a call completes, both phones are likely to change states to being not in use. By having a batching interval configured, it would allow for a single NOTIFY to indicate both devices' state changes instead of having to send two separate NOTIFY requests.

The biggest disadvantage of notification batching is that it becomes possible for transient states for a device to be missed. If you have a batching interval of 3000 milliseconds, and a phone only rings for one second before it is answered, it means that the ringing state of the phone never got transmitted to interested listeners.

Corner Cases

Non-existent List Items

Let's say you have the following list configured in pjsip.conf:

pjsip.conf

```
[sales]
type = resource_list
event = presence
list_item = alice
list_item = bob
list_item = carol
```

And you have the following in extensions.conf

extensions.conf

```
[default]
exten => alice,hint,PJSIP/alice
exten => bob,hint,PJSIP/bob
```

Notice that there is no "carol" extension in extensions.conf. What happens when a user attempts to subscribe to the sales list?

When the subscription arrives, Asterisk recognizes the subscription as being for the list. Asterisk then acts as if it is establishing individual subscriptions to each of the list items the same way it would if a subscription had arrived directly for the list item. In this case, the subscriptions to alice and bob succeed. However, the presence subscription handler complains that it cannot subscribe to carol since the resource does not exist.

The policy currently used is that if subscription to at least one list resource succeeds, then the subscription to the entire list has succeeded. Only the list items that were successfully subscribed to will be reflected in the list subscription. If subscription to **all** list items fails, then the subscription to the list also fails.

Loops

Let's say you have the following pjsip.conf file:

pjsip.conf

```
[sales]
type = resource_list
event = presence
list_item = tech_support

[tech_support]
type = resource_list
event = presence
list_item = sales
```

Notice that the sales list contains the tech_support list, and the tech_support list contains the sales list. We have a loop here. How is that handled?

Asterisk's policy with loops is to try to resolve the issue while being as graceful as possible. The way it does this is that when it detects a loop, it essentially considers the looped subscription to be a failed list item subscription. The process would go something like this:

1. A subscription arrives for the sales list.
2. We attempt to subscribe to the tech_support list item in the sales list.
3. Inside the tech_support list, we see the sales list as a list item.
4. We notice that we've already visited the sales list, so we fail the subscription to the sales list list item.
5. Since subscriptions to all list items in the tech support list failed, the subscription to the tech support list failed.
6. Since the tech support list was the only list item in the sales list, and that subscription failed, the subscription to the sales list fails as well.

What if the configured lists were modified slightly:

pjsip.conf

```
[sales]
type = resource_list
event = presence
list_item = tech_support


[tech_support]
type = resource_list
event = presence
list_item = sales
list_item = alice
```

Notice that the tech_support list now also has alice as a list_item. How does the process change on a subscription attempt to sales?

1. A subscription arrives for the sales list
2. We attempt to subscribe to the tech_support list item in the sales list.
3. Inside the tech_support list, we see the sales list as a list item.
4. We notice that we've already visited the sales list, so we fail the subscription to the sales list list item.
5. We move on to the next list_item in tech_support, alice.
6. We attempt a subscription to alice, and it succeeds.
7. Since at least one subscription to a list item in tech_support succeeded, the subscription to tech_support succeeds.
8. Since the subscription to the only list item in sales succeeded, the subscription to sales succeeds.

So in this case, even though the configuration contains a loop, Asterisk is able to successfully create a subscription while trimming the loops out.

Ambiguity

Duplicated List Names

If a list name is duplicated, then the configuration framework of Asterisk will not allow for the two to exist as separate entities. It is expected that the most recent list in the configuration file will overwrite the earlier ones.

While this may seem like an obvious thing, users may be tempted to configure lists that have the same name but that exist for different SIP event packages. While this may seem like a legitimate configuration, it will not work as intended.

List and Resources with Same Name

One flaw that RLS has is that there is no way to know whether a subscription is intended to be for a list or for an individual resource. Let's say you have the following pjsip configuration:

pjsip.conf

```
[sales]
type = resource_list
event = presence
list_item = alice
list_item = bob
list_item = carol
```

And let's say you have the following `extensions.conf`:

extensions.conf

```
[default]
exten => sales,hint,Custom:sales
```

What happens if someone attempts to subscribe to the "sales" presence resource?

One easy way to determine intent is to check the Supported: header in the incoming SUBSCRIBE request. If "eventlist" does not appear, then the subscriber does not support RLS and is therefore definitely subscribing to the individual sales resource as described in `extensions.conf`.

But if the subscriber does support RLS, then Asterisk's policy is to always assume that the subscriber intends to subscribe to the list, not the individual resource.

Conflicting Batching Intervals

`notification_batch_interval` can be configured on any resource list. Consider the following configuration:

pjsip.conf

```
[sales]
type = resource_list
event = presence
list_item = sales_b
list_item = carol
list_item = david
notification_batch_interval = 3000

[sales_b]
type = resource_list
event = presence
list_item = alice
list_item = bob
notification_batch_interval = 10000
```

What is the batch interval when a user subscribes to the sales list?

The policy that Asterisk enforces is that only the batch interval of the top-most list in the hierarchy is applied. So in the example above, the batch interval would be 3000 milliseconds since the top-most list in the hierarchy is the sales list. If the sales list did not have a batch interval configured, then there would be no batch interval for the list subscription at all.

Limitations

List size

Due to limitations in the PJSIP stack, Asterisk is limited regarding the size of a SIP message that can be transmitted. Asterisk currently works around the built-in size limitation of PJSIP (4000 bytes by default) and can send a message up to 64000 bytes instead. RFC 4662 requires that when sending a NOTIFY request due to an inbound SUBSCRIBE request, we must send the full state of the resource list in response. For large lists, this may mean that the NOTIFY will exceed the size limit.

It is difficult to try to quantify the limit in terms of number of list resources since different body types are more verbose than others, and different configurations will have different variables that will factor into the size of the message (e.g. the length of SIP URIs for one system may be three times as long as the SIP URIs for a separate system, depending on how things are configured).

If you create a very large list, and you find that Asterisk is unable to send NOTIFY requests due to the size of the list, consider breaking the list into smaller sub-lists if possible.

Lack of dynamism

Resource lists can be updated as you please, adding and removing list items, altering the batching interval, etc. However, you will find that when a list is altered, any current subscriptions to the list are not updated to reflect the changes to the list. This is because the list is read from configuration at the time that the subscription is established, and the configuration is never again consulted during the lifetime of the subscription. If configuration is updated, then you must terminate your current subscriptions to the list and create a new subscription in order to apply the changes.

Similarly, the state of resources is locked in at the time the subscription is established. For instance, if a list contains a list item that does not exist at the time the subscription is established, if that resource comes into existence later, then the established subscription is not updated to properly reflect the added list item. The subscription must be terminated and re-established in order to have the corrected list item included.

Configuring Outbound Registrations

This page is under construction. Please refrain from commenting here until this warning is removed.

Overview

Like with `chan_sip`, Asterisk's PJSIP implementation allows for configuration of outbound registrations. Unlike `chan_sip`, it is not implemented in an obnoxious way. Like with most concepts in PJSIP configuration, outbound registrations are confined to a configuration section of their own.

Configuration options

A list of outbound registration configuration options can be found on this page. Here is a simple example configuration for an outbound registration to a provider:

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pjsip.conf

```
[my_provider]
type = registration
server_uri = sip:registrar@example.com
client_uri = sip:client@example.com
contact_user = inbound-calls
```

This results in the following outbound REGISTER request being sent by Asterisk:

```
<--- Transmitting SIP request (557 bytes) to UDP:93.184.216.119:5060 --->
REGISTER sip:registrar@example.com SIP/2.0
Via: SIP/2.0/UDP 10.24.20.249:5060;rport;branch=z9hG4bKPjd1a32b43-82ed-4f98-ae24-20149cdf0749
From: <sip:client@example.com>;tag=904e0db9-8297-4bb0-89c5-5cfe1cfed654
To: <sip:client@example.com>
Call-ID: 03241f7b-3936-4140-8bad-6840774b78d9
CSeq: 10266 REGISTER
Contact: <sip:inbound-calls@10.24.20.249:5060>
Expires: 3600
Allow: OPTIONS, SUBSCRIBE, NOTIFY, PUBLISH, INVITE, ACK, BYE, CANCEL, UPDATE, PRACK, MESSAGE, REFER, REGISTER
Max-Forwards: 70
Content-Length:  0
```

Let's go over how the options were applied to this REGISTER:

- The `server_uri` is the actual URI where the registrar is located. If you are registering with a SIP provider, they should give this information to you.
- The `client_uri` is used in the To and From headers of the REGISTER. In other words, this is the address of record to which you are binding a contact URI. If registering to a SIP provider, they may require you to provide a specific username in order to identify that the REGISTER is coming from you. Note that the domain of the `client_uri` is the same as the server URI. This is common when indicating that the registrar receiving the REGISTER is responsible for the URI being registered to it.
- The `contact_user` option can be seen in the user portion of the URI in the Contact header. This allows for you to control where incoming calls from the provider will be routed. Calls from the provider will arrive in this extension in the dialplan. Note that this option does not relate to endpoint-related options. For information on relating outbound registrations and endpoints, see the following section.

An English translation of the above REGISTER is "Tell the server at sip:registrar@example.com that when SIP traffic arrives addressed to sip:client@example.com, the traffic should be sent to sip:inbound-calls@10.24.20.249." Note in this example that 10.24.20.249 is the IP address of the Asterisk server that sent the outbound REGISTER request.

The transport type, e.g. tcp, for the registration can be specified by appending the details to the client_uri and/or server_uri parameters, e.g.:

```
[my_provider]
type = registration
server_uri = sip:registrar@example.com\;transport=tcp
client_uri = sip:client@example.com\;transport=tcp
contact_user = inbound-calls
```

Outbound registrations and endpoints

If you examine the configuration options linked in the previous section, you will notice that there is nothing that ties an outbound registration to an endpoint. The two are considered completely separate from each other, as far as Asterisk is concerned. However, it is likely that if you are registering to an ITSP, you will want to receive incoming calls from that provider. This means that you will need to set up an endpoint that represents this provider. An example of such an endpoint configuration can be found here, but it is a bit complex. Let's instead make a simpler one just for the sake of explanation. Assuming the previous registration has been configured, we can add the following:

pjsip.conf

```
[my_provider_endpoint]
type = endpoint

[my_provider_identify]
type = identify
match = <ip address of provider>
endpoint = my_provider
```

This represents the bare minimum necessary in order to accept incoming calls from the provider. The `identify` section makes it so that incoming SIP traffic from the IP address in the `match` option will be associated with the endpoint called `my_provider_endpoint`.

If you also wish to make outbound calls to the provider, then you would also need to add an AoR section so that Asterisk can know where to send calls directed to "my_provider_endpoint".

pjsip.conf

```
[my_provider_endpoint]
type = endpoint
aors = my_provider_aor

[my_provider_identify]
type = identify
match = <ip address of provider>
endpoint = my_provider

[my_provider_aor]
type = aor
contact = sip:my_provider@example.com
```

Let me reiterate that this is the **bare minimum**. If you want calls to and from the provider to actually work correctly, you will want to set a context, codecs, authentication, etc. on the endpoint.

Authentication

It is likely that if you are registering to a provider, you will need to provide authentication credentials. Authentication for outbound registrations is configured much the same as it is for endpoints. The `outbound_auth` option allows for you to point to a `type = auth` section in your configuration to refer to when a registrar challenges Asterisk for authentication. Let's modify our configuration to deal with this:

pjsip.conf

```
[my_provider]
type = registration
server_uri = sip:registrar@example.com
client_uri = sip:client@example.com
contact_user = inbound-calls
outbound_auth = provider_auth

[provider_auth]
type = auth
username = my_username
password = my_password
```

With this configuration, now if the registrar responds to a REGISTER by challenging for authentication, Asterisk will use the authentication credentials in the provider_auth section in order to authenticate. Details about what options are available in auth sections can be found here in the "auth" section.

Dealing with Failure

Temporary and Permanent Failures

Whenever Asterisk sends an outbound registration and receives some sort of failure response from the registrar, Asterisk makes a determination about whether a response can be seen as a permanent or temporary failure. The following responses are always seen as temporary failures:

- No Response
- 408 Request Timeout
- 500 Internal Server Error
- 502 Bad Gateway
- 503 Service Unavailable
- 504 Server Timeout
- Any 600-class response

In addition, there is an option called `auth_rejection_permanent` that can be used to determine if authentication-related rejections from a registrar are treated as permanent or temporary failures. By default, this option is enabled, but disabling the setting means the following two responses are also treated as temporary failures:

- 401 Unauthorized
- 407 Proxy Authentication Required

What is meant by temporary and permanent failures? When a temporary failure occurs, Asterisk may re-attempt registering if a `retry_interval` is configured in the outbound registration. The `retry_interval` is the number of seconds Asterisk will wait before attempting to send another REGISTER request to the registrar. By default, outbound registrations have a `retry_interval` of 60 seconds. Another configuration option, `max_retries`, determines how many times Asterisk will attempt to re-attempt registration before permanently giving up. By default, `max_retries` is set to 10.

Permanent failures result in Asterisk immediately ceasing to re-attempt the outbound registration. All responses that were not previously listed as temporary failures are considered to be permanent failures. There is one exception when it comes to permanent failures. The `forbidden_retry_interval` can be set such that if Asterisk receives a 403 Forbidden response from a registrar, Asterisk can wait the number of seconds indicated and re-attempt registration. Retries that are attempted in this manner count towards the same `max_retries` value as temporary failure retries.

Let's modify our outbound registration to set these options to custom values:

pjsip.conf

```
[my_provider]
type = registration
server_uri = sip:registrar@example.com
client_uri = sip:client@example.com
contact_user = inbound-calls
outbound_auth = provider_auth
auth_rejection_permanent = no
retry_interval = 30
forbidden_retry_interval = 300
max_retries = 20
```

In general, this configuration is more lenient than the default. We will retry registration more times, we will retry after authentication requests and forbidden responses, and we will retry more often.

CLI and AMI

Monitoring Status

You can monitor the status of your configured outbound registrations via the CLI and the Asterisk Manager Interface. From the CLI, you can issue the command `pjsip show registrations` to list all outbound registrations. Here is an example of what you might see:

```
<Registration/ServerURI..............................>  <Auth..........>  <Status.......>
 ==========================================================================================
 my_provider/sip:registrar@example.com                   provider_auth     Unregistered
 outreg/sip:registrar@example.com                        n/a               Unregistered
```

On this particular Asterisk instance, there are two outbound registrations configured. The headers at the top explain what is in each column. The "Status" can be one of the following values:

- Unregistered: Asterisk is currently not registered. This is most commonly seen when the registration has not yet been established. This can also be seen when the registration has been forcibly unregistered or if the registration times out.

- Registered: Asterisk has successfully registered.
- Rejected: Asterisk attempted to register but a failure occurred. See the above section for more information on failures that may occur.
- Stopped: The outbound registration has been removed from configuration, and Asterisk is attempting to unregister.

In addition, you can see the details of a particular registration by issuing the `pjsip show registration <registration name>` command. If I issue `pjsip show registration my_provider`, I see the following:

```
<Registration/ServerURI..............................>  <Auth..........>  <Status.......>
 =========================================================================================

 my_provider/sip:registrar@example.com                   provider_auth     Unregistered


 ParameterName            : ParameterValue
 ====================================================
 auth_rejection_permanent : false
 client_uri               : sip:client@example.com
 contact_user             : inbound-calls
 expiration               : 3600
 forbidden_retry_interval : 300
 max_retries              : 20
 outbound_auth            : provider_auth
 outbound_proxy           :
 retry_interval           : 30
 server_uri               : sip:registrar@example.com
 support_path             : false
 transport                :
```

This provides the same status line as before and also provides the configured values for the outbound registration.

AMI provides the `PJSIPShowRegistrationsOutbound` command that provides the same information as the CLI commands. Here is an example of executing the command in an AMI session:

```
action: PJSIPShowRegistrationsOutbound


Response: Success
EventList: start
Message: Following are Events for each Outbound registration


Event: OutboundRegistrationDetail
ObjectType: registration
ObjectName: my_provider
SupportPath: false
AuthRejectionPermanent: false
ServerUri: sip:registrar@example.com
ClientUri: sip:client@example.com
RetryInterval: 30
MaxRetries: 20
OutboundProxy:
Transport:
ForbiddenRetryInterval: 300
OutboundAuth: provider_auth
ContactUser: inbound-calls
Expiration: 3600
Status: Rejected
NextReg: 0


Event: OutboundRegistrationDetail
ObjectType: registration
ObjectName: outreg
SupportPath: false
AuthRejectionPermanent: true
ServerUri: sip:registrar@example.com
ClientUri: sip:client@example.com
RetryInterval: 60
MaxRetries: 10
OutboundProxy:
Transport:
ForbiddenRetryInterval: 0
OutboundAuth:
ContactUser: inbound-calls
Expiration: 3600
Status: Rejected
NextReg: 0


Event: OutboundRegistrationDetailComplete
EventList: Complete
Registered: 0
NotRegistered: 2
```

The command sends `OutboundRegistrationDetail` events for each configured outbound registration. Most information is the same as the CLI displays, but there is one additional piece of data displayed: NextReg. This is the number of seconds until Asterisk will send a new REGISTER request to the registrar. In this particular scenario, that number is 0 because the two outbound registrations have reached their maximum number of retries.

Manually Unregistering

The AMI and CLI provide ways for you to manually unregister if you want. The CLI provides the `pjsip send unregister <registration name>` command. AMI provides the `PJSIPUnregister` command to do the same thing.

> ⚠ After manually unregistering, the specified outbound registration will continue to reregister based on its last registration expiration.

Realtime

At the time of this wiki article writing, it is not possible, nor would it be recommended, to use dynamic realtime for outbound registrations. The code in `res_pjsip_outbound_registration.so`, the module that allows outbound registrations to occur, does not attempt to look outside of `pjsip.conf` for details regarding outbound registrations. This is done because outbound registrations are composed both of the configuration values as well as state (e.g. how many retries have we attempted for an outbound registration). When pulling configuration from a file, a reload is necessary, which makes it easy to have a safe place to transfer state information or alter configuration values when told that things have changed. With dynamic realtime, this is much harder to manage since presumably the configuration could change at any point.

If you prefer to use a database to store your configuration, you are free to use static realtime for outbound registrations instead. Like with a configuration file, you will be forced to reload (from the CLI, `module reload res_pjsip_outbound_registration.so`) in order to apply configuration changes.

Asterisk PJSIP Troubleshooting Guide

This page is currently under construction. Please refrain from commenting until this warning is removed.

Overview

Are you having problems getting your PJSIP setup working properly? If you are encountering a common problem then hopefully your answer can be found on this page.

Before looking any further here, you should make sure that you have gathered enough information from Asterisk to know what your issue is. It is suggested that you perform the following actions at the Asterisk CLI:

- `core set verbose 4`
- `core set debug 4`
- `pjsip set logger on`

With these options enabled, this will allow you to more easily see what is going on behind the scenes in your failing scenario. It also can help you to cross-reference entries on this page since several debug, warning, and error messages will be quoted here.

Inbound Calls

Unrecognized Endpoint

All inbound SIP traffic to Asterisk must be matched to a configured endpoint. If Asterisk is unable to determine which endpoint the SIP request is coming from, then the incoming request will be rejected. If you are seeing messages like:

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```
[2014-10-13 16:12:17.349] DEBUG[27284]: res_pjsip_endpoint_identifier_user.c:106 username_identify: Could not identify endpoint
by username 'eggowaffles'
```

or

```
[2014-10-13 16:13:07.201] DEBUG[27507]: res_pjsip_endpoint_identifier_ip.c:113 ip_identify_match_check: Source address
127.0.0.1:5061 does not match identify 'david-ident'
```

then this is a good indication that the request is being rejected because Asterisk cannot determine which endpoint the incoming request is coming from.

How does Asterisk determine which endpoint a request is coming from? Asterisk uses something called "endpoint identifiers" to determine this. There are three endpoint identifiers bundled with Asterisk: user, ip, and anonymous.

Identify by User

The user endpoint identifier is provided by the `res_pjsip_endpoint_identifier_user.so` module. If nothing has been explicitly configured with regards to endpoint identification, this endpoint identifier is the one being used. The way it works is to use the user portion of the From header from the incoming SIP request to determine which endpoint the request comes from. Here is an example INVITE:

```
<--- Received SIP request (541 bytes) from UDP:127.0.0.1:5061 --->
INVITE sip:service@127.0.0.1:5060 SIP/2.0
Via: SIP/2.0/UDP 127.0.0.1:5061;branch=z9hG4bK-27600-1-0
From: breakfast <sip:eggowaffles@127.0.0.1:5061>;tag=27600SIPpTag001
To: sut <sip:service@127.0.0.1>
Call-ID: 1-27600@127.0.0.1
CSeq: 1 INVITE
Contact: sip:eggowaffles@127.0.0.1:5061
Max-Forwards: 70
Content-Type: application/sdp
Content-Length:   163

v=0
o=user1 53655765 2353687637 IN IP4 127.0.0.1
s=-
c=IN IP4 127.0.0.1
t=0 0
m=audio 6000 RTP/AVP 0
a=rtpmap:8 PCMA/8000
a=rtpmap:0 PCMU/8000
a=ptime:20
```

In this example, the URI in the From header is "sip:eggowaffles@127.0.0.1:5061". The user portion is "eggowaffles", so Asterisk attempts to look up an endpoint called "eggowaffles" in its configuration. If such an endpoint is not configured, then the INVITE is rejected by Asterisk. The most common cause of the problem is that the user name referenced in the From header is not the name of a configured endpoint in Asterisk.

But what if you have configured an endpoint called "eggowaffles"? It is possible that there was an error in your configuration, such as an option name that Asterisk does not recognize. If this is the case, then the endpoint may not have been loaded at all. Here are some troubleshooting steps to see if this might be the case:

- From the CLI, issue the "pjsip show endpoints" command. If the endpoint in question does not show up, then there likely was a problem attempting to load the endpoint on startup.
- Go through the logs from Asterisk startup. You may find that there was an error reported that got lost in the rest of the startup messages. For instance, be on the lookout for messages like:

  ```
  [2014-10-13 16:25:01.674] ERROR[27771]: config_options.c:710 aco_process_var: Could not find option suitable for category
  'eggowaffles' named 'setvar' at line 390 of
  [2014-10-13 16:25:01.674] ERROR[27771]: res_sorcery_config.c:275 sorcery_config_internal_load: Could not create an object
  of type 'endpoint' with id 'eggowaffles' from configuration file 'pjsip.conf'
  ```

  In this case, I set an endpoint option called "setvar" instead of the appropriate "set_var". The result was that the endpoint was not loaded.
- If you do not see such error messages in the logs, but you do not see the endpoint listed in "pjsip show endpoints", it may be that you forgot to put `type = endpoint` in your endpoint section. In this case, the entire section would be ignored since Asterisk did not know that this was an endpoint section.

Identify by IP address

Asterisk can also recognize endpoints based on the source IP address of the SIP request. This requires setting up a `type = identify` section in your configuration to match IP addresses or networks to a specific endpoint. Here are some troubleshooting steps:

- Ensure that `res_pjsip_endpoint_identifier_ip.so` is loaded and running. From the CLI, run `module show like res_pjsip_endpoint_identifier_ip.so`. The output should look like the following:

  ```
  Module                         Description                              Use Count  Status      Support Level
  res_pjsip_endpoint_identifier_ip.so PJSIP IP endpoint identifier             0          Running              core
  ```

- Run the troubleshooting steps from the Identify by User section to ensure that the endpoint you have configured has actually been properly loaded.
- From the Asterisk CLI, run the command `pjsip show endpoint <endpoint name>`. Below the headers at the top of the output, you should see something like the following:

  ```
  Endpoint:  david/6001                                    Unavailable   0 of inf
       InAuth:  david-auth/david
          Aor:  david                                      10
    Transport:  main-transport            udp      0      0  0.0.0.0:5060
     Identify:  10.24.16.36/32
  ```

  Notice the bottom line. This states that the endpoint is matched based on the IP address 10.24.16.36. If you do not see such a line for the endpoint that you expect to be matched, then there is likely a configuration error. If the line does appear, then ensure that the IP address listed matches what you expect for the endpoint.
- If you are noticing that Asterisk is matching the incorrect endpoint by IP address, ensure that there are no conflicts in your configuration. Run the `pjsip show endpoints` command and look for issues such as the following:

```
Endpoint:  carol/6000                                           Unavailable   0 of inf
     InAuth:  carol-auth/carol
        Aor:  carol                                             10
  Transport:  main-transport            udp      0      0  0.0.0.0:5060
   Identify:  10.0.0.0/8

 Endpoint:  david/6001                                           Unavailable   0 of inf
     InAuth:  david-auth/david
        Aor:  david                                             10
  Transport:  main-transport            udp      0      0  0.0.0.0:5060
   Identify:  10.24.16.36/32
```

Notice that if a SIP request arrives from 10.24.16.36, it is ambiguous if the request should be matched to carol or david.

If you run `pjsip show endpoint <endpoint name>` and do not see an "Identify" line listed, then there is likely a configuration issue somewhere. Here are some common pitfalls

- Ensure that your identify section has `type = identify` in it. Without this, Asterisk will completely ignore the configuration section.
- Ensure that your identify section has an `endpoint` option set in it and that the endpoint is spelled correctly.
- Double-check your `match` lines for common errors:
  - You cannot use FQDNs or hostnames. You must use IP addresses.
  - Ensure that you do not have an invalid netmask (e.g. 10.9.3.4/255.255.255.300, 127.0.0.1/33).
  - Ensure that you have not mixed up /0 and /32 when using CIDR notation.
- If you are using a configuration method other than a config file, ensure that `sorcery.conf` is configured correctly. Since identify sections are not provided by the base `res_pjsip.so` module, you must ensure that the configuration resides in the `res_pjsip_endpoint_identifier_ip` section of `sorcery.conf`. For example, if you are using dynamic realtime, you might have the following configuration:

sorcery.conf

```
[res_pjsip_endpoint_identifier_ip]
identify = realtime,ps_endpoint_id_ips
```

And then you would need the corresponding config in `extconfig.conf`:

extconfig.conf

```
[settings]
ps_endpoint_id_ips => odbc
```

Anonymous Identification

Anonymous endpoint identification allows for a specially-named endpoint called "anonymous" to be matched if other endpoint identifiers are not able to determine which endpoint a request originates from. The `res_pjsip_endpoint_identifier_anonymous.so` module is responsible for matching the incoming request to the anonymous endpoint. If SIP traffic that you expect to be matched to the anonymous endpoint is being rejected, try the following troubleshooting steps:

- Ensure that `res_pjsip_endpoint_identifier_anonymous.so` is loaded and running. From the Asterisk CLI, run `module show like res_pjsip_endpoint_identifier_anonymous.so`. The output should look like the following:

```
Module                         Description                             Use Count  Status      Support Level
res_pjsip_endpoint_identifier_anonymous.so PJSIP Anonymous endpoint identifier      0          Running              core
```

- Ensure that the "anonymous" endpoint has been properly loaded. See the troubleshooting steps in the Identify by User section for more details about how to determine if an endpoint has been loaded.

Authentication is failing

The first thing you should check if you believe that authentication is failing is to ensure that this is the actual problem. Consider the following SIP call from endpoint 200 to Asterisk:

```
<--- Received SIP request (1053 bytes) from UDP:10.24.16.37:5060 --->
INVITE sip:201@10.24.20.249 SIP/2.0
Via: SIP/2.0/UDP 10.24.16.37:5060;rport;branch=z9hG4bKPjQevrxvXqk9Lk5xSW.pzQQb8SAWnJ5Lll
Max-Forwards: 70
From: "200" <sip:200@10.24.20.249>;tag=DTD-tYEwFMmbPyu0YWalLQdbEUGSLGN5
To: <sip:201@10.24.20.249>
Contact: "200" <sip:200@10.24.16.37:5060;ob>
Call-ID: q.TF2SAaX3jn8dtaLTOCuIO8FRyDCsSR
CSeq: 9775 INVITE
Allow: PRACK, INVITE, ACK, BYE, CANCEL, UPDATE, SUBSCRIBE, NOTIFY, REFER, MESSAGE, OPTIONS
Supported: replaces, 100rel, timer, norefersub
Session-Expires: 1800
Min-SE: 90
User-Agent: Digium D40 1_4_0_0_57389
Content-Type: application/sdp
```

```
Content-Length:   430

v=0
o=- 108683760 108683760 IN IP4 10.24.16.37
s=digphn
c=IN IP4 10.24.16.37
t=0 0
a=X-nat:0
m=audio 4046 RTP/AVP 0 8 9 111 18 58 118 58 96
a=rtcp:4047 IN IP4 10.24.16.37
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:9 G722/8000
a=rtpmap:111 G726-32/8000
a=rtpmap:18 G729/8000
a=rtpmap:58 L16/16000
a=rtpmap:118 L16/8000
a=rtpmap:58 L16-256/16000
a=sendrecv
a=rtpmap:96 telephone-event/8000
a=fmtp:96 0-15


<--- Transmitting SIP response (543 bytes) to UDP:10.24.16.37:5060 --->
SIP/2.0 401 Unauthorized
Via: SIP/2.0/UDP 10.24.16.37:5060;rport;received=10.24.16.37;branch=z9hG4bKPjQevrxvXqk9Lk5xSW.pzQQb8SAWnJ5Lll
Call-ID: q.TF2SAaX3jn8dtaLTOCuIO8FRyDCsSR
From: "200" <sip:200@10.24.20.249>;tag=DTD-tYEwFMmbPyu0YWalLQdbEUGSLGN5
To: <sip:201@10.24.20.249>;tag=z9hG4bKPjQevrxvXqk9Lk5xSW.pzQQb8SAWnJ5Lll
CSeq: 9775 INVITE
WWW-Authenticate: Digest
realm="asterisk",nonce="1413305427/8dd1b7f56aba97da45754f7052d8a688",opaque="3b9c806b61adf911",algorithm=md5,qop="auth"
Content-Length:  0


<--- Received SIP request (370 bytes) from UDP:10.24.16.37:5060 --->
ACK sip:201@10.24.20.249 SIP/2.0
Via: SIP/2.0/UDP 10.24.16.37:5060;rport;branch=z9hG4bKPjQevrxvXqk9Lk5xSW.pzQQb8SAWnJ5Lll
Max-Forwards: 70
From: "200" <sip:200@10.24.20.249>;tag=DTD-tYEwFMmbPyu0YWalLQdbEUGSLGN5
To: <sip:201@10.24.20.249>;tag=z9hG4bKPjQevrxvXqk9Lk5xSW.pzQQb8SAWnJ5Lll
Call-ID: q.TF2SAaX3jn8dtaLTOCuIO8FRyDCsSR
CSeq: 9775 ACK
Content-Length:  0


<--- Received SIP request (1343 bytes) from UDP:10.24.16.37:5060 --->
INVITE sip:201@10.24.20.249 SIP/2.0
Via: SIP/2.0/UDP 10.24.16.37:5060;rport;branch=z9hG4bKPjCrZnx79augJPtGcTbYlXEs2slZNtwYeC
Max-Forwards: 70
From: "200" <sip:200@10.24.20.249>;tag=DTD-tYEwFMmbPyu0YWalLQdbEUGSLGN5
To: <sip:201@10.24.20.249>
Contact: "200" <sip:200@10.24.16.37:5060;ob>
Call-ID: q.TF2SAaX3jn8dtaLTOCuIO8FRyDCsSR
CSeq: 9776 INVITE
Allow: PRACK, INVITE, ACK, BYE, CANCEL, UPDATE, SUBSCRIBE, NOTIFY, REFER, MESSAGE, OPTIONS
Supported: replaces, 100rel, timer, norefersub
Session-Expires: 1800
Min-SE: 90
User-Agent: Digium D40 1_4_0_0_57389
Authorization: Digest username="200", realm="asterisk", nonce="1413305427/8dd1b7f56aba97da45754f7052d8a688",
uri="sip:201@10.24.20.249", response="2da759314909af8507a59cd1b6bc0baa", algorithm=md5,
cnonce="-me-qsYc.rGU-I5A6n-Dy8IhCBg9wKe8", opaque="3b9c806b61adf911", qop=auth, nc=00000001
Content-Type: application/sdp
Content-Length:   430

v=0
o=- 108683760 108683760 IN IP4 10.24.16.37
s=digphn
c=IN IP4 10.24.16.37
t=0 0
a=X-nat:0
m=audio 4046 RTP/AVP 0 8 9 111 18 58 118 58 96
a=rtcp:4047 IN IP4 10.24.16.37
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:9 G722/8000
a=rtpmap:111 G726-32/8000
a=rtpmap:18 G729/8000
a=rtpmap:58 L16/16000
a=rtpmap:118 L16/8000
a=rtpmap:58 L16-256/16000
a=sendrecv
a=rtpmap:96 telephone-event/8000
a=fmtp:96 0-15


<--- Transmitting SIP response (543 bytes) to UDP:10.24.16.37:5060 --->
SIP/2.0 401 Unauthorized
```

```
Via: SIP/2.0/UDP 10.24.16.37:5060;rport;received=10.24.16.37;branch=z9hG4bKPjCrZnx79augJPtGcTbYlXEs2slZNtwYeC
Call-ID: q.TF2SAaX3jn8dtaLTOCuIO8FRyDCsSR
From: "200" <sip:200@10.24.20.249>;tag=DTD-tYEwFMmbPyu0YWalLQdbEUGSLGN5
To: <sip:201@10.24.20.249>;tag=z9hG4bKPjCrZnx79augJPtGcTbYlXEs2slZNtwYeC
CSeq: 9776 INVITE
WWW-Authenticate: Digest
realm="asterisk",nonce="1413305427/8dd1b7f56aba97da45754f7052d8a688",opaque="0b5a53ab6484480a",algorithm=md5,qop="auth"
Content-Length:  0


<--- Received SIP request (370 bytes) from UDP:10.24.16.37:5060 --->
ACK sip:201@10.24.20.249 SIP/2.0
Via: SIP/2.0/UDP 10.24.16.37:5060;rport;branch=z9hG4bKPjCrZnx79augJPtGcTbYlXEs2slZNtwYeC
Max-Forwards: 70
From: "200" <sip:200@10.24.20.249>;tag=DTD-tYEwFMmbPyu0YWalLQdbEUGSLGN5
To: <sip:201@10.24.20.249>;tag=z9hG4bKPjCrZnx79augJPtGcTbYlXEs2slZNtwYeC
```

```
Call-ID: q.TF2SAaX3jn8dtaLTOCuIO8FRyDCsSR
CSeq: 9776 ACK
Content-Length:  0
```

At first glance, it would appear that the incoming call was challenged for authentication, and that 200 then failed to authenticate on the second INVITE sent. The actual problem here is that the endpoint 200 does not exist within Asterisk. Whenever a SIP request arrives and Asterisk cannot match the request to a configured endpoint, Asterisk will respond to the request with a 401 Unauthorized response. The response will contain a WWW-Authenticate header to make it look as though Asterisk is requesting authentication. Since no endpoint was actually matched, the authentication challenge created by Asterisk is just dummy information and is actually impossible to authenticate against.

The reason this is done is to prevent an information leak. Consider an attacker that sends SIP INVITEs to an Asterisk box, each from a different user. If the attacker happens to send a SIP INVITE from a user name that matches an actual endpoint on the system, then Asterisk will respond to that INVITE with an authentication challenge using that endpoint's authentication credentials. But what happens if the attacker sends a SIP INVITE from a user name that does not match an endpoint on the system? If Asterisk responds differently, then Asterisk has leaked information by responding differently. If Asterisk sends a response that looks the same, though, then the attacker is unable to easily determine what user names are valid for the Asterisk system.

So if you are seeing what appears to be authentication problems, the first thing you should do is to read the Unrecognized Endpoint section above and ensure that the endpoint you think the SIP request is coming from is actually configured properly. If it turns out that the endpoint is configured properly, here are some trouble-shooting steps to ensure that authentication is working as intended:

- Ensure that username and password in the `type = auth` section are spelled correctly and that they are **using the correct case**. If you have "Alice" as the username on your phone and "alice" as the username in Asterisk, things will go poorly.
- If you are using the `md5_cred` option in an auth section, ensure the following:
    - Ensure that you have set `auth_type = md5`.
    - Ensure that the calculated MD5 sum is composed of *username:realm:password*
    - Ensure that the calculated MD5 sum did not contain any extraneous whitespace, such as a newline character at the end.
    - Ensure there were no copy-paste errors. An MD5 sum is exactly 32 hexadecimal characters. If the option in your config file contains fewer or greater than 32 characters, or if any of the characters are not hexadecimal characters, then the MD5 sum is invalid.
- Ensure that you have specified a `username`. Asterisk does not imply a username based on the name of the auth section.
- Ensure that the configured `realm` is acceptable. In most cases, simple SIP devices like phones will authenticate to whatever realm is presented to them, so you do not need to configure one explicitly. However, if a specific realm is required, be sure it is configured. Be sure that if you are using the `md5_cred` option that this realm name is used in the calculation of the MD5 sum.
- Ensure that the endpoint that is communicating with Asterisk uses the "Digest" method of authentication and the "md5" algorithm. If they use something else, then Asterisk will not understand and reject the authentication attempt.

Authentication Not Attempted

The opposite problem of authentication failures is that incoming calls are not being challenged for authentication where it would be expected. Asterisk chooses to challenge for authentication if the endpoint from which the request arrives has a configured `auth` option on it. From the CLI, run the `pjsip show endpoint <endpoint name>` command. Below the initial headers should be something like the following:

```
Endpoint:  david/6001                                          Unavailable   0 of inf
     InAuth:  david-auth/david
        Aor:  david                                             10
  Transport:  main-transport            udp      0      0  0.0.0.0:5060
   Identify:  10.24.16.36/32
```

Notice the "InAuth" on the second line of output. This shows that the endpoint's auth is pointing to a configuration section called "david-auth" and that the auth section has a username of "david". If you do not see an "InAuth" specified for the endpoint, then this means that Asterisk does not see that the endpoint is configured for authentication. Check the following:

- Ensure that there is an `auth` line in your endpoint's configuration.
- Ensure that the auth that your endpoint is pointing to actually exists. Spelling is important.
- Ensure that the auth that your endpoint is pointing to has `type = auth` specified in it.

Asterisk cannot find the specified extension

If you are seeing a message like the following on your CLI when you place an incoming call:

```
[2014-10-14 13:22:45.886] NOTICE[1583]: res_pjsip_session.c:1538 new_invite: Call from '201' (UDP:10.24.18.87:5060) to extension
'456789' rejected because extension not found in context 'default'.
```

then it means that Asterisk was not able to direct the incoming call to an appropriate extension in the dialplan. In the case above, I dialled "456789" on the phone that corresponds with endpoint 201. Endpoint 201 is configured with `context = default` and the "default" context in my dialplan does not have an extension "456789".

The NOTICE message can be helpful in this case, since it tells what endpoint the call is from, what extension it is looking for, and in what context it is searching. Here are some helpful tips to be sure that calls are being directed where you expect:

- Be sure that the endpoint has the expected context configured. Be sure to check spelling.
- Be sure that the extension being dialled exists in the dialplan. From the Asterisk CLI, run `dialplan show <context name>` to see the extensions for a particular context. If the extension you are dialing is not listed, then Asterisk does not know about the extension.

- Ensure that if you have modified `extensions.conf` recently that you have saved your changes and issued a `dialplan reload` from the Asterisk CLI.
- Ensure that the extension being dialled has a 1 priority.
- Ensure the the extension being dialled is in the expected dialplan context.

ARGH! NAT!

NAT is objectively terrible. Before having a look at this section, have a look at this page to be sure that you understand the options available to help combat the problems NAT can cause.

NAT can adversely affect all areas of SIP calls, but we'll focus for now on how they can negatively affect the ability to allow for incoming calls to be set up. The most common issues are the following:

- Asterisk routes responses to incoming SIP requests to the wrong location.
- Asterisk gives the far end an unroutable private address to send SIP traffic to during the call.

Asterisk sends traffic to unroutable address

The endpoint option that controls how Asterisk routes responses is `force_rport`. By default, this option is enabled and causes Asterisk to send responses to the address and port from which the request was received. This default behavior works well when Asterisk is on a different side of a NAT from the device that is calling in. Since Asterisk presumably cannot route responses to the device itself, Asterisk instead routes the response back to where it received the request from.

Asterisk gives unroutable address to device

By default, Asterisk will place its own IP address into Contact headers when responding to SIP requests. This can be a problem if the Asterisk server is not routable from the remote device. The `local_net`, `external_signaling_address`, and `external_signaling_port` transport options can assist in preventing this. By setting these options, Asterisk can detect an address as being a "local" address and replace them with "external" addresses instead.

Outbound Calls

Asterisk says my endpoint does not exist

If you see a message like the following:

```
[2014-10-14 15:50:50.407] ERROR[2004]: chan_pjsip.c:1767 request: Unable to create PJSIP channel - endpoint 'hammerhead' was not
found
```

then this means that Asterisk thinks the endpoint you are trying to dial does not exist. For troubleshooting tips about how to ensure that endpoints are loaded as expected, check the Identify by User subsection in the Incoming Calls section.

Alternatively, if you see a message like the following:

```
[2014-10-14 15:55:06.292] ERROR[2578][C-00000000]: netsock2.c:303 ast_sockaddr_resolve: getaddrinfo("hammerhead", "(null)", ...):
Name or service not known
[2014-10-14 15:55:06.292] WARNING[2578][C-00000000]: chan_sip.c:6116 create_addr: No such host: hammerhead
[2014-10-14 15:55:06.292] DEBUG[2578][C-00000000]: chan_sip.c:29587 sip_request_call: Cant create SIP call - target device not
registered
```

or

```
[2014-10-14 15:55:58.440] WARNING[2700][C-00000000]: channel.c:5946 ast_request: No channel type registered for 'SIP'
[2014-10-14 15:55:58.440] WARNING[2700][C-00000000]: app_dial.c:2431 dial_exec_full: Unable to create channel of type 'SIP'
(cause 66 - Channel not implemented)
```

then it means that your dialplan is referencing "SIP/hammerhead" instead of "PJSIP/hammerhead". Change your dialplan to refer to the correct channel driver, and don't forget to `dialplan reload` when you are finished.

Asterisk cannot route my call

If Asterisk is finding your endpoint successfully, it may be that Asterisk has no address information when trying to dial the endpoint. You may see a message like the following:

```
[2014-10-14 15:58:06.690] WARNING[2743]: res_pjsip/location.c:155 ast_sip_location_retrieve_contact_from_aor_list: Unable to
determine contacts from empty aor list
[2014-10-14 15:58:06.690] WARNING[2834][C-00000000]: app_dial.c:2431 dial_exec_full: Unable to create channel of type 'PJSIP'
(cause 3 - No route to destination)
```

If you see this, then the endpoint you are dialling either has no associated address of record (AoR) or the associated AoR does not have any contact URIs bound to it. AoRs are necessary in order to determine the appropriate destination of the call. To see if your endpoint has an associated AoR, run `pjsip show endpoint <endpoint name>` from the Asterisk CLI. The following is sample output of an endpoint that **does** have an AoR configured on it:

```
Endpoint:  david/6001                                           Unavailable   0 of inf
     InAuth:  david-auth/david
        Aor:  david                                             10
  Transport:  main-transport            udp      0      0  0.0.0.0:5060
   Identify:  10.24.16.36/32
```

Notice the third line. The endpoint points to the AoR section called "david". If your endpoint does not have an AoR associated with it, this third line will be absent.

If you think you have associated your endpoint with an AoR, but one does not appear in the CLI, then here are some troubleshooting steps:

- Ensure that you have set the `aors` option on the endpoint. Notice that the option is not `aor` (there is an 's' at the end).
- Ensure that the AoR pointed to by the `aors` option exists. Check your spelling!

If those appear to be okay, it may be that there was an error when attempting to load the AoR from configuration. From the Asterisk CLI, run the command `pjsip show aor <aor name>`. If you see a message like

```
Unable to find object heman.
```

Then it means the AoR did not get loaded properly. Here are some troubleshooting steps to ensure your AoR is configured correctly:

- Ensure that your AoR has `type = aor` set on it.
- Ensure that there were no nonexistent configuration options set. You can check the logs at Asterisk startup to see if there were any options Asterisk did not understand. For instance, you may see something like:

  ```
  [2014-10-14 16:16:20.658] ERROR[2939]: config_options.c:710 aco_process_var: Could not find option suitable for category
  '1000' named 'awesomeness' at line 219 of
  [2014-10-14 16:16:20.659] ERROR[2939]: res_sorcery_config.c:275 sorcery_config_internal_load: Could not create an object
  of type 'aor' with id '1000' from configuration file 'pjsip.conf'
  ```

  In this case, I tried to set an option called "awesomeness" on the AoR 1000. Since Asterisk did not recognize this option, AoR 1000 was unable to be loaded.
- The `contact` option can be a pitfall. There is an object type called "contact" that is documented on the wiki, which may make you think that the AoR option should point to the name of a contact object that you have configured. On the contrary, the `contact` option for an AoR is meant to be a SIP URI. The resulting contact object will be created by Asterisk based on the provided URI. Make sure when setting the `contact` that you use a full SIP URI and not just an IP address.

Another issue you may encounter is that you have properly configured an AoR on the endpoint but that this particular AoR has no contact URIs bound to it. From the CLI, run the `pjsip show aor <aor name>` command to see details about the AoR. Here is an example of an AoR that has a contact URI bound to it.

```
Aor:  201                                                  1
    Contact:  201/sip:201@10.24.18.87:5060;ob                    Unknown               nan
```

The "Contact:" line shows the URI "sip:201@10.24.18.87:5060;ob" is bound to the AoR 201. If the AoR does not have any contacts bound to it, then no Contact lines would appear. The absence of Contact lines can be explained by any of the following:

- If the device is expected to register, then it may be that the device is either not properly configured or that there was a registration failure. See the Inbound Registrations section for details on how to resolve that problem.
- If the device is not intended to register, then the AoR needs to have a `contact` option set on it. See the previous bulleted list for possible `contact`-related pitfalls.

ARGH! NAT! (Part 2)

NAT makes babies cry.

For outbound calls, the main NAT issue you are likely to come across is Asterisk publishing an unroutable private address in its Contact header. If this is an issue you are facing, this can be corrected by setting the `local_net`, `external_signaling_address`, and `external_signaling_port` options for the transport you are using when communicating with the endpoint. For more information on how this can be set up, please see this page.

Bridged Calls

Direct media is not being used

Direct media is a feature that allows for media to bypass Asterisk and flow directly between two endpoints. This can save resources on the Asterisk system and allow for more simultaneous calls. The following conditions are required for direct media. If any are not met, then direct media is not possible:

- There must only be two endpoints involved in the call.
- Both endpoints involved in the call must have the `direct_media` option enabled.
- The call must be a regular person-to-person call. Calls through ConfBridge() and Meetme() cannot use direct media.
- The sets of codecs in use by each endpoint during the call must have a non-empty intersection. In other words, each endpoint must be using at least one codec that the other endpoint is using.

- Any features in Asterisk that manipulate, record, or inject media may not be used. This includes:
    - The Monitor() and Mixmonitor() applications
    - The Chanspy() application
    - The JACK() application
    - The VOLUME() function
    - The TALK_DETECT() function
    - The SPEEX() function
    - The PERIODIC_HOOK() function
    - The 'L' option to the Dial() application
    - An ARI snoop
    - A jitter buffer
    - A FAX gateway
- No features that require that Asterisk intercept DTMF may be used. This includes the T, t, K, k, W, w, X, and x options to the Dial() application.
- If either endpoint has the `disable_direct_media_on_nat` option set, and a possible media NAT is detected, then direct media will not be used. This option is disabled by default, so you would have to explicitly set this option for this to be a problem.
- The two endpoints must be in the same bridge with each other. If the two endpoints are in separate bridges, and those two bridges are connected with one or more local channels, then direct media is not possible.

Double-check that all requirements are met. Unfortunately, Asterisk does not provide much in the way of debug for determining why it has chosen **not** to use direct media.

Inbound Registrations

For inbound registrations, a lot of the same problems that can happen on inbound calls may occur. Asterisk goes through the same endpoint identification and authentication process as for incoming calls, so if your registrations are failing for those reasons, consult the troubleshooting guide for incoming calls to determine what the problem may be.

If your problem is not solved by looking in those sections, then you may have a problem that relates directly to the act of registering. Before continuing, here is a sample REGISTER request sent to an Asterisk server:

```
REGISTER sip:10.24.20.249:5060 SIP/2.0
Via: SIP/2.0/UDP 10.24.16.37:5060;rport;branch=z9hG4bKPj.rPtUH-P33vMFd68cLZjQj0QQxdu6mNx
Max-Forwards: 70
From: "200" <sip:200@10.24.20.249>;tag=BXs-nct8-XOe7Q7tspK3Vl3iqUa0cmzc
To: "200" <sip:200@10.24.20.249>
Call-ID: C0yYQJ8h776wbheBiUEqCin.ZhcBB.tZ
CSeq: 5200 REGISTER
User-Agent: Digium D40 1_4_0_0_57389
Contact: "200" <sip:200@10.24.16.37:5060;ob>
Expires: 300
Allow: PRACK, INVITE, ACK, BYE, CANCEL, UPDATE, SUBSCRIBE, NOTIFY, REFER, MESSAGE, OPTIONS
Content-Length:  0
```

This REGISTER was sent by the endpoint 200. The URI in the To header is "sip:200@10.24.20.249". Asterisk extracts the username portion of this URI to determine the address of record (AoR) that the REGISTER pertains to. In this case, the AoR has the same name as the endpoint, 200. The URI in the Contact header is "sip:200@10.24.16.37:5060;ob". The REGISTER request is attempting to bind this contact URI to the AoR. Ultimately, what this means is that when someone requests to reach endpoint 200, Asterisk will check the AoRs associated with the endpoint, and send requests to all contact URIs that have been bound to the AoR. In other words, the REGISTER gives Asterisk the means to locate the endpoint.

You can ensure that your configuration is sane by running the the `pjsip show endpoint <endpoint name>` CLI command. Part of the output is to show all AoRs associated with a particular endpoint, as well as contact URIs that have been bound to those AoRs. Here is sample output from running `pjsip show endpoint 200` on a system where registration has succeeded:

```
Endpoint:  200/200                                              Not in use    0 of inf
        Aor:  200                                                1
      Contact:  200/sip:200@10.24.16.37:5060;ob                  Unknown               nan
  Transport:  main-transport            udp      0      0  0.0.0.0:5060
```

This shows that endpoint 200 has AoR 200 associated with it. And you can also see that the contact "sip:200@10.24.16.37:5060;ob" has been bound to the AoR.

If running this command shows no AoR, then ensure that the endpoint has the `aors` option set. Note that the name is `aors`, not `aor`.

More likely, the issue will be that an AoR will be listed, but there will be no associated contact. If this is the case, here are some possible troubleshooting steps:

- Ensure that the AoR has actually been loaded. Run the CLI command `pjsip show aor <aor name>`. If no AoR is displayed, then that means the AoR was not loaded.
    - Ensure that the configuration section has `type = aor` specified.
    - Ensure that all configuration options specified on the AoR are options that Asterisk recognizes.
- Ensure that the `res_pjsip_registrar.so` module is loaded and running. Running `module show like res_pjsip_registrar.so` should show the following:

```
Module                         Description                              Use Count  Status      Support Level
res_pjsip_registrar.so         PJSIP Registrar Support                  0          Running              core
```

- Ensure that the AoR has a `max_contacts` value configured on it. If this option is not set, then registration cannot succeed. You will see this message on the CLI:

  ```
  [2014-10-16 11:34:07.887] WARNING[2940]: res_pjsip_registrar.c:685 registrar_on_rx_request: AOR '200' has no configured
  max_contacts. Endpoint '200' unable to register
  ```

  Asterisk will transmit a 403 Forbidden in response to the registration attempt.

If you initially have successful registrations but they later start failing, then here are some further troubleshooting steps for you to try:

- If you intend for new registrations to replace old ones, then enable the `remove_existing` option for the AoR.
- Ensure that if you are attempting to bind multiple contacts to an AoR that the `max_contacts` for the AoR is large enough. If the `max_contacts` value is not high enough, you will see the following CLI message:

  ```
  [2014-10-16 11:34:07.887] WARNING[2940]: res_pjsip_registrar.c:411 rx_task: Registration attempt from endpoint '200' to
  AOR '200' will exceed max contacts of 1
  ```

  Asterisk will respond to the registration attempt with a 403 Forbidden.

Outbound Registrations

If you are having troubles with outbound registrations and unfamiliar with the mechanics involved, please see this page. It will explain quite a few of the concepts that Asterisk uses and may give you some clues for solving your issue.

If you are still having trouble, here are some troubleshooting steps:

- If Asterisk is not sending an outbound REGISTER at all, then it is likely that there was an error when trying to load the outbound registration.
  - Ensure that the outbound registration has `type = registration` in it.
  - Ensure that there are no configuration options that Asterisk does not recognize.
- Another reason Asterisk may not be sending an outbound REGISTER is that you do not have a valid SIP URI for your `server_uri` or `client_uri`. You may see a message like this on the CLI if this is the case:

  ```
  [2014-10-16 12:05:16.064] ERROR[3187]: res_pjsip_outbound_registration.c:724 sip_outbound_registration_regc_alloc:
  Invalid server URI 'registrar@example.com' specified on outbound registration 'outreg'
  ```

  In this case, I left off the initial "sip:" from the URI.
- If your outbound REGISTER receives no response, then you may have misconfigured the `server_uri` to point somewhere the REGISTER is not meant to be sent.
- If Asterisk has stopped sending REGISTER requests, then either the maximum number of retries has been attempted or the response that Asterisk received from the registrar was considered to be a permanent failure. If you want to get Asterisk to start sending REGISTER requests again after making configuration adjustments, you can do so by running the `module reload res_pjsip_registrar.so` CLI command.

Inbound Subscriptions

The first thing to acknowledge with inbound subscriptions is that the handling of the inbound SUBSCRIBE messages starts the same as for inbound calls. This means that if you are having troubles where Asterisk does not recognize the endpoint sending the SUBSCRIBE or if authentication is failing, you should check the troubleshooting guide for incoming calls for details on how to solve these issues.

It is also important to ensure that `res_pjsip_pubsub.so` is loaded and running. This module is the core of all of Asterisk's handling of subscriptions, and if it is not loaded, then Asterisk will not be able to set up subscriptions properly.

Presence/Dialog Info

A tutorial about subscribing to presence and dialog-info can be found on this page. Reading that page may point you towards how to resolve the issue you are facing.

If you are attempting to subscribe to the presence or dialog event packages, then here are some troubleshooting steps for determining what is going wrong.

- Ensure that the `res_pjsip_exten_state.so` module is loaded.
- Ensure that the Event header in inbound subscribe messages are one of "presence" or "dialog".
- Ensure all necessary modules are loaded, depending on what values are in the Accept header of inbound SUBSCRIBE requests.
  - Subscriptions that use Accept: application/pidf+xml will need to have `res_pjsip_pidf_body_generator.so` loaded.
  - Subscriptions that use Accept: application/xpidf+xml will need to have `res_pjsip_xpidf_body_generator.so` loaded.
  - Subscriptions that use Accept: application/dialog-info+xml will need to have `res_pjsip_dialog_info_body_generator.so` loaded.
- When subscribing, you may see a message like the following on the CLI:

  ```
  [2014-10-16 12:56:58.605] WARNING[3780]: res_pjsip_exten_state.c:337 new_subscribe: Extension blah does not exist or has
  no associated hint
  ```

The warning message is self-explanatory. If you think you have placed extension "blah" in your `extensions.conf` file and it contains a hint, then be sure that it exists in the same context as the `context` option on the endpoint that is attempting to subscribe. Also be sure that if you have recently changed your `extensions.conf` file that you have saved the changes and run the `dialplan reload` CLI command.

MWI

If you are attempting to subscribe to the message-summary package, then here are some troubleshooting steps for determining what is going wrong.

- Ensure that the `res_pjsip_mwi.so` and the `res_pjsip_mwi_body_generator.so` modules are loaded.
- Ensure that the AoR that the MWI SUBSCRIBE is being sent to has `mailboxes` configured on it. Note that the option name is `mailboxes` and not `mailbox`.
- When subscribing to MWI, you may see a message like the following:

  ```
  [2014-10-16 13:06:51.323] NOTICE[3963]: res_pjsip_mwi.c:566 mwi_validate_for_aor: Endpoint '200' already configured for
  unsolicited MWI for mailbox '200'. Denying MWI subscription to 200
  ```

  The most likely cause of something like this is that you have an endpoint and an AoR that both have `mailboxes = 200` in your configuration. The endpoint with `mailboxes = 200` attempts to subscribe to the AoR that has `mailboxes = 200`. In this case, since Asterisk is already sending MWI notifications about mailbox 200 to the endpoint, the subscription to the AoR is denied. To fix this, remove the `mailboxes` option from your endpoint, or configure your device not to attempt to subscribe to MWI.
- Asterisk has multiple ways of having MWI state set, but the most common way is to use `app_voicemail` that comes with Asterisk. `app_voicemail` has a requirement that mailbox names must follow the format "mailbox@context". If you are using `app_voicemail` and you configure MWI in `pjsip.conf` and only provide the mailbox name without a context, then you will not receive MWI updates when the state of the mailbox changes.

Configuration Issues

Can't create an IPv6 transport

You've configured a transport in pjsip.conf to bind to an IPv6 address or block. However, Asterisk fails to create the transport when loading!

If you look into your logs you might messages similar to the following:

```
[Dec 12 00:58:31] ERROR[10157] config_options.c: Error parsing bind=:: at line 8 of
[Dec 12 00:58:31] ERROR[10157] res_sorcery_config.c: Could not create an object of type 'transport' with id 'my-ipv6-transport'
from configuration file 'pjsip.conf'
```

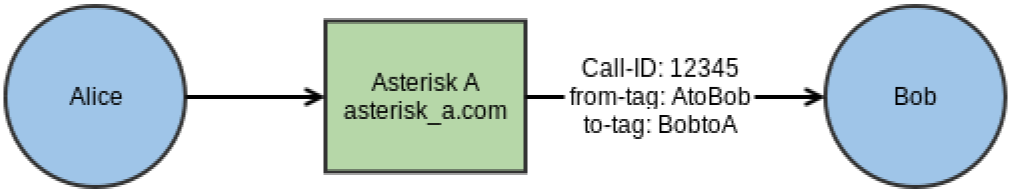
The most likely issue is that you have not compiled **pjproject** with support for IPv6. You can find instructions at PJSIP-pjproject.

res_pjsip Remote Attended Transfers

What is a remote SIP transfer?

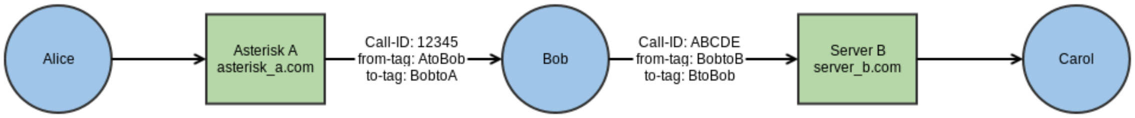
Let's imagine a scenario where Alice places a call to Bob, and then Bob performs an attended transfer to Carol. In this scenario, Alice is registered to Asterisk instance A (asterisk_a.com), and Bob is registered to Server B (server_b.com), a non-Asterisk PBX. The key to this scenario is that Asterisk A has been explicitly configured to be able to call Bob directly, despite the fact that Bob does not register to Asterisk A.

Initially, Alice places a call to Bob through Alice's Asterisk instance:



The arrows indicate the direction of the initial call. The Call-ID, from-tag, and to-tag will become important later.

Now, Bob wants to perform an attended transfer to Carol, so he places a call to Carol:



As you can see, Bob has simultaneous calls through two separate servers. Now when Bob performs the attended transfer, what happens? Bob will send a SIP REFER request to either Asterisk A or Server B to get the two SIP servers in contact with each other. Most phones will send the REFER to Asterisk A since it is the original leg of the call, so that is what we will do in our scenario. The REFER request has a Refer-To header that specifies details of the transfer. The Refer-To header for this particular transfer looks like the following:

```
Refer-To: <sip:carol@server_b.com?Replaces=ABCDE%3Bto-tag%3DBtoBobfrom-tag%3DBobtoB>
```

That's a bit verbose. So let's break it down a little bit. First, there is a SIP URI:

```
sip:carol@server_b.com
```

Next, there is a Replaces URI header. There are some URL-escaped character sequences in there. If we decode them, we get the following:

```
Replaces: ABCDE;to-tag=BtoBob;from-tag=BobtoB
```

If we break down the parts of this, what the Replaces section tells us is that the REFER request is saying that the SIP dialog with Call-ID "ABCDE", to-tag "BtoBob" and from-tag "BobtoB" needs to be replaced by the party (or parties) that Bob is talking to.

Asterisk has built into it a bit of an optimization to avoid unnecessary SIP traffic by looking up the dialog referred to by the Replaces header. If the dialog is found in the Asterisk system, then Asterisk simply performs a local attended transfer. This involves internal operations such as moving a channel from one bridge to another, or creating a local channel to link two bridges together.

However, in this case, the dialog referred to by Bob's Replaces header is not on Asterisk A. It is on Server B. So Asterisk A cannot perform a local attended transfer. This is where a remote attended transfer is required.

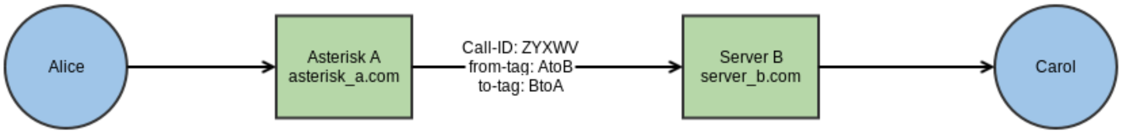
From a SIP point of view

Remote attended transfers are the type of attended transfers referred to in SIP specifications, such as RFC 5589 section 7. When a SIP user agent receives a REFER request, the user agent is supposed to send an INVITE to the URI in the Refer-To header to start a new call with the user agent at that URI. The INVITE should have a Replaces header that has the same contents as the Replaces URI header from the REFER request. This tells the user agent that receives the INVITE to replace the referenced dialog with this new call instead.

In the scenario above, when Asterisk A receives the REFER request from Bob, Asterisk A should respond by sending an INVITE to `sip:carol@server_b.com` and add

```
Replaces: ABCDE;to-tag=BtoBob;from-tag=BobtoB
```

When Server B receives this INVITE, it will essentially swap this new call in for the call referenced by the Replaces header. By doing this, the final picture looks something like the following:



A new dialog with Call-ID ZYXWV has replaced the previous dialog with Call-ID ABCDE. The previously-illustrated dialog between Bob and Asterisk A with Call-ID 12345 is gone because Bob's phone ended that dialog once the transfer was completed.

How Asterisk handles this

Asterisk will rarely ever directly place outbound calls without going through the dialplan. When Asterisk A receives the REFER request from Bob, Asterisk does not immediately send an INVITE with Replaces header to Server B. Instead, Asterisk A looks for a specifically-named extension called "external_replaces". Asterisk searches for this extension in the context specified by the `TRANSFER_CONTEXT` channel variable on Bob's channel. If `TRANSFER_CONTEXT` is not specified, then Asterisk searches for the extension in Bob's endpoint's context setting. Once in the dialplan, it is the job of the dialplan writer to determine whether to complete the transfer or not.

In the external_replaces extension, you will have access to the following channel variables:

- `SIPTRANSFER`: This variable is set to "yes" upon entering the `external_replaces` extension, and indicates that a SIP transfer is happening. This is only useful if, for whatever reason, you are using the `external_replaces` extension for additional purposes than a SIP remote attended transfer.
- `SIPREFERRINGCONTEXT`: This is the dialplan context in which the `external_replaces` extension was found. This may be useful if your `external_replaces` extension calls into subroutines or jumps to other contexts.
- `SIPREFERTOHDR`: This is the SIP URI in the Refer-To header in the REFER request sent by the transferring party.

The big reason why Asterisk calls into the dialplan instead of automatically sending an INVITE to the Refer-To URI is for security purposes. If Asterisk automatically sent an INVITE out without going through the dialplan, there are chances that transfers could be used to place calls to unwanted destinations that could, for instance, charge you a lot of money for the call.

Writing your `external_replaces` extension

Now that the theory has been presented, you'll need to write your `external_replaces` extension. One option you have is to not write an `external_replaces` extension at all. This will prevent any remote attended transfers from succeeding.

If you do want to write an `external_replaces` extension, the first thing you want to do is determine if you want to perform the remote attended transfer. `SIPREFERTOHDR`, and values provided by the `CHANNEL()` dialplan function can help you to decide if you want to allow the transfer. For instance, you might use `CHANNEL(endpoint)` to see which PJSIP endpoint is performing the transfer, and you can inspect `SIPREFERTOHDR` to determine if the transfer is destined for a trusted domain.

> ⚠ Asterisk dialplan contains functions for manipulating strings. A function Asterisk 13 Function_PJSIP_PARSE_URI exists for parsing a URI within the dialplan.

If you decide not to perform the transfer, the simplest thing to do is to call the `Hangup()` application.

> ⚠ Calling `Hangup()` in this situation can have different effects depending on what type of phone Bob is using. Asterisk updates the phone with a notification that the attended transfer failed. It is up to the phone to decide if it wants to try to reinvite itself back into the original conversation with Alice or simply hang up.

If you decide to perform the transfer, the most straightforward way to do this is with the `Dial()` application. Here is an example of how one might complete the transfer

```
exten => external_replaces,1,NoOp()
 same => n,Dial(PJSIP/default_outgoing/${SIPREFERTOHDR}
```

Let's examine that `Dial()` more closely. First, we're dialing using PJSIP, which is pretty obvious. Next, we have the endpoint name. The endpoint name could be any configured endpoint you want to use to make this call. Remember that endpoint settings are things such as what codecs to use, what user name to place in the from header, etc. By default, if you just dial `PJSIP/some_endpoint`, Asterisk looks at some_endpoint's configured `aors` to determine what location to send the outgoing call to. However, you can override this default behavior and specify a URI to send the call to instead. This is what is being done in this `Dial()` statement. We're dialing using settings for an endpoint called "default_outgoing", presumably used as a default endpoint for outgoing calls. We're sending the call out to the URI specified by `SIPREFERTOHDR` though. Using the scenario on this page, the `Dial()` command would route the call to `sip:carol@server_b`.

Avoiding Remote Attended Transfers

In Asterisk, remote attended transfers are sometimes necessary, but avoiding them is typically a good idea. The biggest reason is the security concerns of allowing users to make calls to untrusted domains.

The easiest but most severe way to prevent remote attended transfers is to set `allow_transfer = no` for all endpoints. The problem with doing this is that it also prevents local attended transfers and blind transfers.

A second way has been discussed already, and that is not to write an `external_replaces` extension. This prevents any attempted remote attended transfers from succeeding, but it does not help to prevent the remote attended transfer from happening in the first place.

Another way is to configure your Asterisk server to only call phones that are directly registered to it and trusted SIP servers. In the scenario we have been inspecting, the remote attended transfer could have been avoided by having Asterisk A call Bob through Server B instead of dialing Bob directly. By receiving the initial call through Server B, Bob will send his REFER request to Server B, who being aware of all necessary dialogs, may be able to perform a local attended transfer (assuming it can do the same optimization as Asterisk). Configuring Asterisk this way is not necessarily guaranteed to prevent all remote attended transfer attempts, but it will help to lessen them.

PJSIP Transport Selection

Sending Messages

The process by which an underlying transport is chosen for sending of a message is broken up into different steps depending on the type of message.

SIP Request Handling

1. URI Parsing

The PJSIP stack fundamentally acts on URIs. When sending to a URI it is parsed into the various parts (user, host, port, user parameters). For the purposes of transport selection the transport parameter is examined. This specifies the type of transport. If this parameter is not present it is assumed to be UDP. This is important because it is used in DNS resolution. If a "sips" URI scheme is used an automatic switchover to TLS will occur.

2. DNS SRV Resolution (If host portion is not an IP address and no port is present in the URI)

The transport type from above is used to determine which SRV record to look up. This means that the original URI **must** include the transport type for TCP and TLS types UNLESS the "sips" URI scheme is used which automatically switches to TLS.

3a. Transport Selection (No explicit transport provided)

Now that the underlying type of transport is known and the resolved target exists the transport selection process can begin.

Connection-less protocols (such as UDP)

A transport, decided upon by a hashing mechanism, matching the transport type and address family is selected.

Connection-oriented protocols (such as TCP or TLS)

An already open connection to the resolved IP address and port is searched for. If the connection exists it is reused for the request. If no connection exists the first transport matching the transport type and address family as configured in pjsip.conf is chosen. It is instructed to establish a new connection to the resolved IP address and port.

On this Page

3b. Transport Selection (Explicit transport provided)

Connection-less protocols (such as UDP)

The provided transport is used.

Connection-oriented protocols (such as TCP or TLS)

The provided transport is instructed to establish a new connection to the resolved IP address and port.

> ⓘ If an existing connection exists to the IP address and port using the specific transport type then it is reused and a new one is not established.

4. Multihomed Transport Selection (Connection-less protocols)

Before the message is sent out the transport the routing table is queried to determine what interface it will be going out on.

Local source interface IP address matches source IP address in message

The message is left untouched and passed to the transport.

Local source interface IP address differs from source IP address in message

The message contents are updated with the different source address information. If a transport is bound to the new source address the outgoing transport for the message is changed to it.

5. Message is sent

The message is provided to the transport and it is instructed to send it.

SIP Response Handling

1. Transport Selection

Connection-oriented protocols (such as TCP or TLS)

If the connection the request was received on is still open it is used to send the response.

If no connection exists or the connection is no longer open the first configured transport in pjsip.conf matching the transport type and address family is selected. It is instructed to establish a new connection to the destination IP address and port.

Connection-less protocol with maddr in URI of the topmost Via header

A transport, decided upon by a hashing mechanism, matching the transport type and address family is selected. The transport type and address family of the transport the request was received on is used.

Connection-less protocol with rport in URI of the topmost Via header

The transport the request is received on is used as the transport for the response.

Connection-less protocol without rport in URI of the topmost Via header

A transport, decided upon by a hashing mechanism, matching the transport type and address family is selected. The transport type and address family of the transport the request was received on is used.

2. Message is sent

The message is provided to the selected transport and it is instructed to send it.

Best Configuration Strategies

IPv4 Only (Single Interface)

Configure a wildcard transport. This is simple as it requires no special configuration such as knowing the IP address and has no downsides.

```
[system-udp]
type=transport
protocol=udp
bind=0.0.0.0


[system-tcp]
type=transport
protocol=tcp
bind=0.0.0.0


[system-tls]
type=transport
protocol=tls
bind=0.0.0.0:5061
cert_file=certificate


[phone]
type=endpoint
```

This example includes an endpoint without a transport explicitly defined. Since there is only one transport configured for each address family and transport type each respective one will be used depending on the URI dialed. For requests to this endpoint the logic in section 3a will be used.

IPv4 Only (Multiple Interfaces)

Configure a transport for each interface. This allows each transport to be specified on endpoints and also ensures that the SIP messages contain the right information.

```
[system-internet-udp]
type=transport
protocol=udp
bind=5.5.5.5


[system-internet-tcp]
type=transport
protocol=tcp
bind=5.5.5.5

[system-internet-tls]
type=transport
protocol=tls
bind=5.5.5.5:5061
cert_file=certificate


[system-local-udp]
type=transport
protocol=udp
bind=192.168.1.1


[system-local-tcp]
type=transport
protocol=tcp
bind=192.168.1.1


[system-local-tls]
type=transport
protocol=tls
bind=192.168.1.1:5061
cert_file=certificate

[phone-internet]
type=endpoint
transport=system-internet-udp


[phone-local]
type=endpoint
transport=system-local-udp



[phone-unspecified]
type=endpoint
```

This example includes three endpoints which are each present on different networks. To ensure that outgoing requests to the first two endpoints travel over the correct transport the transport has been explicitly specified on each. For requests to these endpoints the logic in section 3b will be used. For requests to the "phone-unspecified" endpoint since no transport has been explicitly specified the logic in section 3a will be used.

IPv6 Only (Single Interface)

Configure a transport with the IPv6 address:

```
[system-udp6]
type=transport
protocol=udp
bind=[2001:470:e20f:42::42]


[system-tcp6]
type=transport
protocol=tcp
bind=[2001:470:e20f:42::42]
```

IPv4+IPv6 Combined (Single Interface)

Configure two transports, one with the IPv4 address and one with the IPv6 address.

```
[system-udp]
type=transport
protocol=udp
bind=192.168.1.1


[system-tcp]
type=transport
protocol=tcp
bind=192.168.1.1


[system-udp6]
type=transport
protocol=udp
bind=[2001:470:e20f:42::42]


[system-tcp6]
type=transport
protocol=tcp
bind=[2001:470:e20f:42::42]
```

> It might be tempting to use a wildcard IPv6 address to bind a single transport to allow both IPv6 and IPv4. In this configuration IPv6 mapped IPv4 addresses will be used which is unsupported by PJSIP. This will cause a SIP message parsing failure.

Common Issues

Changeover to TCP when sending via UDP

If you turn the "disable_tcp_switch" option off in the pjsip.conf system section it is possible for an automatic switch to TCP to occur when sending a large message out using UDP. If your system has not been configured with a TCP transport this will fail. The sending of the message may also fail if the remote side is not listening on TCP.

Sending using a transport that is not available

If a transport can not be found during the transport selection process you will receive a warning message:

```
Failed to send Request msg INVITE/cseq=7846 (tdta0x7fa920002e50)! err=171060 (Unsupported
transport (PJSIP_EUNSUPTRANSPORT))
```

This can occur due to using a transport type (such as TCP) or address family when a transport meeting the requirements does not exist.

PJSIP Configuration Wizard

The PJSIP Configuration Wizard (module `res_pjsip_config_wizard`) is a new feature in Asterisk 13.2.0. While the basic `chan_pjsip` configuration objects (endpoint, aor, etc.) allow a great deal of flexibility and control they can also make configuring standard scenarios like `trunk` and `user` more complicated than similar scenarios in `sip.conf` and `users.conf`. The PJSIP Configuration Wizard aims to ease that burden by providing a single object called 'wizard' that be used to configure most common `chan_pjsip` scenarios.

The following configurations demonstrate a simple ITSP scenario.

pjsip_wizard.conf

```
[my-itsp]
type = wizard
sends_auth = yes
sends_registrations = yes
remote_hosts = sip.my-itsp.net
outbound_auth/username = my_username
outbound_auth/password = my_password
endpoint/context = default
aor/qualify_frequency = 15
```

pjsip.conf

```
[my-itsp]
type = endpoint
aors = my-itsp
outbound_auth = my-itsp-auth
context = default

[my-itsp]
type = aor
contact = sip:sip.my-itsp.net
qualify_frequency = 15

[my-itsp-auth]
type = auth
auth_type = userpass
username = my_username
password = my_password

[my-itsp-reg]
type = registration
outbound_auth = my-itsp-auth
server_uri = sip:sip.my-itsp.net
client_uri = sip:my_username@sip.my-itsp.net

[my-itsp-identify]
type = identify
endpoint = my-itsp
match = sip.my-itsp.net
```

Both produce the same result. In fact, the wizard creates standard `chan_pjsip` objects behind the scenes. In the above example...

- An endpoint and aor are created with the same name as the wizard.
- The `endpoint/context` and `aor/qualify_fequency` parameters are added to them.
- `remote_hosts` captures the remote host for all objects.
- A contact for the aor is created for each remote host.
- `sends_auth=yes` causes an auth object to be created.
- `outbound_auth/username` and `outbound_auth/password` are added to it.
- An `outbound_auth` line is added to the endpoint.
- `sends_registrations=yes` causes a registration object to be created.
- An `outbound_auth` line is added to the registration.
- The `server_uri` and `client_uri` are constructed using the remote host and username.
- An identify object is created and a match is added for each remote host.

Configuration Reference:

Parameter	Description
type	Must be `wizard`

sends_auth	Will create an outbound auth object for the endpoint and registration. At least `outbound/username` must be specified. default = no
accepts_auth	Will create an inbound auth object for the endpoint. At least 'inbound/username' must be specified. default = no
sends_registrations	Will create an outbound registration object for each host in remote_hosts. default = no
remote_hosts	A comma separated list of remote hosts in the form of `<ipaddress | hostname>[:port][, ... ]` If specified, a static contact for each host will be created in the aor. If `accepts_registrations` is no, an identify object is also created with a match line for each remote host. Hostnames must resolve to A, AAAA or CNAME records. SRV records are not currently supported. default = `""`
transport	The transport to use for the endpoint and registrations default = the pjsip default
server_uri_pattern	The pattern used to construct the registration `server_uri`. The replaceable parameter `${REMOTE_HOST}` is available for use. default = `sip:${REMOTE_HOST}`
client_uri_pattern	The pattern used to construct the registration `client_uri`. The replaceable parameters ${REMOTE_HOST} and ${USERNAME} are available for use. default = {{sip:${USERNAME}@${REMOTE_HOST}}}
contact_pattern	The pattern used to construct the aor contact. The replaceable parameter ${REMOTE_HOST} is available for use. default = `sip:${REMOTE_HOST}`
has_phoneprov	Will create a phoneprov object. If yes, both `phoneprov/MAC` and `phoneprov/PROFILE` must be specified. default = no
has_hint	Enables the automatic creation of dialplan hints. Two entries will be created. One hint for 'hint_exten' and one application to execute when 'hint_exten' is dialed.
hint_context	The context into which hints are placed.
hint_exten	The extension this hint will be registered with.
hint_application	An application with parameters to execute when 'hint_exten' is dialed. `Example: Gosub(stdexten,${EXTEN},1(${HINT}))`
<object>/<parameter>	These parameters are passed unmodified to the native object.

Configuration Notes:

- Wizards must be defined in `pjsip_wizard.conf`.
- Using pjsip_wizard.conf doesn't remove the need for pjsip.conf or any other config file.
- Transport, system and global sections still need to be defined in pjsip.conf.
- You can continue to create discrete endpoint, aor, etc. objects in pjsip.conf but there can be no name collisions between wizard created objects and discretely created objects.

- An endpoint and aor are created for each wizard.
    - The endpoint and aor are named the same as the wizard.
    - Parameters are passed to them using the `endpoint/` and `aor/` prefixes.
    - A contact is added to the aor for each remote host using the `contact_pattern` and `${REMOTE_HOST}`.
- `sends_auth` causes an auth object to be created.
    - The name will be `<wizard_name>-oauth`.
    - Parameters are passed to it using the `outbound_auth/` prefix.
    - The endpoint automatically has an `outbound_auth` parameter added to it.
    - Registrations automatically have an `outbound_auth` parameter added to them (if registrations are created, see below).
- `accepts_auth` causes an auth object to be created.
    - The name will be `<wizard_name>-iauth`.
    - Parameters are passed to it using the `inbound_auth/` prefix.
    - The endpoint automatically has an `auth` parameter added to it.
- `sends_registrations` causes an outbound registration object to be created for each remote host.
    - The name will be `<wizard_name>-reg-`<n> where n starts at 0 and increments by 1 for each remote host.
    - Parameters are passed to them using the `registration/` prefix.
    - You should not use a wizard in situations whereyou need to pass different parameters to each registration.
    - `server_uri` and `client_uri` are constructed using their respective patterns using `${REMOTE_HOST}` and `${USERNAME}`.
- If `accepts_registrations` is specified, `remote_hosts` must NOT be specified and no contacts are added to the aor. This causes registrations to be accepted.
- If `accepts_registrations` is NOT specified or set to no, then an identify object is created to match incoming requests to the endpoint.
    - The name will be `<wizard_name>-identify`.
    - Parameters are passed to it using the `identify/` prefix although there really aren't any to pass.
- If `has_phoneprov` is specified, a phoneprov object object is created.
    - The name will be `<wizard_name>-phoneprov`.
    - Both `phoneprov/MAC` and `phoneprov/PROFILE` must be specified.
- `has_hint` causes hints to be automatically created.
    - `hint_exten` must be specified.
- All created objects must pass the same edit criteria they would have to pass if they were specified discretely.
- All created objects will have a `@pjsip_wizard=<wizard_name>` parameter added to them otherwise they are indistinguishable from discretely created ones.
- All created object are visible via the CLI and AMI as though they were created discretely.

Full Examples:

Phones:

Configuration	Notes

```
[user_defaults](!)
type = wizard
transport = ipv4
accepts_registrations = yes
sends_registrations = no
accepts_auth = yes
sends_auth = no
has_hint = yes
hint_context = DLPN_DialPlan1
hint_application =
Gosub(stdexten,${EXTEN},1(${HINT}))
endpoint/context = DLPN_DialPlan1
endpoint/allow_subscribe = yes
endpoint/allow = !all,ulaw,gsm,g722
endpoint/direct_media = yes
endpoint/force_rport = yes
endpoint/disable_direct_media_on_nat = yes
endpoint/direct_media_method = invite
endpoint/ice_support = yes
endpoint/moh_suggest = default
endpoint/send_rpid = yes
endpoint/rewrite_contact = yes
endpoint/send_pai = yes
endpoint/allow_transfer = yes
endpoint/trust_id_inbound = yes
endpoint/device_state_busy_at = 1
endpoint/trust_id_outbound = yes
endpoint/send_diversion = yes
aor/qualify_frequency = 30
aor/authenticate_qualify = no
aor/max_contacts = 1
aor/remove_existing = yes
aor/minimum_expiration = 30
aor/support_path = yes
phoneprov/PROFILE = profile1

[bob](user_defaults)
hint_exten = 1000
inbound_auth/username = bob
inbound_auth/password = bobspassword

[alice](user_defaults)
hint_exten = 1001
endpoint/callerid = Alice <1001>
endpoint/allow = !all,ulaw
inbound_auth/username = alice
inbound_auth/password = alicespassword
has_phoneprov = yes
phoneprov/MAC = deadbeef4dad
```

This example demonstrates the power of both wizards and templates.

Once the template is created, adding a new phone could be as simple as creating a wizard object with nothing more than a username and password. You don't even have to specify 'type' because it's inherited from the template.

Of course, you can override ANYTHING in the wizard object or specify everything and not use templates at all.

Trunk to an ITSP requiring registration:

Configuration	Notes

[trunk_defaults](!)
type = wizard
transport = ipv4
endpoint/allow_subscribe = no
endpoint/allow = !all,ulaw
aor/qualify_frequency = 30
registration/expiration = 1800
 	
[myitsp](trunk_defaults)
sends_auth = yes
sends_registrations = yes
endpoint/context = DID_myitsp
remote_hosts = sip1.myitsp.net,sip2.myitsp.net
accepts_registrations = no
endpoint/send_rpid = yes
endpoint/send_pai = yes
outbound_auth/username = my_username
outbound_auth/password = my_password
 	
[my_other_itsp](trunk_defaults)
sends_auth = yes
sends_registrations = yes
endpoint/context = DID_myitsp
remote_hosts = sip1.my-other-itsp.net,sip2.my-other-itsp.net
accepts_registrations = no
endpoint/send_rpid = yes
endpoint/send_pai = yes
outbound_auth/username = my_username
outbound_auth/password = my_password
registration/expiration = 900
registration/support_path = no
	This is an example of trunks to 2 different ITSPs each of which has a primary and backup server.
	It also shows most of the endpoint and aor parameters being left at their defaults.
	In this scenario, each wizard object takes the place of an endpoint, aor, auth, identify and 2 registrations.

Trunk between trusted peers:

Configuration	Notes

[trusted-peer](trunk_defaults)
endpoint/context = peer_context
remote_hosts = sip1.peer.com:45060
sends_registrations = no
accepts_registrations = no
sends_auth = no
accepts_auth = no
	This one's even simpler. The `sends_` and `accepts_` parameters all default to no so you don't really even have to specify them unless your template has them set to `yes`.

Configuring res_pjsip for IPv6

Tell Asterisk and PJSIP to Speak IPv6

The configuration described here happens in the pjsip.conf file within transport and endpoint sections. For more information about the transport side of things see PJSIP Transport Selection

Bind PJSIP to a specific interface

To configure res_pjsip for communication over an IPv6 interface you must modify the bind address for your transports in pjsip.conf.

```
[transport-udp6]
type=transport
protocol=udp
bind=[fe80::5e26:aff:fe4b:4399]

[transport-tcp6]
type=transport
protocol=tcp
bind=[fe80::5e26:aff:fe4b:4399]
```

Bind PJSIP to the first available IPv6 interface

A transport can be configured to automatically bind to the first available IPv6 interface. You use "::" as the bind address.

```
[transport-auto-ipv6]
type=transport
protocol=udp
bind=::
```

Configure a PJSIP endpoint to use RTP over IPv6

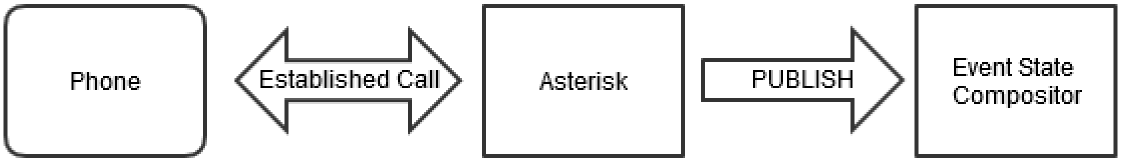
There is no additional configuration required to have an endpoint use RTP over IPv6. IPv4 or IPv6 will be automatically chosen based on the address family of the address for signaling. That is: If an endpoint is using IPv4 it will be IPv4, if it is using IPv6 it will be IPv6.

```
[mytrunk]
type=endpoint
transport=transport-udp6
context=from-external
disallow=all
allow=ulaw
```

Publishing Extension State

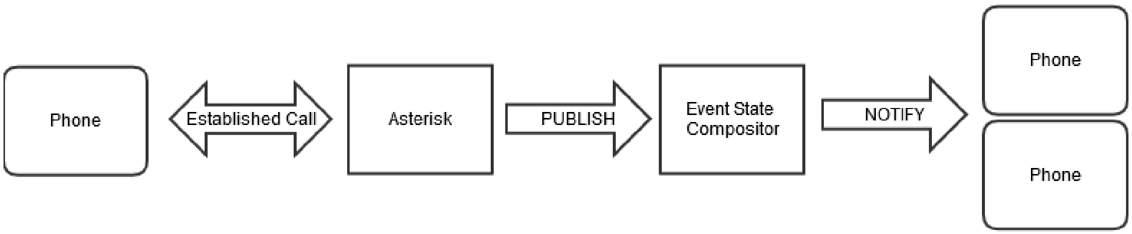
Background

Functionality exists within PJSIP, as of Asterisk 14, that allows extension state to be published to another entity, commonly referred to as an event state compositor. Instead of each device subscribing to Asterisk and receiving a NOTIFY as extension state changes, PJSIP can be configured to send a single PUBLISH request for each extension state change to the other entity. These PUBLISH requests are triggered based on extension state changes made to hints in the dialplan.



Why Do It

Publishing extension state allows the SUBSCRIBE and NOTIFY functionality to be handled by the other entity. Each device subscribes to the event state compositor and receives NOTIFY messages from it instead. This can scale further as less state is present in Asterisk, and also allows multiple Asterisk instances to be used while still making extension state available to everyone from the central event state compositor.



What Can Be Published?

PJSIP has a pluggable body type system. Any type that can be subscribed to for extension state can be published. As of this writing the available body types are:

- application/dialog-info+xml
- application/pidf+xml
- application/xpidf+xml
- application/cpim-pidf+xml

The PUBLISH request will contain the same body that a NOTIFY request would.

Configuration

The publishing of extension state is configured by specifying an **outbound publish** in the pjsip.conf configuration file. This tells PJSIP how to publish to another entity and gives it information about what to publish. The outbound publishing of extension state has some additional arguments, though, which allow more control.

The **@body** option specifies what body type to publish. This is a required option.

The **@context** option specifies a filter for context. This is a regular expression and is optional.

The **@exten** option specifies a filter for extensions. This is a regular expression and is optional.

An additional option which is required on the outbound publish is the **multi_user** option. This enables support in the outbound publish module for publishing to different users. This is needed for extension state publishing so the specific extension can be published to. Without this option enabled all PUBLISH requests would go to the same user.

Example Configuration

This configuration would limit outbound publish to only extension state changes as a result of a hint named "1000" in the context "users".

```
[test-esc]
type=outbound-publish
server_uri=sip:172.16.0.100
from_uri=sip:172.16.0.100
event=dialog
multi_user=yes
@body=application/dialog-info+xml
@context=^users
@exten=^1000
```

This configuration would limit outbound publish to all extension state changes a result of hints in the context "users".

```
[test-esc]
type=outbound-publish
server_uri=sip:172.16.0.100
from_uri=sip:172.16.0.100
event=dialog
multi_user=yes
@body=application/dialog-info+xml
@context=^users
```

You are also not limited to a single configured outbound publish. You can have as many as you want, provided they have different names. Each one can go to the same server with a different body type, or to different servers.

What About Making It More Dynamic?

As part of the work to implement the publishing of extension state, the concept of **autohints** were also created. Autohints are created automatically as a result of a device state change. The extension name used is the name of the device, without the technology. They can be enabled by setting "autohints=yes" in a context in extensions.conf like so:

```
[users]
autohints=yes
```

For example, once enabled, if a device state change occurs for "PJSIP/alice" and no hint named "alice" exists, then one will be automatically created in lieu of explicit definition of the following:

```
exten => alice,hint,PJSIP/alice
```

Despite being added after startup, this hint will still be given to the extension state publishing for publishing.

The Other Entity

Throughout this page, I've mentioned another entity; but what can you use? Kamailio! Kamailio has event state compositor support available using the presence module. It can be configured to accept SUBSCRIBE and PUBLISH requests, persist information in a database, and to then send NOTIFY messages to each subscribed device. The module exports the handle_publish and handle_subscribe functions for handling each.

This module works perfectly with the PJSIP extension state publishing support. The Asterisk configuration needs to use a URI to the Kamailio server and the Kamailio server has to explicitly trust traffic from the Asterisk instance, or authentication needs to be configured.

PJSIP with Proxies

There are many different proxy scenarios Asterisk can be involved in. Not all can be explained here but a few examples can help you adapt to your specific situation. The first, and simplest, scenario is where Asterisk is functioning as a PBX on the same private network that the phones are on but needs connectivity to an Internet telephony Service Provider (ITSP).

Outbound Proxy

We'll assume that the ITSP requires Asterisk to register in order to receive calls.   Of course, even with Asterisk behind a NAT firewall or router, a proxy isn't really necessary but the configuration is a good one to start with. While configuration of a proxy such as Kamailio is beyond the scope of this document, this scenario requires only the simplest of proxy configurations and would probably work with the samples provides by Kamailio.   We'll assume that the proxy is dual homed with one interface on the private network and one interface on the public network. We'll also assume that the proxy is relaying media as well as signalling. We'll use `192.168.0.1` as the proxy's private address and `192.168.0.2` as Asterisk's address.

Asterisk Configuration

There are several pjsip objects that need to be configured for this situation.

- `transport`: Actually, this is an un-configure action. 🙂 If Asterisk were not using a proxy you might have parameters in the transport like `external_signalling_address`, `external_media_address`, `local_net`, etc. These must NOT be set when Asterisk and the proxy are on the same network. Asterisk shouldn't know anything about what's on the other side of the proxy since the proxy's job is to make that invisible.
  Example:
  ```
  [ipv4-udp]
  type = transport
  protocol = udp
  bind = 0.0.0.0:5060
  ```

- `endpoint`: Configure the ITSP's endpoint as you normally would but add an `outbound_proxy` parameter with a URI that points to the proxy's internal address. This will direct (almost) all outbound requests for this endpoint to the proxy. You should also not enable any NAT related parameters like `force_rport`, `ice_support`, etc.
  Example:
  ```
  [myitsp]
  type = endpoint
  ; other stuff
  outbound_proxy = sip:192.168.0.1\;lr
  ```

- `aor`: In order for Asterisk to send `OPTIONS` requests to the ITSP via the proxy, the `outbound_proxy` parameter needs to be added here as well. All other aor parameters, including `contact` should be left just as though there were no proxy.
  Example:
  ```
  [myitsp]
  type = aor
  contact = sip:my.itsp.com:5060
  outbound_proxy = sip:192.168.0.1\;lr
  qualify_frequency = 60
  ```

- `registration`: Same as aor. The client and server URIs should remain as they were for the non-proxy situation and the `outbound_proxy` parameter should be added.
  Example:
  ```
  [myitsp]
  type = registration
  client_uri = sip:my_account@my.itsp.com
  server_uri = sip:my.itsp.com
  outbound_proxy = sip:192.168.0.1\;lr
  ```

- `identify`: Now things get just a little complicated. Most ITSPs don't authenticate back to their clients when sending them calls and they may be sending the caller's CallerID as the From header's user so the (almost) only way to identify calls from the ITSP is by IP address.   If there were no proxy in the picture, you'd probably set up an `identify` object with a `match = my.itsp.com` parameter. In the proxy case though, the match needs to be against the proxy's private address since that's the ip address where the packets will come from.
  Example:
  ```
  [myitsp]
  type = identify
  match = 192.168.0.1
  endpoint = myitsp
  ```

You'll have noticed that the proxy URIs have the "lr" parameter added. This is because most proxies these days follow RFC 3261 and are therefore "loose-routing". If you don't have it set, you'll probably get a 404 response from the proxy. The `"\"` before the semicolon is important to keep the semicolon from being treated as a comment start character in the config file.

At this point you should have a working ITSP trunk for both inbound and outbound calls.

Direct Media

If your proxy supports it, you can enable direct media between the phones and the proxy by setting `direct_media = yes` on the phone and ITSP endpoints. The proxy should take care of the rest. Attempting to do direct media directly between the phones and the ITSP is unlikely to work at all.

Multiple ITSPs

There's a slight issue with the above configuration if you have more than 1 ITSP trunk through the proxy. In the configuration above, the `identify` object is used to direct incoming requests from the proxy to a single endpoint and you can't direct the same ip address to multiple endpoints for obvious reasons. You could define 1 `endpoint` and 1 `identify` for the proxy to act as the receiver of calls from all service providers but that's not always convenient or possible with some front end GUIs. In this case, and if your ITSP supports it, you can use the `line` parameter of the `registration` object as the mechanism to match incoming requests to an endpoint and eliminate the use of `identify` altogether. Here's how it works: When you specify `line = true` and `endpoint = <endpoint>` on a registration, Asterisk appends a "line" parameter to the outgoing REGISTER's Contact URI that contains a unique string. It'll look like this: "`Contact: <sip:s@192.168.0.2.245:5060;line=eylpkkv>`". If the ITSP supports it, when it sends an `INVITE` request to Asterisk, it will include that "line" parameter in either the Request URI or the To header like so: "`INVITE sip:8005551212@192.168.0.2:5060;line=eylpkkv SIP/2.0`" . Asterisk will then use that unique string to match the request to the endpoint specified in the registration.

Example:

```
[myitsp]
type = registration
client_uri = sip:my_account@my.itsp.com
server_uri = sip:my.itsp.com
outbound_proxy = sip:192.168.0.1
line = yes
endpoint = myitsp
```

Header Matching

Some ITSPs include "X-" headers in their requests that contain account numbers or other identifying information. Asterisk 13.15 and 14.5 have a new `identify` feature which enables matching incoming requests to endpoints via those headers.

Example:

```
[myitsp]
type = identify
match_header = X-My-Account-Number: 12345678
endpoint = myitsp
```

Inbound Proxy

In a service provider scenario, Asterisk will most likely be behind a proxy separated from the public internet and the clients, be they phones or PBXes or whatever. In this case, the configuration burden shifts from Asterisk to the proxy. You'll probably want to set the proxy up to handle authentication, qualification, direction of media to media gateways, voicemail servers, etc, and that's all well beyond the scope of this document. Contributions that contain instructions for popular proxies would be most welcomed.

Real-time Text (T.140)

Real-time text in Asterisk

The SIP channel has support for real-time text conversation calls in Asterisk (T.140). This is a way to perform text based conversations in combination with other media, most often video. The text is sent character by character as a media stream.

During a call sometimes there are losses of T.140 packets and a solution to that is to use redundancy in the media stream (RTP). See "http://en.wikipedia.org/wiki/Text_over_IP"http://en.wikipedia.org/wiki/Text_over_IP and RFC 5194 for more information.

The supported real-time text codec is t.140.

Real-time text redundancy support is now available in Asterisk.

ITU-T T.140

You can find more information about T.140 at www.itu.int. RTP is used for the transport T.140, as specified in RFC 4103.

How to enable T.140

In order to enable real-time text with redundancy in Asterisk, modify sip.conf to add:

```
[general]
disallow=all
allow=ulaw
allow = alaw
allow=t140
allow=t140red
textsupport=yes
```

The codec settings may change, depending on your phones. The important settings here are to allow t140 and 140red and enable text support.

General information about real-time text support in Asterisk

With the configuration above, calls will be supported with any combination of real-time text, audio and video.

Text for both t140 and t140red is handled on channel and application level in Asterisk conveyed in Text frames, with the subtype "t140". Text is conveyed in such frames usually only containing one or a few characters from the real-time text flow. The packetization interval is 300 ms, handled on lower RTP level, and transmission redundancy level is 2, causing one original and two redundant transmissions of all text so that it is reliable even in high packet loss situations. Transmitting applications do not need to bother about the transmission interval. The t140red support handles any buffering needed during the packetization intervals.

Clients known to support text, audio/text or audio/video/text calls with Asterisk:

- Omnitor Allan eC - SIP audio/video/text softphone
- AuPix APS-50 - audio/video/text softphone.
- France Telecom eConf â€“audio/video/text softphone.
- SIPcon1 - open source SIP audio/text softphone available in Sourceforge.

Credits

- Asterisk real-time text support is developed by AuPix
- Asterisk real-time text redundancy support is developed by Omnitor

The work with Asterisk real-time text redundancy was supported with funding from the National Institute on Disability and Rehabilitation Research (NIDRR), U.S. Department of Education, under grant number H133E040013 as part of a co-operation between the Telecommunication Access Rehabilitation Engineering Research Center of the University of Wisconsin â€“ Trace Center joint with Gallaudet University, and Omnitor.

Olle E. Johansson, Edvina AB, has been a liason between the Asterisk project and this project.

Inter-Asterisk eXchange protocol, version 2 (IAX2)

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** - IAX2 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. IAX2 uses only 4 bytes of overhead.
- **Internationalization support** – IAX2 transmits language information, so that remote PBX content can be delivered in the native language of the calling party.
- **Remote dialplan polling** – IAX2 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** – IAX2 supports cleartext, MD5, and RSA authentication, providing flexible security models for outgoing calls and registration services.
- **Multimedia protocol** – IAX2 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** – IAX2 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** – IAX2'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 IAX2 rather than starting with SIP.

Introduction to IAX2

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.

IAX2 Configuration

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

Configuring chan_iax2 for IPv6

Configuration

IAX uses the 'bindaddr' and 'bindport' options to specify on which address and port the IAX2 channel driver will listen for incoming requests. They accept IPv6 as well as IPv4 addresses.

Examples

```
bindport=4569
```

The default port to listen on is 4569. Bindport must be specified **before** bindaddr or may be specified on a specific bindaddr if followed by colon and port (e.g. bindaddr=192.168.0.1:4569).

For IPv6 the address needs to be in brackets then colon and port (e.g. bindaddr=[2001:db8::1]:4569).

```
bindaddr=192.168.0.1:459
bindaddr=[2001:db8::1]:4569
```

You can specify 'bindaddr' more than once to bind to multiple addresses, but the first will be the default. IPv6 addresses are accepted.

✓ For details IAX configuration examples see the iax.conf.sample file that comes with the source.

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 `max jitterbuffer=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.

Trunk Timestamps

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 trunk timestamps, 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 stats 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:

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.

- 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.

- 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.

IAX2 Security

IAX2 Security

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Introduction

Overview

A change has been made to the IAX2 protocol to help mitigate denial of service attacks. This change is referred to as call token validation. This change affects how messages are exchanged and is not backwards compatible for an older client connecting to an updated server, so a number of options have been provided to disable call token validation as needed for compatibility purposes.

In addition to call token validation, Asterisk can now also limit the number of connections allowed per IP address to disallow one host from preventing other hosts from making successful connections. These options are referred to as call number limits.

For additional details about the configuration options referenced in this document, see the sample configuration file, `iax.conf.sample`. For information regarding the details of the call token validation protocol modification, see #Protocol Modification.

User Guide

Configuration

Quick Start

We strongly recommend that administrators leave the IAX2 security enhancements in place where possible. However, to bypass the security enhancements completely and have Asterisk work exactly as it did before, the following options can be specified in the `[general]` section of `iax.conf`:

iax.conf

```
[general]
...
calltokenoptional = 0.0.0.0/0.0.0.0
maxcallnumbers = 16382
...
```

Controlled Networks

This section discusses what needs to be done for an Asterisk server on a network where no unsolicited traffic will reach the IAX2 service.

Full Upgrade

If all IAX2 endpoints have been upgraded, then no changes to configuration need to be made.

Partial Upgrade

If only some of the IAX2 endpoints have been upgraded, then some configuration changes will need to be made for interoperability. Since this is for a controlled network, the easiest thing to do is to disable call token validation completely, as described under #Quick Start.

Public Networks

This section discusses the use of the IAX2 security functionality on public networks where it is possible to receive unsolicited IAX2 traffic.

Full Upgrade

If all IAX2 endpoints have been upgraded to support call token validation, then no changes need to be made. However, for enhanced security, the administrator may adjust call number limits to further reduce the potential impact of malicious call number consumption. The following configuration will allow known peers to consume more call numbers than unknown source IP addresses:

iax.conf

```
[general]
; By default, restrict call number usage to a low number.
maxcallnumbers = 16
...

[callnumberlimits]
; For peers with known IP addresses, call number limits can
; be set in this section. This limit is per IP address for
; addresses that fall in the specified range.
; <IP>/<mask> = <limit>
192.168.1.0/255.255.255.0 = 1024
...

[peerA]
; Since we won't know the IP address of a dynamic peer until
; they register, a max call number limit can be set in a
; dynamic peer configuration section.
type = peer
host = dynamic
maxcallnumbers = 1024
...
```

Partial Upgrade

If only some IAX2 endpoints have been upgraded, or the status of an IAX2 endpoint is unknown, then call token validation must be disabled to ensure interoperability. To reduce the potential impact of disabling call token validation, it should only be disabled for a specific peer or user as needed. By using the auto option, call token validation will be changed to required as soon as we determine that the peer supports it.

iax.conf

```
[friendA]
requirecalltoken = auto
...
```

Note that there are some cases where auto should not be used. For example, if multiple peers use the same authentication details, and they have not all upgraded to support call token validation, then the ones that do not support it will get locked out. Once an upgraded client successfully completes an

authenticated call setup using call token validation,
Asterisk will require it from then on. In that case, it would be better to set the requirecalltoken option to no.

Guest Access

Guest access via IAX2 requires special attention. Given the number of existing IAX2 endpoints that do not support call token validation, most systems that allow guest access should do so without requiring call token validation.

iax.conf

```
[guest]
; Note that the name "guest" is special here. When the code
; tries to determine if call token validation is required, it
; will look for a user by the username specified in the
; request. Guest calls can be sent without a username. In
; that case, we will look for a defined user called "guest" to
; determine if call token validation is required or not.
type = user
requirecalltoken = no
...
```

Since disabling call token validation for the guest account allows a huge hole for malicious call number consumption, an option has been provided to segregate the call numbers consumed by connections not using call token validation from those that do. That way, there are resources dedicated to the more secure connections to ensure that service is not interrupted for them.

iax.conf

```
[general]
maxcallnumbers_nonvalidated = 2048
...
```

CLI Commands

`iax2 show callnumber usage`

Usage: `iax2 show callnumber usage [IP address]`

Show current IP addresses which are consuming IAX2 call numbers.

`iax2 show peer`

This command will now also show the configured call number limit and whether or not call token validation is required for this peer.

Protocol Modification

This section discusses the modification that has been made to the IAX2 protocol. This information would be most useful to implementors of IAX2.

Overview

The IAX2 protocol uses a call number to associate messages with which call they belong to. The available amount of call numbers is finite as defined by the protocol. Because of this, it is important to prevent attackers from maliciously consuming call numbers. To achieve this, an enhancement to the IAX2 protocol has been made which is referred to as call token validation.

Call token validation ensures that an IAX2 connection is not coming from a spoofed IP address. In addition to using call token validation, Asterisk will also limit how many call numbers may be consumed by a given remote IP address. These limits have defaults that will usually not need to be changed, but can be modified for a specific need.

The combination of call token validation and call number limits is used to mitigate a denial of service attack to consume all available IAX2 call numbers. An alternative approach to securing IAX2 would be to use a security layer on top of IAX2, such as DTLS RFC 4347 or IPsec RFC 4301.

Call Token Validation

The key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "MAY", and "OPTIONAL" in this document are to be interpreted as described in RFC 2119.

For this section, when the word "request" is used, it is referring to the command that starts an IAX2 dialog.

This modification adds a new IAX2 frame type, and a new information element be defined.

Frame Type: CALLTOKEN — 0x28 (40)

IE: CALLTOKEN — 0x36 (54)

When a request is initially sent, it SHOULD include the CALLTOKEN IE with a zero-length payload to indicate that this client supports the CALLTOKEN exchange. When a server receives this request, it MUST respond with the IAX2 message CALLTOKEN. The CALLTOKEN message MUST be sent with a source call number of 0, as a call number will not yet be allocated for this call.

For the sake of backwards compatibility with clients that do not support token validation, server implementations MAY process requests that do not indicate CALLTOKEN support in their initial request. However, this SHOULD NOT be the default behavior, as it gives up the security benefits gained by CALLTOKEN validation.

After a client sends a request with an empty CALLTOKEN IE, it MUST be prepared to receive a CALLTOKEN response, or to receive a response that would be given in the case of a valid CALLTOKEN. This is how a client must behave to inter operate with IAX2 server implementations that do not yet support CALLTOKEN validation.

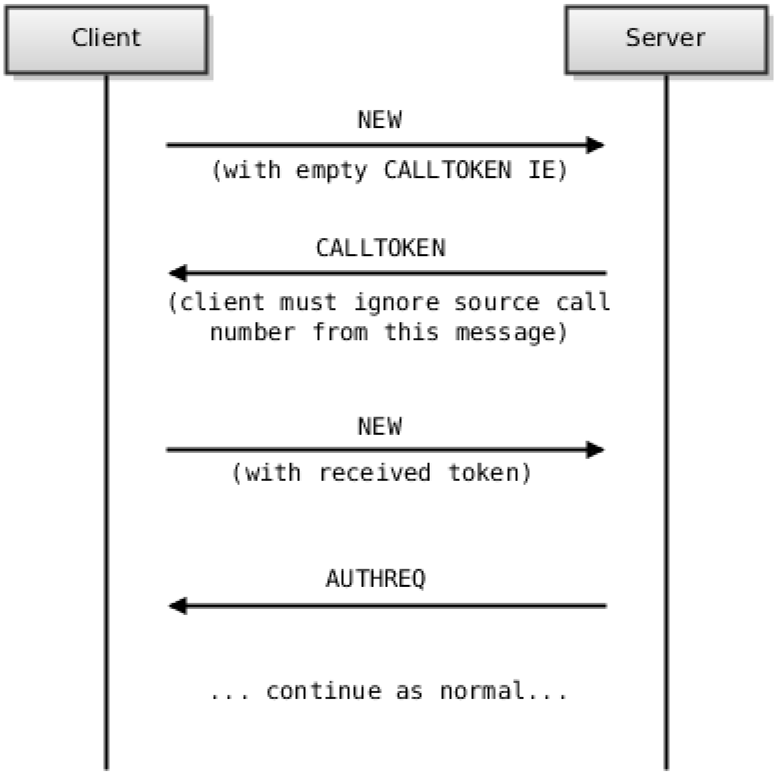
When an IAX2 client receives a CALLTOKEN response, it MUST send its initial request again. This request MUST include the CALLTOKEN IE with a copy of the value of the CALLTOKEN IE received in the CALLTOKEN response. The IE value is an opaque value. Clients MUST be able to accept a CALLTOKEN payload of any length, up to the maximum length allowed in an IAX2 IE.

The value of the payload in the CALLTOKEN IE is an implementation detail. It is left to the implementor to decide how sophisticated it should be. However, it MUST be enough such that when the CALLTOKEN IE is sent back, it can be used to verify that the source IP address and port number has not been spoofed.

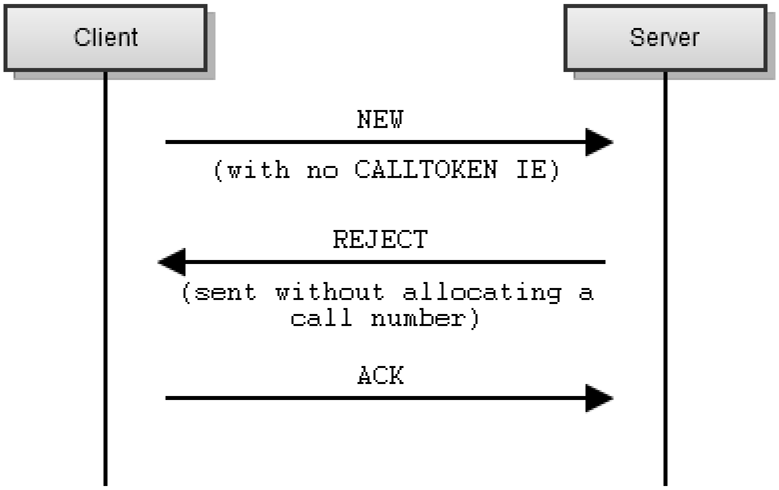
If a server receives a request with an invalid CALLTOKEN IE value, then it MUST drop it and not respond.

Example Message Exchanges

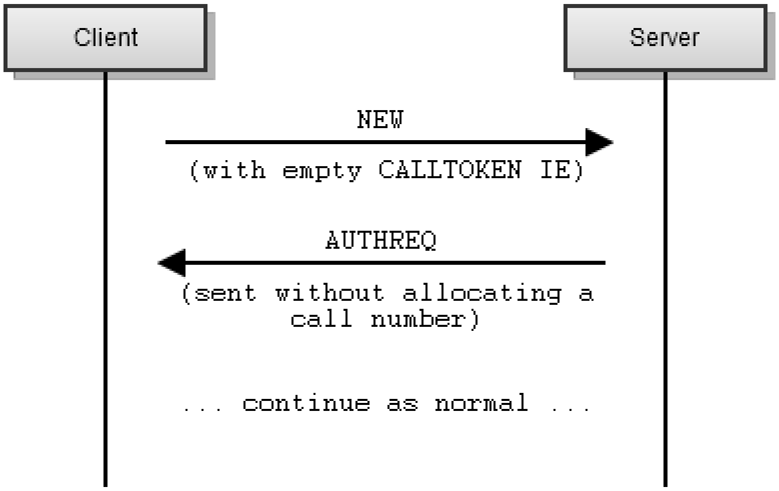
Call Setup


Client
Server
NEW
(with empty CALLTOKEN IE)
CALLTOKEN
(client must ignore source call number from this message)
NEW
(with received token)
AUTHREQ
... continue as normal...

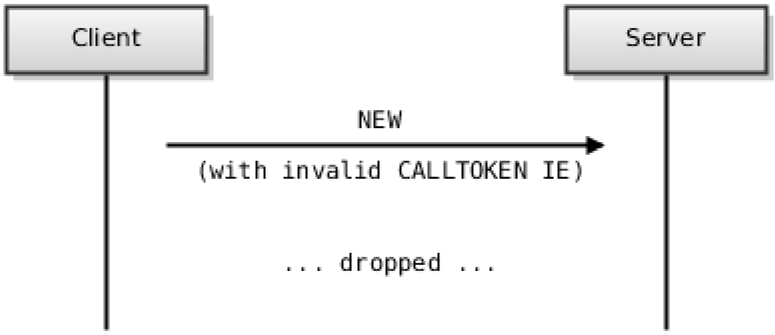
Call Setup, client does not support CALLTOKEN



Call Setup, client supports CALLTOKEN, server does not


Client
Server
NEW
(with empty CALLTOKEN IE)
AUTHREQ
(sent without allocating a call number)
... continue as normal ...

Call Setup from client that sends invalid token



Asterisk Implementation

This section includes some additional details on the implementation of these changes in Asterisk.

CALLTOKEN IE Payload

For Asterisk, we will encode the payload of the CALLTOKEN IE such that the server is able to validate a received token without having to store any information after transmitting the CALLTOKEN response. The CALLTOKEN IE payload will contain:

- A timestamp (epoch based)
- SHA1 hash of the remote IP address and port, the timestamp, as well some random data generated when Asterisk starts.

When a CALLTOKEN IE is received, its validity will be determined by recalculating the SHA1 hash. If it is a valid token, the timestamp is checked to determine if the token is expired. The token timeout will be hard coded at 10 seconds for now. However, it may be made configurable at some point if it seems to be a useful addition. If the server determines that a received token is expired, it will treat it as an invalid token and not respond to the request.

By using this method, we require no additional memory to be allocated for a dialog, other than what is on the stack for processing the initial request, until token validation is complete.

However, one thing to note with this CALLTOKEN IE encoding is that a token would be considered valid by Asterisk every time a client sent it until we considered it an expired token. However, with use of the "maxcallnumbers" option, this is not actually a problem. It just means that an attacker could hit their call number limit a bit quicker since they would only have to acquire a single token per timeout period, instead of a token per request.

DAHDI

Under Construction

Top-level page for DAHDI **channel driver** information

What are the software and hardware requirements for chan_dahdi?

How do I configure it?

What are some examples of Analog and PRI configurations?

Links to use:

DAHDI

Local Channel

Overview

Most Channel Drivers in Asterisk provide capability to connect Asterisk to external devices via specific protocols (e.g. chan_pjsip), whereas Local Channels provide a channel type for calling back into Asterisk itself.

That is, when dialing a Local Channel you are dialing within Asterisk into the Asterisk dialplan.

Usage of Local Channels between other channel technologies can add additional programmatic flexibility, but of course at some level of performance cost. Local Channels are often used to execute dialplan logic from Applications that would expect to connect directly with a channel.

Two of the most common areas where Local channels are used include members configured for queues, and in use with callfiles. Another interesting case could be that you want to ring multiple destinations, but with different information for each call, such as different callerID for each outgoing request.

In this section you'll find Local Channel Examples that illustrate usage plus details on Local Channel Optimization and a list of Local Channel Modifiers.

The Local Channel in Asterisk Architecture

Previous to Asterisk 12, Local Channel functionality was provided by the **chan_local** module. In Asterisk 12, chan_local was moved into the Asterisk system core and is no longer a loadable module.

Local Channel Examples

Local channels are best demonstrated through the use of an example. In the sub-pages here you'll find several examples of Local Channel usage.

Delay Dialing Devices Example

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.

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.

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.

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.

Local Channel Optimization

Default Channel Optimization

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.

Diagrams really help to show what is going on:

Figure 1

This is a call in an unanswered (ringing) state - from SIP to SIP using Local Channels in between.

Figure 2

By default, after the callee answers this is what the call would look like with the Local Channels optimizing out.

Figure 3

This is what the call would look like when established if you called the Local Channel with "/n". You can see the Local Channels get pushed into bridges with channels they were connected with through app_dial previously.

Call SIP to SIP via Local Channels (unanswered state)

Fig 1

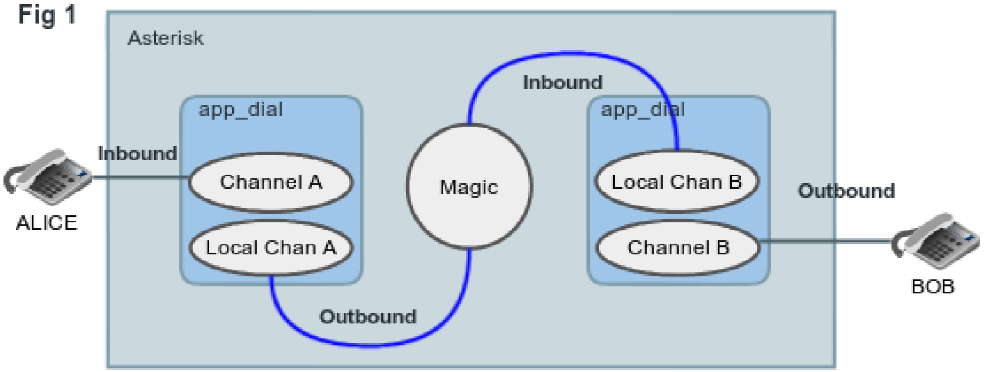


Fig 2

Call established with Local Channels optimized out

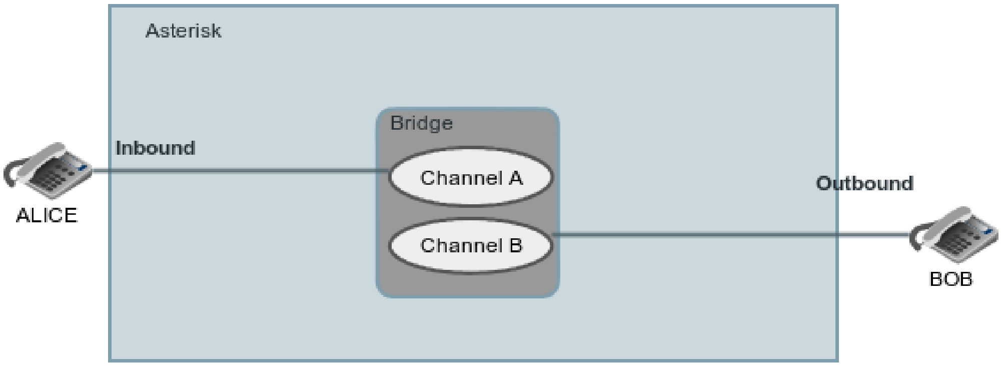
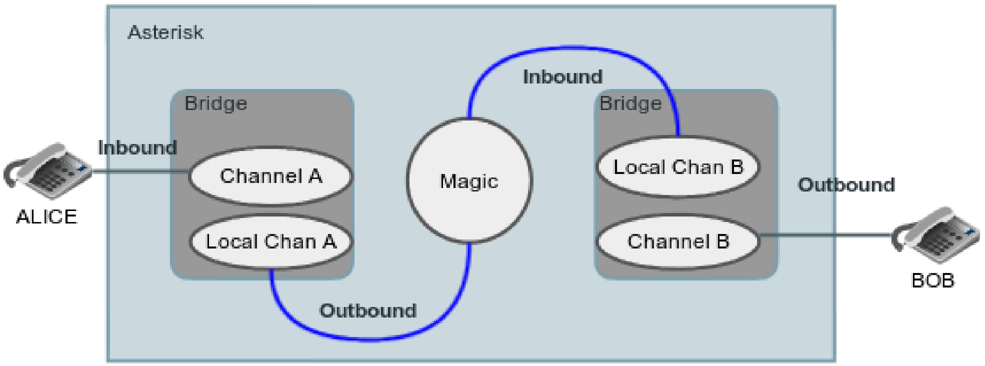


Fig 3

Call established with non-optimized Local Channels



Disabling Local Channel Optimization

You may have read about the /n modifier in Local Channel Modifiers. 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 dial-string, we can keep the Local channel in the call path, along with any channel variables, or other channel specific information.

When to disable optimization

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)
```

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 of the dial-string, our Local channel will now stay in the call path and not go away.

Detailed walk-through of example call-flow

Why does adding the **/n** option all of a sudden 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 the lack of the "/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));
```

Local Channel Modifiers

Usage

Dial-string modifiers exist that allow changing the default behavior of a Local Channel.

The modifiers are added to a channel by adding a slash followed by a flag onto the end of the Local Channel dial-string.

For example below we are adding the "n" modifier to the dial-string.

```
Local/101@mycontext/n
```

You can add more than one modifier by adding them directly adjacent to the previous modifier.

```
Local/101@mycontext/nj
```

List of Modifiers

- 'n' - Instructs the Local channel to not do a native transfer (the "n" stands for *No 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. Read about Local Channel Optimization to better understand when this option is necessary.
- 'j' - 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' - 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' - This 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 and was removed in Asterisk 12.

Motif

Under Construction

Page for information on the Motif channel driver, describing configuration, pointing to any resources and a top-level page for any examples or tutorials such as calling with Google Voice.

Calling using Google

> This new page replaces the old page. The old page documents behavior that is not functional or supported going forward. This new page documents behavior as of Asterisk 11. For more information, please see the blog posting http://blogs.digium.com/2012/07/24/asterisk-11-development-the-motive-for-motif/

Prerequisites

Asterisk communicates with Google Voice and Google Talk using the chan_motif Channel Driver and the res_xmpp Resource module. Before proceeding, please ensure that both are compiled and part of your installation. Compilation of res_xmpp and chan_motif for use with Google Talk / Voice are dependant on the iksemel library files as well as the OpenSSL development libraries presence on your system.

Calling using Google Voice or via the Google Talk web client requires the use of Asterisk 11.0 or greater. Older versions of Asterisk will not work.

For basic calling between Google Talk web clients, you need a Google Mail account.

For calling to and from the PSTN, you will need a Google Voice account.

In your Google account, you'll want to change the Chat setting from the default of "Automatically allow people that I communicate with often to chat with me and see when I'm online" to the second option of "Only allow people that I've explicitly approved to chat with me and see when I'm online."

IPv6 is currently not supported. Use of IPv4 is required.

Google Voice can now be used with Google Apps accounts.

RTP configuration

ICE support is required for chan_motif to operate. It is disabled by default and must be explicitly enabled in the RTP configuration file rtp.conf as follows.

```
[general]
icesupport=yes
```

If this option is not enabled you will receive the following error message.

```
Unable to add Google ICE candidates as ICE support not available or no candidates available
```

Motif configuration

The Motif channel driver is configured with the motif.conf configuration file, typically located in /etc/asterisk. What follows is an example configuration for successful operation.

Example Motif Configuration

```
[google]
context=incoming-motif
disallow=all
allow=ulaw
connection=google
```

This general section of this configuration specifies several items.

1. That calls will terminate to or originate from the **incoming-motif** context; context=incoming-motif
2. That all codecs are first explicitly disallowed
3. That G.711 ulaw is allowed
4. The an XMPP connection called "google" is to be used

Google lists supported audio codecs on this page - https://developers.google.com/talk/open_communications

Per section, 5, the supported codecs are:

1. PCMA
2. PCMU
3. G.722
4. GSM
5. iLBC
6. Speex

Our experience shows this not to be the case. Rather, the codecs, supported by Asterisk, and seen in an invite from Google Chat are:

1. PCMA
2. PCMU
3. G.722
4. iLBC
5. Speex 16kHz
6. Speex 8kHz

It should be noted that calling using Google Voice requires the G.711 ulaw codec. So, if you want to make sure Google Voice calls work, allow G.711 ulaw, at a minimum.

XMPP Configuration

The res_xmpp Resource is configured with the xmpp.conf configuration file, typically located in /etc/asterisk. What follows is an example configuration for successful operation.

Example XMPP Configuration

```
[general]
[google]
type=client
serverhost=talk.google.com
username=example@gmail.com
secret=examplepassword
priority=25
port=5222
usetls=yes
usesasl=yes
status=available
statusmessage="I am available"
timeout=5
```

The default general section does not need any modification.

The google section of this configuration specifies several items.

1. The type is set to client, as we're connecting to Google as a service; type=client
2. The serverhost is Google's talk server; serverhost=talk.google.com
3. Our username is configured as your_google_username@gmail.com; username=your_google_username@gmail.com
4. Our password is configured using the secret option; secret=your_google_password
5. Google's talk service operates on port 5222; port=5222
6. Our priority is set to 25; priority=25
7. TLS encryption is required by Google; usetls=yes
8. Simple Authentication and Security Layer (SASL) is used by Google; usesasl=yes
9. We set a status message so other Google chat users can see that we're an Asterisk server; statusmessage="I am available"
10. We set a timeout for receiving message from Google that allows for plenty of time in the event of network delay; timeout=5

More about Priorities

As many different connections to Google are possible simultaneously via different client mechanisms, it is important to understand the role of priorities in the routing of inbound calls. Proper usage of the priority setting can allow use of a Google account that is not otherwise entirely dedicated to voice services.

With priorities, the higher the setting value, the more any client using that value is preferred as a destination for inbound calls, in deference to any other client with a lower priority value. Known values of commonly used clients include the Gmail chat client, which maintains a priority of **20**, and the Windows GTalk client, which uses a priority of **24**. The maximum allowable value is **127**. Thus, setting one's **priority** option for the XMPP peer in res_xmpp.conf to a value higher than 24 will cause inbound calls to flow to Asterisk, even while one is logged into either Gmail or the Windows GTalk client.

Outbound calls are unaffected by the priority setting.

Phone configuration

Now, let's create a phone. The configuration of a SIP device for this purpose would, in sip.conf, typically located in /etc/asterisk, look something like:

```
[malcolm]
type=peer
secret=my_secure_password
host=dynamic
context=local
```

Dialplan configuration

Incoming calls

Next, let's configure our dialplan to receive an incoming call from Google and route it to the SIP phone we created. To do this, our dialplan, extensions.conf, typically located in /etc/asterisk, would look like:

```
[incoming-motif]
exten => s,1,NoOp()
 same => n,Wait(1)
 same => n,Answer()
 same => n,SendDTMF(1)
 same => n,Dial(SIP/malcolm,20)
```

⚠ Did you know that the Google Chat client does this same thing; it waits, and then sends a DTMF 1. Really.

This example uses the "s" unmatched extension, because we're only configuring one client connection in this example.

In this example, we're Waiting 1 second, answering the call, sending the DTMF "1" back to Google, and **then** dialing the call.
We do this, because inbound calls from Google enable, even if it's disabled in your Google Voice control panel, call screening.
Without this SendDTMF event, you'll have to confirm with Google whether or not you want to answer the call.

✓ **Using Google's voicemail**
Another method for accomplishing the sending of the DTMF event is to use Dial option "D." The D option tells Asterisk to send a specified DTMF string after the called party has answered. DTMF events specified before a colon are sent to the **called** party. DTMF events specified after a colon are sent to the **calling** party.

In this example then, one does not need to actually answer the call first, though one should still wait at least a second for things, like STUN setup, to finish. This means that if the called party doesn't answer, Google will resort to sending the call to one's Google Voice voicemail box, instead of leaving it at Asterisk.

```
exten => s,1,Dial(SIP/malcolm,20,D(:1))
```

✓ **Filtering Caller ID**
The inbound CallerID from Google is going to look a bit nasty, e.g.:

```
+15555551212@voice.google.com/srvres-MTAuMjE4LjIuMTk3Ojk4MzM=
```

Your VoIP client (SIPDroid) might not like this, so let's simplify that Caller ID a bit, and make it more presentable for your phone's display. Here's the example that we'll step through:

```
exten => s,1,NoOp()
 same => n,Set(crazygooglecid=${CALLERID(name)})
 same => n,Set(stripcrazysuffix=${CUT(crazygooglecid,@,1)})
 same => n,Set(CALLERID(all)=${stripcrazysuffix})
 same => n,Dial(SIP/malcolm,20,D(:1))
```

First, we set a variable called **crazygooglecid** to be equal to the name field of the CALLERID function. Next, we use the CUT function to grab everything that's before the @ symbol, and save it in a new variable called **stripcrazysuffix.** We'll set this new variable to the CALLERID that we're going to use for our Dial. Finally, we'll actually Dial our internal destination.

Outgoing calls

Outgoing calls to Google Talk users take the form of:

```
exten => 100,1,Dial(Motif/google/mybuddy@gmail.com,,r)
```

Where the technology is "Motif," the dialing peer is "google" as defined in xmpp.conf, and the dial string is the Google account name.

We use the Dial option "r" because Google doesn't provide ringing indications.

Outgoing calls made to Google Voice take the form of:

```
exten => _1XXXXXXXXXX,1,Dial(Motif/google/${EXTEN}@voice.google.com,,r)
```

Where the technology is "Motif," the dialing peer is "google" as defined in motif.conf, and the dial string is a full E.164 number, sans the plus character.

Again, we use Dial option "r" because Google doesn't provide ringing indications.

mISDN

Introduction to mISDN

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

mISDN 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

mISDN 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

mISDN 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

mISDN 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] subsection

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})
```

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.

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.*

mISDN Debugging and 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 https://issues.asterisk.org in the "channel drivers" project, "chan_misdn" category. Read the bug guidelines to make sure you provide all the information needed.

mISDN 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.

mISDN 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.

Mobile Channel

chan_mobile pages

Introduction to the Mobile Channel

Asterisk Channel Driver to allow Bluetooth Cell/Mobile Phones to be used as FXO devices, and Headsets as FXS devices.

Mobile Channel Features

- Multiple Bluetooth Adapters supported.
- Multiple phones can be connected.
- Multiple headsets can be connected.
- Asterisk automatically connects to each configured mobile phone / headset when it comes in range.
- CLI command to discover bluetooth devices.
- Inbound calls on the mobile network to the mobile phones are handled by Asterisk, just like inbound calls on a Zap channel.
- CLI passed through on inbound calls.
- Dial outbound on a mobile phone using Dial(Mobile/device/nnnnnnn) in the dialplan.
- Dial a headset using Dial(Mobile/device) in the dialplan.
- Application MobileStatus can be used in the dialplan to see if a mobile phone / headset is connected.
- Supports devicestate for dialplan hinting.
- Supports Inbound and Outbound SMS.
- Supports 'channel' groups for implementing 'GSM Gateways'

Mobile Channel Requirements

In order to use chan_mobile, you must have a working bluetooth subsystem on your Asterisk box. This means one or more working bluetooth adapters, and the BlueZ packages.

Any bluetooth adapter supported by the Linux kernel will do, including usb bluetooth dongles.

The BlueZ package you need is bluez-utils. If you are using a GUI then you might want to install bluez-pin also. You also need libbluetooth, and libbluetooth-dev if you are compiling Asterisk from source.

You need to get bluetooth working with your phone before attempting to use chan_mobile. This means 'pairing' your phone or headset with your Asterisk box. I dont describe how to do this here as the process differs from distro to distro. You only need to pair once per adapter.

See http://www.bluez.org for details about setting up Bluetooth under Linux.

Mobile Channel Concepts

chan_mobile deals with both bluetooth adapters and bluetooth devices. This means you need to tell chan_mobile about the bluetooth adapters installed in your server as well as the devices (phones / headsets) you wish to use.

chan_mobile currently only allows one device (phone or headset) to be connected to an adapter at a time. This means you need one adapter for each device you wish to use simultaneously. Much effort has gone into trying to make multiple devices per adapter work, but in short it doesnt.

Periodically chan_mobile looks at each configured adapter, and if it is not in use (i.e. no device connected) will initiate a search for devices configured to use this adapater that may be in range. If it finds one it will connect the device and it will be available for Asterisk to use. When the device goes out of range, chan_mobile will disconnect the device and the adapter will become available for other devices.

Configuring chan_mobile

The configuration file for chan_mobile is /etc/asterisk/mobile.conf. It is a normal Asterisk config file consisting of sections and key=value pairs.

See configs/mobile.conf.sample for an example and an explanation of the configuration.

Using chan_mobile

chan_mobile.so must be loaded either by loading it using the Asterisk CLI, or by adding it to /etc/asterisk/modules.conf
Search for your bluetooth devices using the CLI command 'mobile search'. Be patient with this command as it will take 8 - 10 seconds to do the discovery. This requires a free adapter.
Headsets will generally have to be put into 'pairing' mode before they will show up here.
This will return something like the following :-

```
*CLI> mobile search
Address Name Usable Type Port
00:12:56:90:6E:00 LG TU500 Yes Phone 4
00:80:C8:35:52:78 Toaster No Headset 0
00:0B:9E:11:74:A5 Hello II Plus Yes Headset 1
00:0F:86:0E:AE:42 Daves Blackberry Yes Phone 7
```

This is a list of all bluetooth devices seen and whether or not they are usable with chan_mobile. The Address field contains the 'bd address' of the device. This is like an ethernet mac address. The Name field is whatever is configured into the device as its name. The Usable field tells you whether or not the device supports the Bluetooth Handsfree Profile or Headset profile. The Type field tells you whether the device is usable as a Phone line (FXO) or a headset (FXS) The Port field is the number to put in the configuration file.

Choose which device(s) you want to use and edit /etc/asterisk/mobile.conf. There is a sample included with the Asterisk-addons source under configs/mobile.conf.sample.

Be sure to configure the right bd address and port number from the search. If you want inbound calls on a device to go to a specific context, add a context= line, otherwise the default will be used. The 'id' of the device [bitinbrackets] can be anything you like, just make it unique.

If you are configuring a Headset be sure to include the type=headset line, if left out it defaults to phone.

The CLI command 'mobile show devices' can be used at any time to show the status of configured devices, and whether or not the device is capable of sending / receiving SMS via bluetooth.

```
*CLI> mobile show devices
ID Address Group Adapter Connected State SMS
headset 00:0B:9E:11:AE:C6 0 blue No Init No
LGTU550 00:E0:91:7F:46:44 1 dlink No Init No
```

As each phone is connected you will see a message on the Asterisk console :-

```
Loaded chan_mobile.so => (Bluetooth Mobile Device Channel Driver)
– Bluetooth Device blackberry has connected.
– Bluetooth Device dave has connected.
```

To make outbound calls, add something to you Dialplan like the following :- (modify to suit)

```
; Calls via LGTU5500
exten => _9X.,1,Dial(Mobile/LGTU550/${EXTEN:1},45)
exten => _9X.,n,Hangup
```

To use channel groups, add an entry to each phones definition in mobile.conf like group=n where n is a number.

Then if you do something like Dial(Mobile/g1/123456) Asterisk will dial 123456 on the first connected free phone in group 1.

Phones which do not have a specific 'group=n' will be in group 0.

To dial out on a headset, you need to use some other mechanism, because the headset is not likely to have all the needed buttons on it. res_clioriginate is good for this :-

```
*CLI> originate Mobile/headset extension NNNNN@context
```

This will call your headset, once you answer, Asterisk will call NNNNN at context context

Mobile Channel Dialplan Hints

chan_mobile supports 'device status' so you can do somthing like

```
exten => 1234,hint,SIP/30&Mobile/dave&Mobile/blackberry
```

MobileStatus Application

chan_mobile also registers an application named MobileStatus. You can use this in your Dialplan to determine the 'state' of a device.
For example, suppose you wanted to call dave's extension, but only if he was in the office. You could test to see if his mobile phone was attached to Asterisk, if it is dial his extension, otherwise dial his mobile phone.

```
exten => 40,1,MobileStatus(dave,DAVECELL)
exten => 40,2,GotoIf($["${DAVECELL}" = "1"]?3:5)
exten => 40,3,Dial(ZAP/g1/0427466412,45,tT)
exten => 40,4,Hangup
exten => 40,5,Dial(SIP/40,45,tT)
exten => 40,6,Hangup
```

MobileStatus sets the value of the given variable to :-

- 1 = Disconnected. i.e. Device not in range of Asterisk, or turned off etc etc
- 2 = Connected and Not on a call. i.e. Free
- 3 = Connected and on a call. i.e. Busy

Mobile Channel DTMF Debouncing

DTMF detection varies from phone to phone. There is a configuration variable that allows you to tune this to your needs. e.g. in mobile.conf

```
[LGTU550]
address=00:12:56:90:6E:00
port=4
context=incoming-mobile
dtmfskip=50
```

change dtmfskip to suit your phone. The default is 200. The larger the number, the more chance of missed DTMF. The smaller the number the more chance of multiple digits being detected.

Mobile Channel SMS Sending and Receiving

If Asterisk has detected your mobile phone is capable of SMS via bluetooth, you will be able to send and receive SMS.

Incoming SMS's cause Asterisk to create an inbound call to the context you defined in mobile.conf or the default context if you did not define one. The call will start at extension 'sms'. Two channel variables will be available, SMSSRC = the number of the originator of the SMS and SMSTXT which is the text of the SMS. This is not a voice call, so grab the values of the variables and hang the call up.

So, to handle incoming SMS's, do something like the following in your dialplan

```
[incoming-mobile]
exten => sms,1,Verbose(Incoming SMS from ${SMSSRC} ${SMSTXT})
exten => sms,n,Hangup()
```

The above will just print the message on the console.

If you use res_jabber, you could do something like this :-

```
[incoming-mobile]
exten => sms,1,JabberSend(transport,user@jabber.somewhere.com,SMS from ${SMSRC}
${SMSTXT})
exten => sms,2,Hangup()
```

To send an SMS, use the application MobileSendSMS like the following :-

```
exten => 99,1,MobileSendSMS(dave,0427123456,Hello World)
```

This will send 'Hello World' via device 'dave' to '0427123456'

Mobile Channel Debugging

Different phone manufacturers have different interpretations of the Bluetooth Handsfree Profile Spec. This means that not all phones work the same way, particularly in the connection setup / initialisation sequence. I've tried to make chan_mobile as general as possible, but it may need modification to support some phone i've never tested.

Some phones, most notably Sony Ericsson 'T' series, dont quite conform to the Bluetooth HFP spec. chan_mobile will detect these and adapt accordingly. The T-610 and T-630 have been tested and work fine.

If your phone doesnt behave has expected, turn on Asterisk debugging with 'core set debug 1'.

This will log a bunch of debug messages indicating what the phone is doing, importantly the rfcomm conversation between Asterisk and the phone. This can be used to sort out what your phone is doing and make chan_mobile support it.

Be aware also, that just about all mobile phones behave differently. For example my LG TU500 wont dial unless the phone is a the 'idle' screen. i.e. if the phone is showing a 'menu' on the display, when you dial via Asterisk, the call will not work. chan_mobile handles this, but there may be other phones that do other things too...

Important: Watch what your mobile phone is doing the first few times. Asterisk wont make random calls but if chan_mobile fails to hangup for some reason and you get a huge bill from your telco, dont blame me

Unistim

Introduction to the Unistim channel

Unified Networks IP Stimulus (UNIStim) Channel Driver for Asterisk

This is a channel driver for Unistim protocol. You can use a least a Nortel i2002, i2004 and i2050.

Following features are supported :

- Send/Receive CallerID
- Redial
- SoftKeys
- SendText()
- Music On Hold
- Message Waiting Indication (MWI)
- Distinctive ring
- Transfer
- History
- Forward
- Dynamic SoftKeys.

How to configure the i2004

1. Power on the phone
2. Wait for message "Nortel Networks"
3. Press quickly the four buttons just below the LCD screen, in sequence from left to right
4. If you see "Locating server", power off or reboot the phone and try again
5. DHCP : 0
6. SET IP : a free ip of your network
7. NETMSK / DEF GW : netmask and default gateway
8. S1 IP : ip of the asterisk server
9. S1 PORT : 5000
10. S1 ACTION : 1
11. S1 RETRY COUNT : 10
12. S2 : same as S1

How to place a call

The line=> entry in unistim.conf does not add an extension in asterisk by default. If you want to do that, add extension=line in your phone context.

If you have this entry on unistim.conf :

```

[violet]
device=006038abcdef
line => 102
```

then use:

```
exten => 2100,1,Dial(USTM/102@violet)
```

You can display a text with :

```
exten => 555,1,SendText(Sends text to client. Greetings)
```

Rebooting a Nortel phone

- Press mute,up,down,up,down,up,mute,9,release(red button)

Distinctive ring

1. You need to append /r to the dial string.
2. The first digit must be from 0 to 7 (inclusive). It's the 'melody' selection.
3. The second digit (optional) must be from 0 to 3 (inclusive). It's the ring volume. 0 still produce a sound.

Select the ring style #1 and the default volume :

```
exten => 2100,1,Dial(USTM/102@violet/r1)
```

Select the ring style #4 with a very loud volume :

```
exten => 2100,1,Dial(USTM/102@violet/r43)
```

Country code

- You can use the following codes for country= (used for dial tone) - us fr au nl uk fi es jp no at nz tw cl se be sg il br hu lt pl za pt ee mx in de ch dk cn

- If you want a correct ring, busy and congestion tone, you also need a valid entry in indications.conf and check if res_indications.so is loaded.

- language= is also supported but it's only used by Asterisk (for more information see http://www.voip-info.org/wiki/view/Asterisk+multi-language). The end user interface of the phone will stay in english.

Bookmarks, Softkeys

Layout

```
  |--------------------|
   |  5           2    |
   |  4           1    |
   |  3           0    |
```

- When the second letter of bookmark= is @, then the first character is used for positioning this entry
- If this option is omitted, the bookmark will be added to the next available sofkey
- Also work for linelabel (example : linelabel="5@Line 123")
- You can change a softkey programmatically with SendText(@position@icon@label@extension) ex: SendText(@1@55@Stop Forwd@908)

Autoprovisioning

- This feature must only be used on a trusted network. It's very insecure : all unistim phones will be able to use your asterisk pbx.
- You must add an entry called template. Each new phones will be based on this profile.
- You must set a least line=>. This value will be incremented when a new phone is registered. device= must not be specified. By default, the phone will asks for a number. It will be added into the dialplan. Add extension=line for using the generated line number instead.

Example :

```
[general]
port=5000
autoprovisioning=yes

[template]
line => 100
bookmark=Support@123  ; Every phone will have a softkey Support
```

- If a first phone have a mac = 006038abcdef, a new device named USTM/100@006038abcdef will be created.
- If a second phone have a mac = 006038000000, it will be named USTM/101@006038000000 and so on.

- When autoprovisioning=tn, new phones will ask for a tn, if this number match a tn= entry in a device, this phone will be mapped into.

Example:

```
[black]
tn=1234
line => 100
```

- If a user enter TN 1234, the phone will be known as USTM/100@black.

History

- Use the two keys located in the middle of the Fixed feature keys row (on the bottom of the phone) to enter call history.

- By default, chan_unistim add any incoming and outgoing calls in files (/var/log/asterisk/unistimHistory). It can be a privacy issue, you can disable this feature by adding callhistory=0. If history files were created, you also need to delete them. callhistory=0 will NOT disable normal asterisk CDR logs.

Forward

- This feature requires chan_local (loaded by default)

Generic asterisk features

You can use the following entries in unistim.conf

- Billing - accountcode amaflags
- Call Group - callgroup pickupgroup (untested)
- Music On Hold - musiconhold
- Language - language (see section Coutry Code)
- RTP NAT - nat (control ast_rtp_setnat, default = 0. Obscure behaviour)

Trunking

- It's not possible to connect a Nortel Succession/Meridian/BCM to Asterisk via chan_unistim. Use either E1/T1 trunks, or buy UTPS (UNISTIM Terminal Proxy Server) from Nortel.

Wiki, Additional infos, Comments :

- http://www.voip-info.org/wiki-Asterisk+UNISTIM+channels

**BSD :*

- Comment #define HAVE_IP_PKTINFO in chan_unistim.c
- Set public_ip with an IP of your computer
- Check if unistim.conf is in the correct directory

Issues

- As always, NAT can be tricky. If a phone is behind a NAT, you should port forward UDP 5000 (or change general port= in unistim.conf) and UDP 10000 (or change yourphone rtp_port=)

- Only one phone per public IP (multiple phones behind the same NAT don't work). You can either :
  - Setup a VPN
  - Install asterisk inside your NAT. You can use IAX2 trunking if you're master asterisk is outside.
  - If asterisk is behind a NAT, you must set general public_ip= with your public IP. If you don't do that or the bindaddr is invalid (or no longer valid, eg dynamic IP), phones should be able to display messages but will be unable to send/receive RTP packets (no sound)
- Don't forget : this work is based entirely on a reverse engineering, so you may encounter compatibility issues. At this time, I know three ways to establish a RTP session. You can modify yourphone rtp_method= with 0, 1, 2 or 3. 0 is the default method, should work. 1 can be used on new firmware (black i2004) and 2 on old violet i2004. 3 can be used on black i2004 with chrome.
- If you have difficulties, try unistim debug and set verbose 3 on the asterisk CLI. For extra debug, uncomment #define DUMP_PACKET 1 and recompile chan_unistim.

Protocol information

Protocol versions

31 October 2008
UNIStim Firmware Release 3.1 for IP Phones, includes:

- 0604DCG for Phase II IP Phones (2001, 2002 2004),
- 0621C6H for IP Phone 2007,
- 0623C6J, 0624C6J, 0625C6J and 0627C6J for IP Phone 1110, 1120E,1140E and 1150E respectively
- 062AC6J for IP Phone 1210, 1220, and 1230

27 February 2009
UNIStim Firmware Release 3.2 for IP Phones, including:

- 0604DCJ for Phase II IP Phones (2001, 2002 & 2004),
- 0621C6M for IP Phone 2007,
- 0623C6N, 0624C6N, 0625C6N and 0627C6N for IP Phone 1110, 1120E,1140E and 1150E respectively
- 062AC6N for IP Phone 1210, 1220, and 1230

30 June 2009
UNIStim Firmware Release 3.3 for IP Phones:

- 0604DCL for Phase II IP Phones (2001, 2002 & 2004),
- 0621C6P for IP Phone 2007,
- 0623C6R, 0624C6R, 0625C6R and 0627C6R for IP Phone 1110, 1120E,1140E and 1150E respectively
- 062AC6R for IP Phone 1210, 1220, and 1230

27 November 2009
UNIStim Software Release 4.0 for IP Phones, includes:

- 0621C7A for IP Phone 2007,
- 0623C7F, 0624C7F, 0625C7F and 0627C7F for IP Phone 1110, 1120E,1140E and 1150E respectively
- 062AC7F for IP Phone 1210, 1220, and 1230

28 February 2010
UNIStim Software Release 4.1 IP Deskphone Software

- 0621C7D / 2007 IP Deskphone
- 0623C7J / 1110 IP Deskphone
- 0624C7J / 1120E IP Deskphone
- 0625C7J / 1140E IP Deskphone
- 0627C7J / 1150E IP Deskphone
- 0626C7J / 1165E IP Deskphone
- 062AC7J / 1210 IP Deskphone
- 062AC7J / 1220 IP Deskphone
- 062AC7J / 1230 IP Deskphone

29 2010
UNIStim Software Release 4.2 IP Deskphone Software

- 0621C7G / 2007 IP Deskphone
- 0623C7M / 1110 IP Deskphone
- 0624C7M / 1120E IP Deskphone
- 0625C7M / 1140E IP Deskphone
- 0627C7M / 1150E IP Deskphone
- 0626C7M / 1165E IP Deskphone
- 062AC7M / 1210 IP Deskphone
- 062AC7M / 1220 IP Deskphone
- 062AC7M / 1230 IP Deskphone

Protocol description

Query Audio Manager
(16 xx 00 xx…)
Note:
Ensure that the handshake commands
1A 04 01 08
1A 07 07 01 23 45 67
are sent to i2004 before sending the commands in column 2. (Requests
attributes of the Audio manager)
16 05 00 01 00
Note: Last byte can contain any value. The message length should be 5.
If the length is wrong it is ignored e.g. send

16 04 00 01
16 06 00 01 00 03
(Requests options setting of the Audio manager)
16 05 00 02 03
Note: Last byte can contain any value. The message length should be 5.
If the length is wrong it is ignored.
(Requests Alerting selection)
16 05 00 04 0F
Note: Last byte can contain any value. The message length should be 5.
If the length is wrong it is ignored.
(Requests adjustable Rx volume information command)
16 05 00 08 00
Note: Last byte can contain any value. The message length should be 5.
If the length is wrong it is ignored.
(Requests the i2004 to send the APB's Default Rx Volume command. The
APB Number or stream based tone is provided in the last byte of the
command below)
16 05 00 10 00 (none)
16 05 00 10 01 (Audio parameter bank 1, NBHS)
16 05 00 10 02 (Audio parameter bank 2, NBHDS)
16 05 00 10 03 (Audio parameter bank 3, NBHF)
16 05 00 10 04 (Audio parameter bank 4, WBHS)
16 05 00 10 05 (Audio parameter bank 5, WBHDS)
16 05 00 10 06 (Audio parameter bank 6, WBHF)
16 05 00 10 07 (Audio parameter bank 7,)
16 05 00 10 08 (Audio parameter bank 8,)
16 05 00 10 09 (Audio parameter bank 9,)
16 05 00 10 0A (Audio parameter bank 0xA,)
16 05 00 10 0B (Audio parameter bank 0xB,)
16 05 00 10 0C (Audio parameter bank 0xC,)
16 05 00 10 0D (Audio parameter bank 0xD,)
16 05 00 10 0E (Audio parameter bank 0xE,)
16 05 00 10 0F (Audio parameter bank 0xF,)
16 05 00 10 10 (Alerting tone)
16 05 00 10 11 (Special tones)
16 05 00 10 12 (Paging tones)
16 05 00 10 13 (Not Defined)
16 05 00 10 1x (Not Defined)
(Set the volume range in configuration message for each of the APBs
and for alerting, paging and special tones (see below) and then send
the following commands)
(Requests handset status, when NBHS is 1) connected 2) disconnected)
16 05 00 40 09
Note: Last byte can contain any value. The message length should be 5.
If the length is wrong it is ignored
(Requests headset status, when HDS is
disconnected)
16 05 00 80 0A
(Requests headset status, when HDS is connected)
16 05 00 80 0A
Note: Last byte can contain any value. The message length should be 5.
If the length is wrong it is ignored
(Requests handset and headset status when NBHS
and HDS are disconnected)
16 05 00 C0 05
(Requests handset and headset status when NBHS
and HDS are connected)
16 05 00 C0 05
(Send an invalid message)
16 03 00
(Send an invalid message. Is this an invalid msg??)
16 06 00 22 22 22
Query Supervisory headset status
(16 03 01)
16 03 01

Audio Manager Options
(16 04 02 xx)
(Maximum tone volume is one level lower than physical maximum
Volume level adjustments are not performed locally in the i2004
Adjustable Rx volume reports not sent to the NI when volume keys are pressed
Single tone frequency NOT sent to HS port while call in progress.
Single tone frequency NOT sent to HD port while call in progress.
Automatic noise squelching disabled.
HD key pressed command sent when i2004 receives make/break sequence.)
16 04 02 00
(Maximum tone volume is set to the physical maximum)

16 04 02 01
then requests options setting of the Audio manager by sending 16 04 00 02)
(Volume level adjustments are performed locally in the i2004)
16 04 02 02
(then requests options setting of the Audio manager by sending 16 04 00 02)
(Adjustable Rx volume reports sent to the NI when volume keys are pressed)
16 04 02 04
(then requests options setting of the Audio manager by sending 16 04 00 02)
(Single tone frequency sent to HS port while call in progress)
16 04 02 08
(then requests options setting of the Audio manager by sending 16 04 00 02)
(Single tone frequency sent to HD port while call in progress)
16 04 02 10
(then requests options setting of the Audio manager by sending 16 04 00 02)
(Automatic noise squelching enabled.)
16 04 02 20
(then requests options setting of the Audio manager by sending 16 04 00 02)
(Headset Rfeature Key Pressed command sent when i2004 receives
make/break sequence.)
16 04 02 40
(then requests options setting of the Audio manager by sending 16 04 00 02)
(In this case both bit 1 and bit 3 are set, hence Volume level
adjustments are performed
locally in the i2004 and Single tone frequency sent to HS port while
call in progress.)
16 04 02 0A

Mute/un-mute
(16 xx 04 xx...)
(In this case two phones are conneted. Phone 1 is given the ID
47.129.31.35 and phone 2
is given the ID 47.129.31.36. Commands are sent to phone 1)
(TX is muted on stream ID 00)
16 05 04 01 00
(TX is un-muted on stream ID 00)
16 05 04 00 00
(RX is muted on stream ID 00)
16 05 04 03 00
(RX is un-muted on stream ID 00)
16 05 04 02 00
(TX is muted on stream ID 00, Rx is un-muted on stream ID 00)
16 07 04 01 00 02 00
(TX is un-muted on stream ID 00, Rx is muted on stream ID 00)
16 07 04 00 00 03 00
(TX is un-muted on stream ID 00, Rx is un-muted on stream ID 00)
16 07 04 00 00 02 00

Transducer Based tone on
(16 04 10 xx)
(Alerting on)
16 04 10 00
(Special tones on, played at down loaded tone volume level)
16 04 10 01
(paging on)
16 04 10 02
(not defined)
16 04 10 03
(Alerting on, played at two steps lower than down loaded tone volume level)
16 04 10 08
(Special tones on, played at two steps lower than down loaded tone volume level)
16 04 10 09

Transducer Based tone off
(16 04 10 xx)
16 04 11 00 (Alerting off)
16 04 11 01 (Special tones off)
16 04 11 02 (paging off)
16 04 11 03 (not defined)

Alerting tone configuration
(16 05 12 xx xx)
(Note: Volume range is set here for all tones. This should be noted
when testing the volume level message)
(HF speaker with different warbler select values, tone volume range set to max)
16 05 12 10 00
16 05 12 11 0F
16 05 12 12 0F

16 05 12 13 0F
16 05 12 14 0F
16 05 12 15 0F
16 05 12 16 0F
16 05 12 17 0F
(HF speaker with different cadence select values, tone volume range set to max)
16 05 12 10 0F
16 05 12 10 1F
16 05 12 10 2F
16 05 12 10 3F
16 05 12 10 4F
16 05 12 10 5F
16 05 12 10 6F
16 05 12 10 7F (configure cadence with alerting tone cadence download message before sending this message)
(HS speaker with different warbler select values, tone volume level set to max)
16 05 12 00 0F
16 05 12 01 0F
16 05 12 02 0F
16 05 12 03 0F
16 05 12 04 0F
16 05 12 05 0F
16 05 12 06 0F
16 05 12 07 0F
(HS speaker with different cadence select values, tone volume range set to max)
16 05 12 00 0F
16 05 12 00 1F
16 05 12 00 2F
16 05 12 00 3F
16 05 12 00 4F
16 05 12 00 5F
16 05 12 00 6F
16 05 12 00 7F (configure cadence with alerting tone cadence download message before sending this message)
(HD speaker with different warbler select values, tone volume range set to max)
16 05 12 08 0F
16 05 12 09 0F
16 05 12 0A 0F
16 05 12 0B 0F
16 05 12 0C 0F
16 05 12 0D 0F
16 05 12 0E 0F
16 05 12 0F 0F
(HD speaker with different cadence select values, tone volume level set to max)
16 05 12 08 0F
16 05 12 08 1F
16 05 12 08 2F
16 05 12 08 3F
16 05 12 08 4F
16 05 12 08 5F
16 05 12 08 6F
16 05 12 08 7F (configure cadence with alerting tone cadence download message before sending this message)

Special tone configuration
(16 06 13 xx xx)
(Note: Volume range is set here for all tones. This should be noted when testing the volume level message)
(HF speaker with different tones, tone volume range is varied)
16 06 13 10 00 01
16 06 13 10 01 01
16 06 13 10 08 01
16 06 13 10 02 07
16 06 13 10 03 07
16 06 13 10 04 11
16 06 13 10 05 11
16 06 13 10 06 18
16 06 13 10 07 18
16 06 13 10 08 1F
(HF speaker with different cadences and tones; tone volume level is varied)
16 06 13 10 00 01
16 06 13 10 10 01
16 06 13 10 20 07
16 06 13 10 30 07
16 06 13 10 40 11
16 06 13 10 50 11
16 06 13 10 60 18

16 06 13 10 70 18 (configure cadence with special tone cadence
download message before sending this message)
(HS speaker with different tones, tone volume range is varied)
16 06 13 00 00 01
16 06 13 00 01 01
16 06 13 00 02 07
16 06 13 00 03 07
16 06 13 00 04 11
16 06 13 00 05 11
16 06 13 00 06 18
16 06 13 00 07 18
(HS speaker with different cadences and tones; tone volume range is varied)
16 06 13 00 00 01
16 06 13 00 10 01
16 06 13 00 20 07
16 06 13 00 30 07
16 06 13 00 40 11
16 06 13 00 50 11
16 06 13 00 60 18
16 06 13 00 70 18 (configure cadence with special tone cadence
download message before sending this message)
(HD speaker with different tones, tone volume range is varied)
16 06 13 08 00 01
16 06 13 08 01 01
16 06 13 08 02 07
16 06 13 08 03 07
16 06 13 08 04 11
16 06 13 08 05 11
16 06 13 08 06 18
16 06 13 08 07 18
(HD speaker with different cadences and tones; tone volume range is varied)
16 06 13 08 00 01
16 06 13 08 10 01
16 06 13 08 20 07
16 06 13 08 30 07
16 06 13 08 40 11
16 06 13 08 50 11
16 06 13 08 60 18
16 06 13 08 70 18 (configure cadence with special tone cadence
download message before sending this message)

Paging tone configuration
(16 05 14 xx xx)
(Note: Volume range is set here for all tones. This should be noted
when testing the volume level message)
(HF speaker with different cadence select values, tone volume range set to max)
16 05 14 10 0F
16 05 14 10 1F
16 05 14 10 2F
16 05 14 10 3F
16 05 14 10 4F
16 05 14 10 5F
16 05 14 10 6F
16 05 14 10 7F (configure cadence with paging tone cadence download
message before sending this message)
(HS speaker with different cadence select values, tone volume range set to max)
16 05 14 00 0F
16 05 14 00 1F
16 05 14 00 2F
16 05 14 00 3F
16 05 14 00 4F
16 05 14 00 5F
16 05 14 00 6F
16 05 14 00 7F (configure cadence with paging tone cadence download
message before sending this message)
(HD speaker with different cadence select values, tone volume level set to max)
16 05 14 08 0F
16 05 14 08 1F
16 05 14 08 2F
16 05 14 08 3F
16 05 14 08 4F
16 05 14 08 5F
16 05 14 08 6F
16 05 14 08 7F (configure cadence with paging tone cadence download
message before sending this message)

Alerting Tone Cadence Download

(16 xx 15 xx xx...)
16 08 15 00 0A 0f 14 1E
(.5 sec on, 0.75 sec off; 1 sec on 1.5 sec off, cyclic)
16 0C 15 01 0A 0f 14 1E 05 0A 0A 14
(.5 sec on, 0.75 sec off; 1 sec on 1.5 sec off; 0.25sec on, 0.5sec
off; 0.5 sec on, 1 sec off , one shot)

Special Tone Cadence Download
(16 xx 16 xx xx...)
16 05 16 0A 10
(125ms on, 200 ms off)
16 09 16 0A 10 14 1E
(125ms on, 200 ms off; 250ms on, 375ms off)

Paging Tone Cadence Download
(16 xx 17 xx xx...)
16 06 17 01 0A 10
(125ms on, 200 ms off, 250HZ)
16 06 17 04 05 10
(62.5ms on, 200 ms off, 500 Hz)
16 09 17 01 0A 10 10 14 1E
(125ms on, 200 ms off; 250ms on, 375ms off, 250 Hz, 100Hz)
16 0C 17 01 0A 10 04 14 1E 10 0A 10
(125ms on, 200 ms off; 250ms on, 375ms off; 125ms on, 200 ms off,
250Hz, 1000Hz, 500 Hz)
16 0C 17 01 1E 10 12 3c 1E 10 28 10
(375ms on, 200 ms off; 750ms on, 375ms off; 500ms on, 200 ms off,
250Hz, (333Hz,1000Hz), 500 Hz)

Transducer Based Tone Volume Level
(16 04 18 xx)
(Ensure that the volume range is set properly in the alerting, special
and paging
tone configuration e.g if the volume range is set to zero, this
message will always output
max volume) (Different volume level for alerting tone. Note: Send the
command below
and then send the alerting on command and alerting off commands)
16 04 18 00
16 04 18 10
16 04 18 20
16 04 18 30
16 04 18 40
16 04 18 50
16 04 18 60
16 04 18 70
16 04 18 80
16 04 18 90
16 04 18 F0
(HF:Volume range for alerting tone is changed here using these commands)
16 05 12 10 0F
16 05 12 10 00
16 05 12 10 04
(HD:Volume range for alerting tone is changed here using these commands)
16 05 12 08 0F
16 05 12 08 00
16 05 12 08 04
(Different volume level for special tone)
16 04 18 01
16 04 18 11
16 04 18 21
16 04 18 31
16 04 18 41
16 04 18 51
16 04 18 61
16 04 18 71
16 04 18 81
16 04 18 91
16 04 18 A1
16 04 18 B1
16 04 18 C1
16 04 18 D1
16 04 18 E1
16 04 18 F1
(HF:Volume range for special tone is changed here using these commands)
16 06 13 10 20 07
16 06 13 10 25 07

16 06 13 10 2F 07
(HD:Volume range for special tone is changed here using these commands)
16 06 13 08 20 07
16 06 13 08 25 07
16 06 13 08 2F 07
(Different volume level for paging tone)
16 04 18 02
16 04 18 12
16 04 18 22
16 04 18 32
16 04 18 42
16 04 18 52
16 04 18 62
16 04 18 72
16 04 18 82
16 04 18 92
16 04 18 F2
(HF:Volume range for paging tone is changed here using these commands)
16 05 14 10 0F
16 06 14 10 00
16 06 14 10 04
(HD:Volume range for paging tone is changed here using these commands)
16 06 14 08 0F
16 06 14 08 00
16 06 14 08 04

Alerting Tone Test
(16 04 19 xx)
(tones 667Hz, duration 50 ms and 500Hz duration 50 ms)
16 04 19 00
(tones 333Hz, duration 50 ms and 250Hz duration 50 ms)
16 04 19 01
(tones 333 Hz + 667 Hz duration 87.5 ms and 500Hz + 1000Hz duration 87.5 ms)
16 04 19 02
(tones 333 Hz, duration 137.5 ms; 500Hz duration 75 ms; 667Hz duration 75 ms)
16 04 19 03
(tones 500Hz, duration 100 ms and 667Hz duration 100 ms)
16 04 19 04
(tones 500Hz, duration 400 ms and 667Hz duration 400 ms)
16 04 19 05
(tones 250Hz, duration 100 ms and 333Hz duration 100 ms)
16 04 19 06
(tones 250Hz, duration 400 ms and 333 Hz, duration 400ms)
16 04 19 07

Visual Transducer Based Tones Enable
(16 04 1A xx)
Visual tone enabled
16 04 1A 01
(Visual tone disabled)
16 04 1A 00

Stream Based Tone On
(16 06 1B xx xx xx)
(Dial tone is summed with data on Rx stream 00 at volume level -3dBm0)
16 06 1B 00 00 08
(Dial tone replaces the voice on Rx stream 00 at volume level -6dBm0)
16 06 1B 80 00 10
(Dial tone is summed with voice on Tx stream 00 at volume level -3dBm0)
16 06 1B 40 00 08
(Dial tone replaces the voice on Tx stream 00 at volume level -3dBm0)
16 06 1B C0 00 08

(Line busy tone is summed with data on Rx stream 00 at volume level -3dBm0)
16 06 1B 02 00 08
(Line busy tone replaces the voice on Rx stream 00 at volume level -6dBm0)
16 06 1B 82 00 10
(Line busy tone is summed with voice on Tx stream 00 at volume level -3dBm0)
16 06 1B 42 00 08
(Line busy tone replaces the voice on Tx stream 00 at volume level -3dBm0)
16 06 1B C2 00 08

(ROH tone is summed with data on Rx stream 00 at volume level -3dBm0)
16 06 1B 05 00 08
(ROH tone replaces the voice on Rx stream 00 at volume level -6dBm0)
16 06 1B 85 00 10
(ROH tone is summed with voice on Tx stream 00 at volume level -3dBm0)
16 06 1B 45 00 08

(ROH tone replaces the voice on Tx stream 00 at volume level -3dBm0)
16 06 1B C5 00 08
(Recall dial tone is summed with data on Rx stream 00 at volume level -3dBm0)
16 06 1B 01 00 08
(Recall dial tone replaces the voice on Rx stream 00 at volume level -6dBm0)
16 06 1B 81 00 10

(Reorder tone is summed with data on Rx stream 00 at volume level -3dBm0)
16 06 1B 03 00 08
(Reorder dial tone replaces the voice on Rx stream 00 at volume level -6dBm0)
16 06 1B 83 00 10

(Audible Ringing tone is summed with data on Rx stream 00 at volume level -3dBm0)
16 06 1B 04 00 08
(Audible Ringing tone replaces the voice on Rx stream 00 at volume level -6dBm0)
16 06 1B 84 00 10

(Stream based tone ID 06 is summed with data on Rx stream 00 at volume level -3dBm0;
Tone ID 06 is downloaded using both the frequency and cadence down load commands)
16 06 1B 06 00 08
(Stream based tone ID 06 replaces the voice on Rx stream 00 at volume level -6dBm0)
16 06 1B 86 00 10

(Stream based tone ID 0F is summed with data on Rx stream 00 at volume level -3dBm0;
Tone ID 0x0F is downloaded using both the frequency and cadence down load commands)
16 06 1B 0F 00 08
(Stream based tone ID 0F replaces the voice on Rx stream 00 at volume level -6dBm0)
16 06 1B 8F 00 10

Stream Based Tone Off
(16 05 1C xx xx)
(Dial tone is turned off on Rx stream 00)
16 05 1C 00 00
(Dial tone is turned off on Tx stream 00)
16 05 1C 40 00
(Line busy tone is turned off on Rx stream 00)
16 05 1C 02 00
(Line busy tone is turned off on Tx stream 00)
16 05 1C 42 00
(ROH tone is turned off on Rx stream 00)
16 05 1C 05 00
(ROH tone is turned off on Tx stream 00)
16 05 1C 45 00
(Recall dial tone is turned off on Rx stream 00)
16 05 1C 01 00
(Reorder tone is turned off on Rx stream 00)
16 05 1C 03 00
(Audible Ringing tone is turned off on Rx stream 00)
16 05 1C 04 00
(Stream based tone ID 06 is turned off on Rx stream 00)
16 05 1C 06 00
(Stream based tone ID 0F is turned off on Rx stream 00)
16 05 1C 0F 00

Stream Based Tone Frequency Component List Download (up to 4 frequencies can be specified)
(16 xx 1D xx...)
Note: Frequency component download and cadence download commands sent to the i2004 first.
Then send the stream based tone ID on command to verify that tones are turned on.
16 06 1D 06 2C CC
(1400Hz)
16 08 1D 07 2C CC 48 51
(1400 Hz and 2250Hz)

Stream Based Tone Cadence Download (up to 4 cadences can be specified)
(16 xx 1E xx...)
Note: Frequency component download and cadence download commands sent to the i2004 first. Then
send the stream based tone ID on command to verify that tones are turned on.

16 06 1E 26 0A 0A
(200 ms on and 200 ms off with tone turned off after the full sequence)
16 08 1E 07 0A 0A 14 14
(20 ms on and 20 ms off for first cycle, 400 ms on and 400 ms off fo
rthe second cycle with sequence repeated)
16 05 1E 26 0A
(In this case tone off period is not specified hence tone is played
until stream based
tone off command is received.

Select Adjustable Rx Volume
(16 04 20 xx)
16 04 20 01
(Audio parameter block 1)
16 04 20 03
(Audio parameter block 3)
16 04 20 08
(Alerting Rx volume)
16 04 20 09
(Special tone Rx volume)
16 04 20 0a
(Paging tone Rx volume)

Set APB's Rx Volume Levels
(16 05 21 xx xx)
16 05 21 01 25
(? Rx volume level 5 steps louder than System RLR)
16 05 21 01 05
(? Rx volume level 5 steps quieter than System RLR)

Change Adjustable Rx Volume
16 03 22
(Rx volume level is one step quieter for the APB/tones selected
through Select Adjustable Rx Volume command)
16 03 23
(Rx volume level is one step louder for the APB/tones selected through
Select Adjustable Rx Volume command)

Adjust Default Rx Volume
(16 04 24 xx)
16 04 24 01
(Default Rx volume level is one step quieter for the APB 1)
16 04 25 01
(Default Rx volume level is one step louder for the APB 1)

Adjust APB's Tx and/or STMR Volume Level
(16 04 26 xx)
(First ensure that the Tx and STMR volume level are set to maximum by
repeatedly (if needed) sending the command
16 04 26 F2 to APB2.
Rest of the commands are sent to i2004 individually and then the query
command below is used to verify
if the commands are sent correctly) (Enable both Tx Vol adj. and STMR
adj; Both Tx volume and STMR volume
are one step louder on APB 2)
16 04 26 F2
(Enable both Tx Vol adj. and STMR adj; Both Tx volume and STMR volume
are one step quieter on APB 2)
16 04 26 A2
(Enable Tx Vol adj. and disable STMR adj; Tx volume is one step louder
on APB 3)
16 04 26 C3
(Enable Tx Vol adj. and disable STMR adj; Tx volume is one step
quieter on APB 3)
16 04 26 83
(Disable both Tx Vol adj. and STMR adj on APB 1)
16 04 26 01

Query APB's Tx and/or STMR Volume Level
(16 04 27 XX)
(Query Tx volume level and STMR volume level on APB 2)
16 04 27 32
(Query STMR volume level on APB 1)
16 04 27 11
(Query STMR volume level on APB 2)
16 04 27 12
(Query STMR volume level on APB 3)
16 04 27 13

(Query Tx volume level on APB 1)
16 04 27 21
(Query Tx volume level on APB 2)
16 04 27 22
(Query Tx volume level on APB 3)
16 04 27 23

APB Download
(16 xx-1F xx...)
16 09 28 FF AA 88 03 00 00

Open Audio Stream
(16 xx 30 xx...)
(If Audio stream is already open it has to be closed before another open audio stream command is sent)
16 15 30 00 00 00 00 01 00 13 89 00 00 13 89 00 00 2F 81 1F 23
(Open G711 ulaw Audio stream to 2F.81.1F.9F)
16 15 30 00 00 08 08 01 00 13 89 00 00 13 89 00 00 2F 81 1F 23
(Open G711 Alaw Audio stream to 2F.81.1F.9F)
16 15 30 00 00 12 12 01 00 13 89 00 00 13 89 00 00 2F 81 1F 23
(Open G729 Audio stream to 2F.81.1F.9F)
16 15 30 00 00 04 04 01 00 13 89 00 00 13 89 00 00 2F 81 1F 23
(Open G723? ulaw Audio stream to 2F.81.1F.9F)

Close Audio Stream
(16 05 31 xx xx)
16 05 31 00 00

Connect Transducer
(16 06 32 xx xx xx)
16 06 32 C0 11 00
(Connect the set in Handset mode with no side tone)
16 06 32 C0 01 00
(Connect the set in Handset mode with side tone)
16 06 32 C1 12 00
(Connect the set in Headset mode with no side tone)
16 06 32 C1 02 00
(Connect the set in Headset mode with side tone)
16 06 32 C2 03 00
(Connect the set in Hands free mode)

Frequency Response Specification
(16 xx 33 xx...)
Filter Block Download 16 xx 39 xx
Voice Switching debug 16 04 35 11
(Full Tx, Disable switch loss bit)
16 04 35 12
(Full Rx, Disable switch loss bit)
Voice Switching Parameter Download 16 08 36 01 2D 00 00 02
(APB 1, AGC threshold index 0, Rx virtual pad 0, Tx virtual pad 0, dynamic side tone enabled)
Query RTCP Statistics 16 04 37 12
(queries RTCP bucket 2, resets RTCP bucket info.)
Configure Vocoder Parameters 16 0A 38 00 00 CB 00 E0 00 A0
(For G711 ulaw 20 ms, NB)
16 0A 38 00 08 CB 00 E0 00 A0
(G711 Alaw 20 ms, NB)
16 0A 38 00 00 CB 01 E0 00 A0
(For G711 ulaw 10 ms, WB)
16 0A 38 00 08 CB 01 E0 00 A0
(G711 Alaw 10 ms, WB)
16 08 38 00 12 C1 C7 C5
(For G729 VAD On, High Pass Filter Enabled, Post Filter Enabled)
16 09 38 00 04 C9 C5 C7 C1
(G723 VAD On, High Pass Filter Enabled, Post Filter Enabled at 5.3 KHz)
16 09 38 00 04 C0 C7 C5 C9
(G723 VAD Off, High Pass Filter Enabled, Post Filter Enabled at 5.3 KHz)
16 09 38 00 04 C1 C5 C7 C8
(G723 VAD On, High Pass Filter Enabled, Post Filter Enabled at 6.3 KHz)
16 09 38 00 04 C0 C7 C5 C8
(G723 VAD Off, High Pass Filter Enabled, Post Filter Enabled at 6.3 KHz)
Query RTCP Bucket's SDES Information (39 XX) (The first nibble in the last byte indicates the bucket ID)
16 04 39 21
16 04 39 22
16 04 39 23
16 04 39 24
16 04 39 25

16 04 39 26
16 04 39 27

16 04 39 01
16 04 39 12
16 04 39 23
16 04 39 34
16 04 39 45
16 04 39 56
16 04 39 67

Skinny

chan_skinny stuff

Skinny call logging

Page for details about call logging.

Calls are logged in the devices placed call log (directories->Place Calls) when a call initially connects to another device. Subsequent changes in the device (eg forwarded) are not reflected in the log.

If a call is not placed to a channel they will not be recorded in the log. eg a call to voicemail will not be recorded. You can force these to be recorded by including progress(), then ringing() in the dialplan.

Example (This will produce a logged call):

```
exten => 100,1,NoOp
exten => 100,n,Progress
exten => 100,n,Ringing
exten => 100,n,VoicemailMain(${CALLERID(num)@mycontext,s)
```

Example (This will not):

```
exten => 100,1,NoOp
exten => 100,n,VoicemailMain(${CALLERID(num)@mycontext,s)
```

Skinny Dev Notes

A spot to keep development notes.

Keepalives

Been doing some mucking around with cisco phones. Things found out about keepalives documented here.

It appears the minimum keepalive is 10. Any setting below this reverts to the the device setting 10 seconds.

Keepalive timings seem to vary by device type (and probably firmware).

Device	F/Ware	Proto	1st KA	Behavior w/ no response
7960	7.2(3.0)	6	15 Sec	KA, KA, KA, UNREG
7961	8.5.4(1.6	17	As set	KA, KA*2, KA*2, UNREG
7920	4.0-3-2	5	As set	KA, KA, KA, KA+Reset Conn

For example, with keepalive set to 20:

- the 7960 will UNREG in 75 sec (ka@15, ka@35, ka@55, unreg@75) (straight after registration); or
- the 7960 will UNREG in 80 sec (ka@20, ka@40, ka@60, unreg@80) (after 1 keepalive ack sent);
- the 7961 will UNREG in 120 sec (ka@20, ka@60, ka@100, unreg@120).

Other info:

- Devices appear to consider themselves still registered (with no indication provided to user) until the unregister/reset conn occurs.
- Devices generally do not respond to keepalives or reset their own timings (see below for exception)
- After unregister (but no reset obviously) keepalives are still sent, further, the device now responds to keepalives with a keepalive_ack, but this doesn't affect the timing of their own keepalives.

chan_skinny impact:

- need to revise keepalive timing with is currently set to unregister at 1.1 * keepalive time

Testing wifi (7920 with keepalive set to 20), immediately after a keepalive:

- removed from range for 55 secs - at 58 secs 3 keepalives received, connection remains.
- removed from range for 65 secs - at about 80 secs, connection reset and device reloads.
- server set to ignore 2 keepalives - 3rd keepalive at just under 60secs, connection remains.
- server set to ignore 3 keepalives - 4th keepalive at just under 80secs, connection reset by device anyway.
- looks like timing should be about 3*keepalive (ie 60secs), maybe 5*keepalive for 7961 (v17?)

More on ignoring keepalives at the server (with the 7920) (table below)

- if keepalive is odd, the time used is rounded up to the next even number (ie 15 will result in 16 secs)
- the first keepalive is delayed by 1 sec if keepalive is less than 30, 15 secs if less than 120, else 105 secs
- these two lead to some funny numbers
- if set to 119, the first will be at 135 secs (119 rounded up + 15), and subsequent each 120 secs
- if set to 120, the first will be at 225 secs (120 not rounded + 105), and subsequent each 120 secs
- similarly if set to 29, the first will be 31 then 30, where if set to 30 the first will be 45 then 30
- only tested out to 600 secs (where the first is still delayed by 105 secs)
- device resets the connection 20 secs after the 3rd unreplied keepalive
- keepalives below 20 seem unreliable in that they do not reset the connection
- above 20secs and after the first keepalive, the device will reset at (TRUNC((KA+1)/2)*2)*3+20
- before the first keepalive, add 1 if KA<30, add 15 if KA<120, else add 105
- actually, about a second earlier. After the first missed KA, the next will be about a second early
- not valid for other devices

Set	First (s)	Then (s)	Packets (#)	Reset (s)
20	21	20	3	20
25	27	26	3	20
26	27	26	3	20
29	31	30	3	20
30	45	30	3	20
60	75	60	3	20
90	105	90	3	20
119	135	120	3	20
120	225	120	3	20
600	705	600	3	20

Skinny device stuff

Collection of notes on weird device stuff.

7937 Conference Phone

- firmware appears to have 10 speedial buttons hardcoded into the firmware.

RTP Packetization

Overview

Asterisk currently supports configurable RTP packetization per codec for select RTP-based channels.

Channels

These channel drivers allow RTP packetization on a user/peer/friend or global level:

- chan_sip
- chan_skinny
- chan_h323
- chan_ooh323 (Asterisk-Addons)
- chan_gtalk
- chan_jingle
- chan_motif (Asterisk 11+)

Configuration

To set a desired packetization interval on a specific codec, append that inteval to the allow= statement.

Example:

```
allow=ulaw:30,alaw,g729:60
```

No packetization is specified in the case of alaw in this example, so the default of 20ms is used.

Autoframing

In addition, chan_sip has the ability to negotiate the desired framing at call establishment.

In `sip.conf` if `autoframing=yes` is set in the global section, then all calls will try to set the packetization based on the remote endpoint's preferences. This behaviour depends on the endpoints ability to present the desired packetization (`ptime\:`) in the SDP. If the endpoint does not include a ptime attribute, the call will be established with 20ms packetization.

Autoframing can be set at the global level or on a user/peer/friend basis. If it is enabled at the global level, it applies to all users/peers/friends regardless of their prefered codec packetization.

Codec framing options

The following table lists the minimum and maximum values that are valid per codec, as well as the increment value used for each. Please note that the maximum values here are only recommended maximums, and should not exceed the RTP MTU.

Name	Minimum (ms)	Maximum (ms)	Default (ms)	Increment (ms)
g723	30	300	30	30
gsm	20	300	20	20
ulaw	10	150	20	10
alaw	10	150	20	10
g726	10	300	20	10
ADPCM	10	300	20	10
SLIN	10	70	20	10
lpc10	20	20	20	20
g729	10	230	20	10
speex	10	60	20	10
ilbc	30	30	30	30
g726_aal2	10	300	20	10

Invalid framing options are handled based on the following rules:

1. If the specified framing is less than the codec's minimum, then the minimum value is used.
2. If the specific framing is greater than the codec's maximum, then the maximum value is used

3. If the specificed framing does not meet the increment requirement, the specified framing is rounded down to the closest valid framing options.

IP Quality of Service

Introduction

Asterisk supports different QoS settings at the application level for various protocols on both signaling and media. The 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.

Asterisk running on Linux can also set 802.1p CoS marks in VLAN packets for the VoIP protocols it uses. This is useful when working in a 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]
```

The table below shows all VoIP channel drivers and other Asterisk modules that support QoS settings for network traffic. It also shows the type(s) of traffic for which each module can support setting QoS settings:

	Signaling	Audio	Video	Text
chan_sip	+	+	+	+
chan_skinny	+	+	+	
chan_mgcp	+	+		
chan_unistm	+	+		
chan_h323		+		
chan_iax2	+			
chan_pjsip	+	+	+	
DUNDI	+ (tos setting)			
IAXProv	+ (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 a Linux system, Asterisk must be compiled with libcap in order to use the ef tos setting if Asterisk is not run as root.

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

ToS decimal equivalence table:

name	decimal value
cs0	0
cs1	32
af11	40
af12	48
af13	56
cs2	64
af21	72
af22	80
af23	88
cs3	96
af31	104

af32	112
af33	120
cs4	128
af41	136
af42	144
af43	152
cs5	160
ef	184
cs6	192
cs7	224

802.1p CoS values

Because 802.1p uses 3 bits of the VLAN header, this parameter can take integer values from 0 to 7.

Recommended values

The recommended values shown below are 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.

CHAN_SIP

In chan_sip, there are four 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 control what TOS values are used for RTP audio, video, and text packets, respectively.
There are four parameters to control 802.1p CoS: "cos_sip", "cos_audio", "cos_video" and "cos_text". The behavior of these parameters is the same as for the SIP TOS settings described above.

CHAN_PJSIP

In chan_pjsip, there are three parameters that control the TOS settings: a **tos** option for a **type=transport** that controls the TOS of SIP signaling packets, a **tos_audio** option for a **type=endpoint** that controls the TOS of RTP audio packets, and a **tos_video** option for a **type=endpoint** that controls the TOS of video packets.

Similarly, there are there parameters that control the 802.1p CoS settings: a **cos** option for a **type=transport** that controls the 802.1p value for SIP signaling packets, a **cos_audio** option for a **type=endpoint** that controls the 802.1p value of RTP audio packets, and a **cos_video** option for a **type=endpoint** that controls the 802.1p value for video packets.

> Changes to a chan_pjsip **type=transport** require an Asterisk restart to be affected. They are not affected by simply reloading Asterisk.

Other RTP channels

chan_mgcp, chan_h323, chan_skinny and chan_unistim also support TOS and CoS via setting tos and cos parameters in their corresponding configuration files. Naming style and behavior are the same as for chan_sip.

Reference

IEEE 802.1Q Standard: http://standards.ieee.org/getieee802/download/802.1Q-1998.pdfRelated 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

AudioSocket

AudioSocket

AudioSocket is a simple TCP-based protocol for sending and receiving real-time audio streams.

There exists a protocol definition (below), a Go library, and Asterisk application and channel interfaces.

Protocol definition

The singular design goal of AudioSocket is to present the simplest possible audio streaming protocol, initially based on the constraints of Asterisk audio. Each packet contains a three-byte header and a variable payload. The header is composed of a one-byte type and a two-byte length indicator.

The minimum message length is three bytes: type and payload-length. Hangup indication, for instance, is `0x00 0x00 0x00`.

Types

- `0x00` - Terminate the connection (socket closure is also sufficient)
- `0x01` - Payload will contain the UUID (16-byte binary representation) for the audio stream
- `0x10` - Payload is signed linear, 16-bit, 8kHz, mono PCM (little-endian)
- `0xff` - An error has occurred; payload is the (optional) application-specific error code. Asterisk-generated error codes are listed below.

Payload length

The payload length is a 16-bit unsigned integer (big endian) indicating how many bytes are in the payload.

Payload

The content of the payload is defined by the header: type and length.

Dialplan

The Asterisk dialplan

The dialplan is essentially a scripting language specific to Asterisk and one of the primary ways of instructing Asterisk on how to behave. It ties everything together, allowing you to route and manipulate calls in a programmatic way. The pages in this section will describe what the elements of dialplan are and how to use them in your configuration.

Dialplan configuration file

The Asterisk dialplan is found in the extensions.conf file in the **configuration directory**, typically /etc/asterisk.

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

Example dialplan

The example dial plan, in the configs/samples/extensions.conf.sample file is installed as extensions.conf if you run "make samples" after installation of Asterisk. The sample file includes many examples of dialplan programming for specific scenarios and environments often common to Asterisk implementations.

On This Page

Topics

Contexts, Extensions, and Priorities

Dialplan Format

The dialplan in extensions.conf is organized into sections, called contexts. Contexts are the basic organizational unit within the dialplan, and as such, they keep different sections of the dialplan independent from each other. You can use contexts to separate out functionality and features, enforce security boundaries between the various parts of our dialplan, as well as to provide different classes of service to groups of users.

Dialplan contexts

The syntax for a context is exactly the same as any other section heading in the configuration files, as explained in Sections and Settings. Simply place the context name in square brackets. For example, here we define an example context called 'users'.

```
[users]
```

Dialplan extensions

Within each context, we can define one or more **extensions**. An extension is simply a named set of actions. Asterisk will perform each action, in sequence, when that extension number is dialed. The syntax for an extension is:

```
exten => number,priority,application([parameter[,parameter2...]])
```

Let's look at an example extension.

```
exten => 6001,1,Dial(PJSIP/demo-alice,20)
```

In this case, the extension number is **6001**, the priority number is **1**, the application is **Dial()**, and the two parameters to the application are **PJSIP/demo-alice** and **20**.

Dialplan priorities

Within each extension, there must be one or more *priorities*. A priority is simply a sequence number. The first priority on an extension is executed first. When it finishes, the second priority is executed, and so forth.

Priority numbers
Priority numbers must begin with 1, and must increment sequentially. If Asterisk can't find the next priority number, it will terminate the call. We call this *auto-fallthrough*. Consider the example below:

```
exten => 6123,1,do something
exten => 6123,2,do something else
exten => 6123,4,do something different
```

In this case, Asterisk would execute priorities one and two, but would then terminate the call, because it couldn't find priority number three.

Priority letter n

Priority numbers can also be simplified by using the letter **n** in place of the priority numbers greater than one. The letter **n** stands for **next**, and when Asterisk sees priority **n** it replaces it in memory with the previous priority number plus one. Note that you must still explicitly declare priority number one.

```
exten => 6123,1,NoOp()
exten => 6123,n,Verbose("Do something!")
exten => 6123,n,Verbose("Do something different!")
```

Every time an extension and priority is executed Asterisk searches for the next best match in priority sequence.

Consider the dialplan below.

```
exten => 1234,1,Verbose("Valid Number")
exten => 4567,1,Verbose("Another Valid Number")
exten => _.!,1,Verbose("Catch all for invalid numbers")
exten => _.!,n,Verbose("Surprise - executed for all numbers!")
```

It may not be immediately intuitive, but the "_.!" extension with the "n" priority will be executed after any of the preceding lines are executed.

Application calls

You'll notice that each priority is calling a dialplan application (such as NoOp, or Verbose in the example above). That is how we tell Asterisk to "do something" with the channel that is executing dialplan. See the Applications section for more detail.

Priority labels

You can also assign a label (or alias) to a particular priority number by placing the label in parentheses directly after the priority number, as shown below. Labels make it easier to jump back to a particular location within the extension at a later time.

```
exten => 6123,1,NoOp()
exten => 6123,n(repeat),Verbose("Do something!")
exten => 6123,n,Verbose("Do something different!")
```

Here, we've assigned a label named **repeat** to the second priority.

Included in the Asterisk 1.6.2 branch (and later) there is a way to avoid having to repeat the extension name/number or pattern using the **same =>** prefix.

```
exten => 6123,1,NoOp()
 same => n(repeat),Verbose("Do something!")
 same => n,Verbose("Do something different!")
```

Dialplan search order

The order of matching within a context is always exact extensions, pattern match extensions, include statements, and switch statements. Includes are always processed depth-first. So for example, if you would like a switch "A" to match before context "B", simply put switch "A" in an included context "C", where "C" is included in your original context before "B".

Search order:

- Explicit extensions
- Pattern match extensions
- Includes
- Switches

Make sure to see the Pattern Matching page for a description of pattern matching order.

Special Dialplan Extensions

Special Asterisk Dialplan Extensions

Here we'll list all of the special built-in dialplan extensions and their usage.

> Other than special extensions, there is a special context "default" that 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).

a: Assistant extension

This extension is similar to the **o** extension, only it gets triggered when the caller presses the asterisk (*) key while recording a voice mail message. This is typically used to reach an assistant.

e: Exception Catchall extension

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.

h: Hangup extension

When a call is hung up, Asterisk executes the **h** extension in the current context. This is typically used for some sort of clean-up after a call has been completed.

i: Invalid entry extension

If Asterisk can't find an extension in the current context that matches the digits dialed during the **Background()** or **WaitExten()** applications, it will send the call to the **i** extension. You can then handle the call however you see fit.

o: Operator extension

If a caller presses the zero key on their phone keypad while recording a voice mail message, and the **o** extension exists, the caller will be redirected to the o extension. This is typically used so that the caller can press zero to reach an operator.

s: Start extension

When an analog call comes into Asterisk, the call is sent to the **s** extension. The s extension is also used in macros.

Please note that the **s** extension is **not** a catch-all extension. It's simply the location that analog calls and macros begin. In our example above, it simply makes a convenient extension to use that can't be easily dialed from the **Background()** and **WaitExten()** applications.

t: Response timeout extension

When the caller waits too long before entering a response to the **Background()** or **WaitExten()** applications, and there are no more priorities in the current extension, the call is sent to the t extension.

T: Absolute timeout extension

This is the extension that is executed when the 'absolute' timeout is reached. See "core show function TIMEOUT" for more information on setting timeouts.

Include Statements

Include statements allow us to split up the functionality in our dialplan into smaller chunks, and then have Asterisk search multiple contexts for a dialed extension. Most commonly, this functionality is used to provide security boundaries between different classes of callers.

It is important to remember that when calls come into the Asterisk dialplan, they get directed to a particular context by the channel driver. Asterisk then begins looking for the dialed extension in the context specified by the channel driver. By using include statements, we can include other contexts in the search for the dialed extension.

Asterisk supports two different types of include statements: regular includes and time-based includes.

Include Statements Basics

To set the stage for our explanation of include statements, let's say that we want to organize our dialplan and create a new context called **features**. We'll leave our extensions **6001** and **6002** for Alice and Bob in the **users** context, and place extensions such as **6500** in the new **features** context. When calls come into the users context and doesn't find a matching extension, the include statement tells Asterisk to also look in the new **features** context.

The syntax for an include statement is very simple. You simply write **include =>** and then the name of the context you'd like to include from the existing context. If we reorganize our dialplan to add a **features** context, it might look something like this:

```
[users]
include => features

exten => 6001,1,Dial(SIP/demo-alice,20)
   same => n,VoiceMail(6001@vm-demo,u)

exten => 6002,1,Dial(SIP/demo-bob,20)
   same => n,VoiceMail(6002@vm-demo,u)

[features]
exten => 6000,1,Answer(500)
   same => n,Playback(hello-world)
   same => n,Hangup()

exten => 6500,1,Answer(500)
   same => n,VoiceMailMain(@vm-demo)
```

Location of Include Statements
Please note that in the example above, we placed the include statement before extensions **6001** and **6002**. It could have just as well come after. The order in which Asterisk tries to find a matching extension is always current context first, then all the include statements.

Be careful with overlapping patterns/extensions
Because Asterisk doesn't stop processing the dialplan after the first matching extension is found, always ensure that you don't have overlapping patterns or duplicate extensions among included contexts, or else you'll get an unexpected behavior.
To prevent convoluted bugs it's recommended to end each extension with a Hangup call to explicitly exit the dialplan.

How calling 6001 may go wrong

```
[users]
include => features
include => catchall

exten => 6001,1,Dial(SIP/demo-alice,20) ; <- Priority 1
   same => n,VoiceMail(6001@vm-demo,u)  ; <- Priority 2

exten => 6002,1,Dial(SIP/demo-bob,20)
   same => n,VoiceMail(6002@vm-demo,u)

[features]
exten => 6000,1,Answer(500)
   same => n,Playback(hello-world)
   same => n,Hangup()
exten => 6500,1,Answer(500)
   same => n,VoiceMailMain(@vm-demo)

[catchall]
exten => _.,1,NoOp();
exten => _.,2,NoOp();
exten => _.,3,NoOp();  ; <- Priority 3 ends up being here, which is NOT what you want
```

Using Include Statements to Create Classes of Service

Now that we've shown the basic syntax of include statements, let's put some include statements to good use. Include statements are often used to build chains of functionality or classes of service. In this example, we're going to build several different contexts, each with its own type of outbound calling. We'll then use include statements to chain these contexts together.

Numbering Plans
The examples in this section use patterns designed for the North American Number Plan, and may not fit your individual circumstances. Feel free to use this example as a guide as you build your own dialplan.

In these examples, we're going to assuming that a seven-digit number that does not begin with a zero or a one is a local (non-toll) call. Ten-digit numbers (where neither the first or fourth digits begin with zero or one) are also treated as local calls. A one, followed by ten digits (where neither the first or fourth digits begin with zero or one) is considered a long-distance (toll) call. Again, feel free to modify these examples to fit your own particular circumstances.

Outbound dialing
These examples assume that you have a SIP provider named provider configured in **sip.conf**. The examples dial out through this SIP provider using the **SIP/provider/number** syntax.
Obviously, these examples won't work unless you setup a SIP provider for outbound calls, or replace this syntax with some other type of outbound connection.

First, let's create a new context for local calls.

```
[local]
; seven-digit local numbers
exten => _NXXXXXX,1,Dial(SIP/provider/${EXTEN})

; ten-digit local numbers
exten => _NXXNXXXXXX,1,Dial(SIP/provider/${EXTEN})

; emergency services (911), and other three-digit services
exten => NXX,1,Dial(SIP/provider/${EXTEN})

; if you don't find a match in this context, look in [users]
include => users
```

Remember that the variable **${EXTEN}** will get replaced with the dialed extension. For example, if Bob dials **5551212** in the **local** context, Asterisk will execute the Dial application with **SIP/provider/5551212** as the first parameter. (This syntax means "Dial out to the account named provider using the SIP channel driver, and dial the number **5551212**.)

Next, we'll build a long-distance context, and link it back to the **local** context with an include statement. This way, if you dial a local number and your phone's channel driver sends the call to the **longdistance** context, Asterisk will search the **local** context if it doesn't find a matching pattern in the **longdistance** context.

```
[longdistance]
; 1+ ten digit long-distance numbers
exten => _1NXXNXXXXXX,1,Dial(SIP/provider/${EXTEN})

; if you don't find a match in this context, look in [local]
include => local
```

Last but not least, let's add an **international** context. In North America, you dial 011 to signify that you're going to dial an international number.

```
[international]
; 1+ ten digit long-distance numbers
exten => _011.,1,Dial(SIP/provider/${EXTEN})

; if you don't find a match in this context, look in [longdistance]
include => longdistance
```

And there we have it -- a simple chain of contexts going from most privileged (international calls) down to lease privileged (local calling).

At this point, you may be asking yourself, "What's the big deal? Why did we need to break them up into contexts, if they're all going out the same outbound connection?" That's a great question! The primary reason for breaking the different classes of calls into separate contexts is so that we can enforce some security boundaries.

Do you remember what we said earlier, that the channel drivers point inbound calls at a particular context? In this case, if we point a phone at the **local** context, it could only make local and internal calls. On the other hand, if we were to point it at the **international** context, it could make international and long-distance and local and internal calls. Essentially, we've created different classes of service by chaining contexts together with include statements, and using the channel driver configuration files to point different phones at different contexts along the chain.

Please take the next few minutes and implement a series of chained contexts into your own dialplan, similar to what we've explained above. You can then change the configuration for Alice and Bob (in **sip.conf**, since they're SIP phones) to point to different contexts, and see what happens when you attempt to make various types of calls from each phone.

Switch Statements

Dialplan Switch Statements

The **switch** statement permits a server to share the dialplan with another server. To understand when a switch would be searched for dialplan extensions you should read the Contexts, Extensions, and Priorities section as it covers Dialplan search order.

> Use with care: Reciprocal switch statements are not allowed (e.g. both A -> B and B -> A), and the switched server need to be on-line or else dialing can be severely delayed.

Basic switch statement

As an example, with remote IAX switching you get transparent access to the remote Asterisk PBX.

```
[iaxprovider]
switch => IAX2/user:password@myserver/mycontext
```

The lswitch statement

An "lswitch" is like a switch but is literal, in that variable substitution is not performed at load time but is passed to the switch directly (presumably to be substituted in the switch routine itself)

```
lswitch => Loopback/12${EXTEN}@othercontext
```

The eswitch statement

An "eswitch" is like a switch but the evaluation of variable substitution is performed at runtime before being passed to the switch routine.

```
eswitch => IAX2/context@${CURSERVER}
```

Variables

Variables are used in most programming and scripting languages. In Asterisk, we can use variables to simplify our dialplan and begin to add logic to the system. A variable is simply a container that has both a name and a value. For example, we can have a variable named **COUNT** which has a value of three. Later on, we'll show you how to route calls based on the value of a variable. Before we do that, however, let's learn a bit more about variables. The names of variables are case-sensitive, so **COUNT** is different than **Count** and **count**. Any channel variables created by Asterisk will have names that are completely upper-case, but for your own channels you can name them however you would like.

In Asterisk, we have two different types of variables: *channel variables* and *global variables*.

Channel Variables

What's a channel variable? Read on to find out why they're important and how they'll improve your quality of life.

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.

Parameter Quoting

```
exten => s,5,BackGround,my-custom-sound
```

The parameter (blabla) can be quoted with double quotes (**"**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.

Special characters that must be escaped to be used literally, are "**[**", "**]**", "****" (backslash) and **"** (double quote).
Dollar sign "**$**" does not require escaping, as long as it doesn't trigger variable expansion or expression evaluation (i.e. "**$[**" or "**${**"), - in that case you'd have to either surround it with double quotes or escape the next character with a backslash (See example code below).

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 later.

Useful Examples

```
; Comma truncation
exten => s,1,Verbose(Hi, James!)   ; <- , James!
exten => s,2,Verbose("Hi, James!") ; <- "Hi, James!"

; Variable expansion
exten => s,1,Verbose(${EXTEN} is a standard variable)   ; <- s is a standard variable
exten => s,2,Verbose($\{EXTEN} is a standard variable)  ; <- ${EXTEN} is a standard variable
exten => s,3,Verbose("$"{EXTEN} is a standard variable) ; <- ${EXTEN} is a standard variable

; Expression expansion
exten => s,1,Verbose(:-] smile!)  ; <- (Syntax error)
exten => s,2,Verbose(:-\] smile!) ; <- :-] smile!

; Quote stripping
exten => s,1,Verbose("haxxor1337" is calling)   ; <- haxxor1337 is calling
exten => s,2,Verbose(\"haxxor1337\" is calling) ; <- "haxxor1337" is calling
```

Setting and Substituting Channel 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}.

Here is a simple example.

```
exten => 1,1,Set(COUNT=3)
exten => 1,n,SayNumber(${COUNT})
```

In the second line of this example, Asterisk replaces the ${COUNT} text with the value of the COUNT variable, so that it ends up calling SayNumber(3).

For another example, to stringwise append $varname2 to $varname3 and store result in $varname1, do:

```
exten => 1,2,Set(varname1=${varname2}${varname3})
```

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(varname1=varname2)
exten => 1,3,Set(${varname1}=value)
```

The above dialplan stores to the variable "varname1" the value "varname2" and to variable "varname2" the value "value".

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

Variable Inheritance

Single Inheritance

Variable names which are prefixed by "_" (one underbar character) 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. Meaning it will not be inherited any further than a single level, that is one child channel.

```
exten = 1234,1,Set(_FOO=bar)
```

Multiple Inheritance

If the name is prefixed by "__" (two underbar characters) in the channel, then the variable is inherited and the "__" will remain intact in the new channel. Therefore any channels then created by the new channel will also receive the variable with "__", continuing the inheritance indefinitely.

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.

```
exten = 1234,1,Set(__FOO=bar)
```

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})
```

Asterisk Standard Channel Variables

There are a number of variables that are defined or read by Asterisk. Here is a listing 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.

Variables present in Asterisk 1.8 and forward:

- ${CDR(accountcode)} * - Account code (if specified)
- ${BLINDTRANSFER} - The name of the channel on the other side of a blind transfer
- ${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 information or OLI
- ${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 ${CHANNEL(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
- ${DYNAMIC_PEERNAME} - The name of the channel on the other side when a dynamic feature is used (removed)
- ${DYNAMIC_FEATURENAME} - The name of the last triggered dynamic feature
- ${DYNAMIC_WHO_ACTIVATED} - Gives the channel name that activated the dynamic feature
- ${UNIQUEID} * - Current call unique identifier
- ${SYSTEMNAME} * - value of the systemname option of asterisk.conf
- ${ENTITYID} * - Global Entity ID set automatically, or from asterisk.conf

Variables present in Asterisk 11 and forward:

- ${AGIEXITONHANGUP} - set to `1` to force the behavior of a call to AGI to behave as it did in 1.4, where the AGI script would exit immediately on detecting a channel hangup
- ${CALENDAR_SUCCESS} * - Status of the CALENDAR_WRITE function. Set to `1` if the function completed successfully; `0` otherwise.
- ${SIP_RECVADDR} * - the address a SIP MESSAGE request was received from
- ${VOICEMAIL_PLAYBACKSTATUS} * - Status of the VoiceMailPlayMsg application. `SUCCESS` if the voicemail was played back successfully, {{FAILED} otherwise

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()

Various application variables

- `${CURL}` - Resulting page content for `CURL()`
- `${ENUM}` - Result of application `EnumLookup()`
- `${EXITCONTEXT}` - Context to exit to in IVR menu (`Background()`) or in the `RetryDial()` application
- `${MONITOR}` - Set to "TRUE" if the channel is/has been monitored (app monitor())
- `${MONITOR_EXEC}` - Application to execute after monitoring a call
- `${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
- `${QUEUESTATUS}` - Status of the call, one of: (TIMEOUT | FULL | JOINEMPTY | LEAVEEMPTY | JOINUNAVAIL | LEAVEUNAVAIL)
- `${QUEUEPOSITION}` - When a caller is removed from a queue, his current position is logged in this variable. If the value is 0, then this means that the caller was serviced by a queue member. If non-zero, then this was the position in the queue the caller was in when he left.
- `${RECORDED_FILE}` - Recorded file in record()
- `${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)
- `${TOUCH_MONITOR_MESSAGE_START}` - Recorded file to play for both channels at start of monitoring session
- `${TOUCH_MONITOR_MESSAGE_STOP}` - Recorded file to play for both channels at end of monitoring session
- `${TOUCH_MIXMONITOR}` - The filename base to use with Touch MixMonitor (auto record)
- `${TOUCH_MIXMONITOR_FORMAT}` - The audio format to use with Touch MixMonitor (auto record)
- `${TOUCH_MIXMONITOR_OUTPUT}` - Recorded file from Touch MixMonitor (auto record)
- `${TXTCIDNAME}` - Result of application TXTCIDName
- `${VPB_GETDTMF}` - chan_vpb

MeetMe Channel Variables

- ${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 conference
- ${CONF_LIMIT_TIMEOUT_FILE} - File to play when time is up. Used with the L() option.
- ${CONF_LIMIT_WARNING_FILE} - File to play as warning if 'y' is defined. The default is to say the time remaining. Used with the L() option.
- ${MEETMEBOOKID} * - This variable exposes the bookid column for a realtime configured conference bridge.
- ${MEETME_EXIT_KEY} - DTMF key that will allow a user to leave a conference

VoiceMail Channel Variables

- ${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

VMAuthenticate Channel Variables

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

DUNDiLookup Channel Variables

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

chan_dahdi Channel Variables

- ${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_specific, 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'

chan_sip Channel Variables

- ${SIPCALLID} * - SIP Call-ID: header verbatim (for logging or CDR matching)
- ${SIPDOMAIN} * - SIP destination domain of an inbound call (if appropriate)
- ${SIPFROMDOMAIN} - Set SIP domain portion of From header on outbound calls
- ${SIPUSERAGENT} * - SIP user agent (deprecated)
- ${SIPURI} * - SIP uri
- ${SIP_MAX_FORWARDS} - Set the value of the Max-Forwards header for outbound call
- ${SIP_CODEC} - Set the SIP codec for an inbound call
- ${SIP_CODEC_INBOUND} - Set the SIP codec for an inbound call
- ${SIP_CODEC_OUTBOUND} - Set the SIP codec for an outbound 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

chan_agent Channel Variables

- ${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

Dial Channel Variables

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

Variables present in Asterisk 16.4.0 and forward:

- ${RINGTIME} - Time in seconds between creation of the dialing channel and receiving the first RINGING signal
- ${RINGTIME_MS} - Time in milliseconds between creation of the dialing channel and receiving the first RINGING signal
- ${PROGRESSTIME} - Time in seconds between creation of the dialing channel and receiving the first PROGRESS signal
- ${PROGRESSTIME_MS} - Time in milliseconds between creation of the dialing channel and receiving the first PROGRESS signal
- ${DIALEDTIME_MS} - Time for the call (milliseconds). Is only set if the call was answered.
- ${ANSWEREDTIME_MS} - Time from answer to hangup (milliseconds)

Chanisavail() Channel Variables

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

Dialplan Macros Channel Variables

- ${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

ChanSpy Channel Variables

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

Open Settlement Protocol (OSP) Channel Variables

- ${OSPINHANDLE} - The inbound call OSP transaction handle.
- ${OSPINTOKEN} - The inbound OSP token.
- ${OSPINTIMELIMIT} - The inbound call duration limit in seconds.
- ${OSPINPEERIP} - The last hop IP address.
- ${OSPINNETWORKID} - The inbound source network ID.
- ${OSPINNPRN} - The inbound routing number.
- ${OSPINNPCIC} - The inbound carrier identification code.
- ${OSPINNPDI} - The inbound number portability database dip indicator.
- ${OSPINSPID} - The inbound service provider identity.
- ${OSPINOCN} - The inbound operator company number.
- ${OSPINSPN} - The inbound service provider name.
- ${OSPINALTSPN} - The inbound alternate service provider name.
- ${OSPINMCC} - The inbound mobile country code.
- ${OSPINMNC} - The inbound mobile network code.
- ${OSPINDIVUSER} - The inbound Diversion header user part.
- ${OSPINDIVHOST} - The inbound Diversion header host part.
- ${OSPINTOHOST} - The inbound To header host part.
- ${OSPINCUSTOMINFOn} - The inbound custom information. Where n is the index beginning with 1 upto 8.
- ${OSPOUTHANDLE} - The outbound call OSP transaction handle.
- ${OSPOUTTOKEN} - The outbound OSP token.
- ${OSPOUTTIMELIMIT} - The outbound call duration limit in seconds.
- ${OSPOUTTECH} - The outbound channel technology.
- ${OSPOUTCALLIDTYPES} - The outbound Call-ID types.
- ${OSPOUTCALLID} - The outbound Call-ID. Only for H.323.
- ${OSPDESTINATION} - The destination IP address.
- ${OSPDESTREMAILS} - The number of remained destinations.
- ${OSPOUTCALLING} - The outbound calling number.
- ${OSPOUTCALLED} - The outbound called number.
- ${OSPOUTNETWORKID} - The outbound destination network ID.
- ${OSPOUTNPRN} - The outbound routing number.
- ${OSPOUTNPCIC} - The outbound carrier identification code.
- ${OSPOUTNPDI} - The outbound number portability database dip indicator.
- ${OSPOUTSPID} - The outbound service provider identity.
- ${OSPOUTOCN} - The outbound operator company number.
- ${OSPOUTSPN} - The outbound service provider name.
- ${OSPOUTALTSPN} - The outbound alternate service provider name.
- ${OSPOUTMCC} - The outbound mobile country code.
- ${OSPOUTMNC} - The outbound mobile network code.
- ${OSPDIALSTR} - The outbound Dial command string.
- ${OSPINAUDIOQOS} - The inbound call leg audio QoS string.
- ${OSPOUTAUDIOQOS} - The outbound call leg audio QoS string.

Digit Manipulation Channel Variables

- `${REDIRECTING_CALLEE_SEND_MACRO}`
  Macro to call before sending a redirecting update to the callee
- `${REDIRECTING_CALLEE_SEND_MACRO_ARGS}`
  Arguments to pass to ${REDIRECTING_CALLEE_SEND_MACRO}

—

- `${REDIRECTING_CALLER_SEND_MACRO}`
  Macro to call before sending a redirecting update to the caller
- `${REDIRECTING_CALLER_SEND_MACRO_ARGS}`
  Arguments to pass to ${REDIRECTING_CALLER_SEND_MACRO}

- `${CONNECTED_LINE_CALLEE_SEND_MACRO}`
  Macro to call before sending a connected line update to the callee
- `${CONNECTED_LINE_CALLEE_SEND_MACRO_ARGS}`
  Arguments to pass to ${CONNECTED_LINE_CALLEE_SEND_MACRO}

—

- `${CONNECTED_LINE_CALLER_SEND_MACRO}`
  Macro to call before sending a connected line update to the caller
- `${CONNECTED_LINE_CALLER_SEND_MACRO_ARGS}`
  Arguments to pass to ${CONNECTED_LINE_CALLER_SEND_MACRO}

Case Sensitivity

Case sensitivity of channel variables in Asterisk is dependent on the version of Asterisk in use.

Versions prior to Asterisk 12

This includes versions

- Asterisk 1.0.X
- Asterisk 1.2.X
- Asterisk 1.4.X
- Asterisk 1.6.0.X
- Asterisk 1.6.1.X
- Asterisk 1.6.2.X
- Asterisk 1.8.X
- Asterisk 10.X
- Asterisk 11.X

These versions of Asterisk follow these three rules:

- Variables evaluated in the dialplan are **case-insensitive**
- Variables evaluated within Asterisk's internals are **case-sensitive**
- Built-in variables are **case-sensitive**

This is best illustrated through the following examples

Example 1: A user-set variable

In this example, the user retrieves a value from the AstDB and then uses it as the destination for a `Dial` command.

```
[default]
exten => 1000,1,Set(DEST=${DB(egg/salad)})
    same => n,Dial(${DEST},15)
```

Since the `DEST` variable is set and evaluated in the dialplan, its evaluation is case-insensitive. Thus the following would be equivalent:

```
exten => 1000,1,Set(DEST=${DB(egg/salad)})
    same => n,Dial(${dest},15)
```

As would this:

```
exten => 1000,1,Set(DeSt=${DB(egg/salad)})
    same => n,Dial(${dEsT},15)
```

Example 2: Using a built-in variable

In this example, the user wishes to use a built-in variable in order to determine the destination for a call.

```
exten => _X.,1,Dial(SIP/${EXTEN})
```

Since the variable `EXTEN` is a built-in variable, the following would **not** be equivalent:

```
exten => _X.,1,Dial(SIP/${exten})
```

The lowercase `exten` variable would evaluate to an empty string since no previous value was set for `exten`.

Example 3: A variable used internally by Asterisk

In this example, the user wishes to suggest to the SIP channel driver what codec to use on the call.

```
exten => 1000,Set(SIP_CODEC=g729)
same => n,Dial(SIP/1000,15)
```

`SIP_CODEC` is set in the dialplan, but it gets evaluated inside of Asterisk, so the evaluation is case-sensitive. Thus the following dialplan would not be equivalent:

```
exten => 1000,Set(sip_codec=g729)
    same => n,Dial(SIP/1000,15)
```

This can lead to some rather confusing situations. Consider that a user wrote the following dialplan. He intended to set the variable `SIP_CODEC` but instead made a typo:

```
exten => 1000,Set(SIP_CODEc=g729)
    same => n,Dial(SIP/1000,15)
```

As has already been discussed, this is not equivalent to using SIP_CODEC. The user looks over his dialplan and does not notice the typo. As a way of debugging, he decides to place a NoOp in the dialplan:

```
exten => 1000,Set(SIP_CODEc=g729)
    same => n,NoOp(${SIP_CODEC})
    same => n,Dial(SIP/1000,15)
```

When the user checks the verbose logs, he sees that the second priority has evaluated SIP_CODEC to be "g729". This is because the evaluation in the dialplan was done case-insensitively.

Asterisk 12 and above

Due to potential confusion stemming from the policy, for Asterisk 12, it was proposed that variables should be evaluated consistently. E-mails were sent to the Asterisk-developers and Asterisk-users lists about whether variables should be evaluated case-sensitively or case-insensitively. The majority opinion swayed towards case-sensitive evaluation. Thus in Asterisk 12, all variable evaluation, whether done in the dialplan or internally, will be case-sensitive.

For those who are upgrading to Asterisk 12 from a previous version, be absolutely sure that your variables are used consistently throughout your dialplan.

Global Variables Basics

Global variables are variables that don't live on one particular channel — they pertain to all calls on the system. They have global scope. There are two ways to set a global variable. The first is to declare the variable in the **[globals]** section of **extensions.conf**, like this:

```
[globals]
MYGLOBALVAR=somevalue
```

You can also set global variables from dialplan logic using the **GLOBAL()** dialplan function along with the **Set()** application. Simply use the syntax:

```
exten=>6124,1,Set(GLOBAL(MYGLOBALVAR)=somevalue)
```

To retrieve the value of a global channel variable, use the same syntax as you would if you were retrieving the value of a channel variable.

Manipulating Variables Basics

It's often useful to do string manipulation on a variable. Let's say, for example, that we have a variable named **NUMBER** which represents a number we'd like to call, and we want to strip off the first digit before dialing the number. Asterisk provides a special syntax for doing just that, which looks like **${variable[:skip[:length]}**.

The optional **skip** field tells Asterisk how many digits to strip off the front of the value. For example, if **NUMBER** were set to a value of **98765**, then **${NUMBER:2}** would tell Asterisk to remove the first two digits and return **765**.

If the skip field is negative, Asterisk will instead return the specified number of digits from the end of the number. As an example, if **NUMBER** were set to a value of **98765**, then **${NUMBER:-2}** would tell Asterisk to return the last two digits of the variable, or **65**.

If the optional **length** field is set, Asterisk will return at most the specified number of digits. As an example, if **NUMBER** were set to a value of **98765**, then **${NUMBER:0:3}** would tell Asterisk not to skip any characters in the beginning, but to then return only the three characters from that point, or **987**. By that same token, **${NUMBER:1:3}** would return **876**.

Using the CONTEXT, EXTEN, PRIORITY, UNIQUEID, and CHANNEL Variables

Now that you've learned a bit about variables, let's look at a few of the variables that Asterisk automatically creates.

Asterisk creates channel variables named **CONTEXT**, **EXTEN**, and **PRIORITY** which contain the current context, extension, and priority. We'll use them in pattern matching (below), as well as when we talk about macros in Section 308.10. Macros. Until then, let's show a trivial example of using **${EXTEN}** to read back the current extension number.

```
exten=>6123,1,SayNumber(${EXTEN})
```

If you were to add this extension to the **[users]** context of your dialplan and reload the dialplan, you could call extension **6123** and hear Asterisk read back the extension number to you.

Another channel variable that Asterisk automatically creates is the **UNIQUEID** variable. Each channel within Asterisk receives a unique identifier, and that identifier is stored in the **UNIQUEID** variable. The **UNIQUEID** is in the form of **1267568856.11**, where **1267568856** is the Unix epoch, and **11** shows that this is the eleventh call on the Asterisk system since it was last restarted.

Last but not least, we should mention the **CHANNEL** variable. In addition to a unique identifier, each channel is also given a channel name and that channel name is set in the **CHANNEL** variable. A SIP call, for example, might have a channel name that looks like **SIP/george-0000003b**, for example.

Pattern Matching

The next concept we'll cover is called *pattern matching*. Pattern matching allows us to create extension patterns in our dialplan that match more than one possible dialed number. Pattern matching saves us from having to create an extension in the dialplan for every possible number that might be dialed.

When Alice dials a number on her phone, Asterisk first looks for an extension (in the context specified by the channel driver configuration) that matches exactly what Alice dialed. If there's no exact match, Asterisk then looks for a pattern that matches. After we show the syntax and some basic examples of pattern matching, we'll explain how Asterisk finds the best match if there are two or more patterns which match the dialed number.

Special Characters Used in Pattern Matching

Pattern matches always begin with an underscore. This is how Asterisk recognizes that the extension is a pattern and not just an extension with a funny name. Within the pattern, we use various letters and characters to represent sets or ranges of numbers. Here are the most common letters:

X

The letter **X** or **x** represents a single digit from 0 to 9.

Z

The letter **Z** or **z** represents any digit from 1 to 9.

N

The letter **N** or **n** represents a single digit from 2 to 9.

Now let's look at a sample pattern. If you wanted to match all four-digit numbers that had the first two digits as six and four, you would create an extension that looks like:

```
exten => _64XX,1,SayDigits(${EXTEN})
```

In this example, each **X** represents a single digit, with any value from zero to nine. We're essentially saying "The first digit must be a six, the second digit must be a four, the third digit can be anything from zero to nine, and the fourth digit can be anything from zero to nine".

Character Sets

If we want to be more specific about a range of numbers, we can put those numbers or number ranges in square brackets to define a character set. For example, what if we wanted the second digit to be either a three or a four? One way would be to create two patterns (**_64XX** and **_63XX**), but a more compact method would be to do **_6[34]XX**. This specifies that the first digit must be a six, the second digit can be either a three or a four, and that the last two digits can be anything from zero to nine.

You can also use ranges within square brackets. For example, **[1-468]** would match a single digit from one through four or six or eight. It does not match any number from one to four hundred sixty-eight!

> ⚠ The X, N, and Z convenience notations mentioned earlier have no special meaning within a set.
>
> The only characters with special meaning within a set are the '-' character, to define a range between two characters, the '\' character to escape a special character available within a set, and
> the ']' character which closes the set. The treatment of the '\' character in pattern matching is somewhat haphazard and may not escape any special character meaning correctly.

Other Special Characters

Within Asterisk patterns, we can also use a couple of other characters to represent ranges of numbers. The period character (**.**) at the end of a pattern matches one or more remaining **characters**. You put it at the end of a pattern when you want to match extensions of an indeterminate length. As an example, the pattern _**9876.** would match any number that began with **9876** and had at least one more character or digit.

The exclamation mark (**!**) character is similar to the period and matches zero or more remaining characters. It is used in overlap dialing to dial through Asterisk. For example, **_9876!** would match any number that began with **9876** including **9876**, and would respond that the number was complete as soon as there was an unambiguous match.

Asterisk treats a period or exclamation mark as the end of a pattern. If you want a period or exclamation mark in your pattern as a plain character you should put it into a character set: **[.]** or **[!]**.

Be Careful With Wildcards in Pattern Matches
Please be extremely cautious when using the period and exclamation mark characters in your pattern matches. They match more than just digits. They match on characters. If you're not careful to filter the input from your callers, a malicious caller might try to use these wildcards to bypass security boundaries on your system.

For a more complete explanation of this topic and how you can protect yourself, please refer to the **README-SERIOUSLY.bestpractices.txt** file in the Asterisk source code.

Order of Pattern Matching

Now let's show what happens when there is more than one pattern that matches the dialed number. How does Asterisk know which pattern to choose as the best match?

Asterisk uses a simple set of rules to sort the extensions and patterns so that the best match is found first. The best match is simply the most specific pattern. The sorting rules are:

1. The dash (**-**) character is ignored in extensions and patterns except when it is used in a pattern to specify a range in a character set. It has no effect in matching or sorting extensions.
2. Non-pattern extensions are sorted in ASCII sort order before patterns.
3. Patterns are sorted by the most constrained character set per digit first. By most constrained, we mean the pattern that has the fewest possible matches for a digit. As an example, the **N** character has eight possible matches (two through nine), while **X** has ten possible matches (zero through nine) so **N** sorts first.
4. Character sets that have the same number of characters are sorted in ASCII sort order as if the sets were strings of the set characters. As an example, **X** is **0123456789** and **[a-j]** is **abcdefghij** so **X** sorts first. This sort ordering is important if the character sets overlap as with **[0-4]** and **[4-8]**.
5. The period (**.**) wildcard sorts after character sets.
6. The exclamation mark (**!**) wildcard sorts after the period wildcard.

Let's look at an example to better understand how this works. Let's assume Alice dials extension **6421**, and she has the following patterns in her dialplan:

```
exten => _6XX1,1,SayAlpha(A)
exten => _64XX,1,SayAlpha(B)
exten => _640X,1,SayAlpha(C)
exten => _6.,1,SayAlpha(D)
exten => _64NX,1,SayAlpha(E)
exten => _6[45]NX,1,SayAlpha(F)
exten => _6[34]NX,1,SayAlpha(G)
```

Can you tell (without reading ahead) which one would match?

Using the sorting rules explained above, the extensions sort as follows:
_640X sorts before **_64NX** because of rule 3 at position 4. (0 before N)
_64NX sorts before **_64XX** because of rule 3 at position 4. (N before X)
_64XX sorts before **_6[34]NX** because of rule 3 at position 3. (4 before [34])
_6[34]NX sorts before **_6[45]NX** because of rule 4 at position 3. ([34] before [45])
_6[45]NX sorts before **_6XX1** because of rule 3 at position 3. ([45] before X)
_6XX1 sorts before **_6.** because of rule 5 at position 3. (X before .)

Sorted extensions

```
exten => _640X,1,SayAlpha(C)
exten => _64NX,1,SayAlpha(E)
exten => _64XX,1,SayAlpha(B)
exten => _6[34]NX,1,SayAlpha(G)
exten => _6[45]NX,1,SayAlpha(F)
exten => _6XX1,1,SayAlpha(A)
exten => _6.,1,SayAlpha(D)
```

When Alice dials **6421**, Asterisk searches through its list of sorted extensions and uses the first matching extension. In this case _**64NX** is found.

To verify that Asterisk actually does sort the extensions in the manner that we've shown, add the following extensions to the **[users]** context of your own dialplan.

```
exten => _6XX1,1,SayAlpha(A)
exten => _64XX,1,SayAlpha(B)
exten => _640X,1,SayAlpha(C)
exten => _6.,1,SayAlpha(D)
exten => _64NX,1,SayAlpha(E)
exten => _6[45]NX,1,SayAlpha(F)
exten => _6[34]NX,1,SayAlpha(G)
```

Reload the dialplan, and then type **dialplan show 6421@users** at the Asterisk CLI. Asterisk will show you all extensions that match in the **[users]** context. If you were to dial extension **6421** in the **[users]** context the first found extension will execute.

```
server*CLI> dialplan show 6421@users
[ Context 'users' created by 'pbx_config' ]
  '_64NX' =>        1. SayAlpha(E)                                [pbx_config]
  '_64XX' =>        1. SayAlpha(B)                                [pbx_config]
  '_6[34]NX' =>     1. SayAlpha(G)                                [pbx_config]
  '_6[45]NX' =>     1. SayAlpha(F)                                [pbx_config]
  '_6XX1' =>        1. SayAlpha(A)                                [pbx_config]
  '_6.' =>          1. SayAlpha(D)                                [pbx_config]

-= 6 extensions (6 priorities) in 1 context. =-
```

```
server*CLI> dialplan show users
[ Context 'users' created by 'pbx_config' ]
  '_640X' =>        1. SayAlpha(C)                                [pbx_config]
  '_64NX' =>        1. SayAlpha(E)                                [pbx_config]
  '_64XX' =>        1. SayAlpha(B)                                [pbx_config]
  '_6[34]NX' =>     1. SayAlpha(G)                                [pbx_config]
  '_6[45]NX' =>     1. SayAlpha(F)                                [pbx_config]
  '_6XX1' =>        1. SayAlpha(A)                                [pbx_config]
  '_6.' =>          1. SayAlpha(D)                                [pbx_config]

-= 7 extensions (7 priorities) in 1 context. =-
```

You can dial extension **6421** to try it out on your own.

Be Careful with Pattern Matching
Please be aware that because of the way auto-fallthrough works, if Asterisk can't find the next priority number for the current extension or pattern match, it will also look for that same priority in a less specific pattern match. Consider the following example:

```
exten => 6410,1,SayDigits(987)
exten => _641X,1,SayDigits(12345)
exten => _641X,n,SayDigits(54321)
```

If you were to dial extension **6410**, you'd hear "nine eight seven five four three two one".

We strongly recommend you make the **Hangup()** application be the last priority of any extension to avoid this behaviour, unless you purposely want to fall through to a less specific match.

Matching on Caller ID

Within an extension handler, it is also possible to match based upon the Caller ID of the incoming channel by appending a forward slash to the dialed extension or pattern, followed by a Caller ID pattern to be matched. Consider the following example, featuring phones with Caller IDs of 101, 102 and 103.

```
exten => 306,1,NoOp()
same => n,Background(goodbye)
same => n,Hangup()

exten => 306/_101,1,NoOp()
same => n,Background(year)
same => n,Hangup()

exten => 306/_102,1,NoOp()
same => n,Background(beep)
same => n,Hangup()
```

The phone with Caller ID 101, when dialing 306, will hear the prompt "year" and will be hung up. The phone with Caller ID 102, when dialing 306, will hear the "beep" sound and will be hung up. The phone with Caller ID 103, or any other caller, when dialing 306, will hear the "goodbye" prompt and will be hung up.

Rewriting Caller ID
Changing the value of **CALLERID(num)** variable inside of extension handler matched by Caller ID can immediately **throw the call to another handler**. Consider the following example:

```
[unexpected-jump-test]

exten => s/_1XX,1,Set(CALLERID(num)=200) ; <- Example call starts here
same  =>        2,Hangup()               ; <- This line is NEVER reached normally because of the assignment above

exten => s/_2XX,1,SayDigits(1)
same  =>        2,SayDigits(2)           ; <- This is where the dialplan proceeds instead
same  =>        3,SayDigits(3)
same  =>        4,SayDigits(4)
```

You'd expect the call with Caller ID 100 to hang up, but instead you'd hear Asterisk saying "two, three, four".
As soon as the Caller ID changes, the next priority Asterisk goes to will be inside extension handler that matches the **new** Caller ID.

Subroutines

Subroutines in Asterisk are defined similarly to standard dialplan contexts and are referred to as Macros and Gosubs. They are invoked via the Macro and Gosub applications, but may also be invoked in the context of Pre-Dial Handlers, Pre-Bridge Handlers and Hangup Handlers via the use of flags and arguments within other applications.

Gosub

Overview

Gosub is a dialplan application. It replaces (is recommended in place of, and deprecates) the Macro application.

Gosub allows you to execute a specific block (context or section) of dialplan as well as pass and return information via arguments to/from the scope of the block. Whereas Macro has issues with nesting, Gosub does not and Gosub should be used wherever you would have used a Macro.

Other dialplan applications, such as Dial and Queue make use of Gosub functionality from within their applications. That means they allow you to perform actions like calling Gosub on the called party's channel from a Dial, or on a Queue member's channel after they answer. See the Pre-Dial Handlers and Pre-Bridge Handlers sections for more information.

Defining a dialplan context for use with Gosub

No special syntax is needed when defining the dialplan code that you want to call with Gosub, *unless* you want to return back to where you called Gosub from. In the case of wanting to return, then you should call the Return application.

Here is an example of dialplan we could call with Gosub when we don't wish to return.

```
[my-gosub]
exten => s,1,Verbose("Here we are in a subroutine! Let's listen to some weasels")
 same => n,Playback(tt-weasels)
```

Here is an example where we do wish to return.

```
[my-gosub]
exten => s,1,Verbose("Here we are in a subroutine! Let's listen to some weasels")
 same => n,Playback(tt-weasels)
 same => n,Return()
```

Calling Gosub

Gosub syntax is simple, you only need to specify the priority, and then optionally the context and extension plus any arguments you wish to use.

```
Gosub([[context,]exten,]priority[(arg1[,...][,argN])])
```

Here is an example within Asterisk dialplan.

```
[somecontext]
exten => 7000,1,Verbose("We are going to run a Gosub before Dialing!")
 same => n,Gosub(my-gosub,s,1)
 same => n,Dial(PJSIP/ALICE)
```

Here we are calling the my-gosub context at extension s , priority 1.

Calling Gosub with arguments

If you want to pass information into your Gosub routine then you need to use arguments.

Here is how we call Gosub with an argument. We are substituting the EXTEN channel variable for the first argument field (ARG1).

```
[somecontext]
exten => 7000,1,Verbose("We are going to run a Gosub before Dialing!")
 same => n,Gosub(my-gosub,s,1(${EXTEN}))
 same => n,Dial(PJSIP/ALICE)
```

Below we make use of ARG1 in the Verbose message we print during the subroutine execution.

```
[my-gosub]
exten => s,1,Verbose("Here we are in a subroutine! This subroutine was called from
extension ${ARG1}")
 same => n,Playback(tt-weasels)
 same => n,Return()
```

To use multiple arguments, simply separate them via commas when defining them in the `Gosub` call. Then within the `Gosub` reference them as `ARG1`, `ARG2`, `ARG3`, etc.

Hangup Handlers

Hangup Handlers were added in **Asterisk 11**

Overview

Hangup handlers are subroutines attached to a channel that will execute when that channel hangs up. Unlike the traditional h extension, hangup handlers follow the channel. Thus hangup handlers are always run when a channel is hung up, regardless of where in the dialplan a channel is executing.

Multiple hangup handlers can be attached to a single channel. If multiple hangup handlers are attached to a channel, the hangup handlers will be executed in the order of most recently added first.

NOTES

- Please note that when the hangup handlers execute in relation to the h extension is not defined. They could execute before or after the h extension.
- Call transfers, call pickup, and call parking can result in channels on both sides of a bridge containing hangup handlers.
- Hangup handlers can be attached to any call leg using pre-dial handlers.

WARNINGS

- As hangup handlers are subroutines, they must be terminated with a call to Return.
- Adding a hangup handler in the h extension or during a hangup handler execution is undefined behaviour.
- As always, hangup handlers, like the h extension, need to execute quickly because they are in the hangup sequence path of the call leg. Specific channel driver protocols like ISDN and SIP may not be able to handle excessive delays completing the hangup sequence.

Dialplan Applications and Functions

All manipulation of a channel's hangup handlers are done using the CHANNEL function. All values manipulated for hangup handlers are write-only.

hangup_handler_push

Used to push a hangup handler onto a channel.

```
same => n,Set(CHANNEL(hangup_handler_push)=[[context,]exten,]priority[(arg1[,...][,argN])]);
```

hangup_handler_pop

Used to pop a hangup handler off a channel. Optionally, a replacement hangup handler can be added to the channel.

```
same => n,Set(CHANNEL(hangup_handler_pop)=[[[context,]exten,]priority[(arg1[,...][,argN])]]);
```

hangup_handler_wipe

Remove all hangup handlers on the channel. Optionally, a new hangup handler can be pushed onto the channel.

```
same => n,Set(CHANNEL(hangup_handler_wipe)=[[[context,]exten,]priority[(arg1[,...][,argN])]]);
```

Examples

Adding hangup handlers to a channel

In this example, three hangup handlers are added to a channel: hdlr3, hdlr2, and hdlr1. When the channel is hung up, they will be executed in the order of most recently added first - so hdlr1 will execute first, followed by hdlr2, then hdlr3.

```
; Some dialplan extension
same => n,Set(CHANNEL(hangup_handler_push)=hdlr3,s,1(args));
same => n,Set(CHANNEL(hangup_handler_push)=hdlr2,s,1(args));
same => n,Set(CHANNEL(hangup_handler_push)=hdlr1,s,1(args));
; Continuing in some dialplan extension

[hdlr1]

exten => s,1,Verbose(0, Executed First)
same => n,Return()

[hdlr2]

exten => s,1,Verbose(0, Executed Second)
same => n,Return()

[hdlr3]

exten => s,1,Verbose(0, Executed Third)
same => n,Return()
```

Removing and replacing hangup handlers

In this example, three hangup handlers are added to a channel: hdlr3, hdlr2, and hdlr1. Using the CHANNEL function's **hangup_handler_pop** value, hdlr1 is removed from the stack of hangup handlers. Then, using the **hangup_handler_pop** value again, hdlr2 is replaced with hdlr4. When the channel is hung up, hdlr4 will be executed, followed by hdlr3.

```
; Some dialplan extension
same => n,Set(CHANNEL(hangup_handler_push)=hdlr3,s,1(args));
same => n,Set(CHANNEL(hangup_handler_push)=hdlr2,s,1(args));
same => n,Set(CHANNEL(hangup_handler_push)=hdlr1,s,1(args));
; Remove hdlr1
same => n,Set(CHANNEL(hangup_handler_pop)=)
; Replace hdlr2 with hdlr4
same => n,Set(CHANNEL(hangup_handler_pop)=hdlr4,s,1(args));

; Continuing in some dialplan extension

[hdlr1]

exten => s,1,Verbose(0, Not Executed)
same => n,Return()

[hdlr2]

exten => s,1,Verbose(0, Not Executed)
same => n,Return()

[hdlr3]

exten => s,1,Verbose(0, Executed Second)
same => n,Return()

[hdlr4]

exten => s,1,Verbose(0, Executed First)
same => n,Return()
```

CLI Commands

Single channel

```
core show hanguphandlers <chan>
```

Output

```
Channel       Handler
<chan-name>   <first handler to execute>
              <second handler to execute>
              <third handler to execute>
```

All channels

```
core show hanguphandlers all
```

Output

```
Channel       Handler
<chan1-name>  <first handler to execute>
              <second handler to execute>
              <third handler to execute>
<chan2-name>  <first handler to execute>
<chan3-name>  <first handler to execute>
              <second handler to execute>
```

Macros

Overview

Macros are very similar in function to the Gosub application **which deprecates Macro**. This information is here for historical purposes and you should really use Gosub wherever you would have previously used Macro.

Macro is a dialplan application that facilitates code-reuse within the dialplan. That is, a macro, once defined can be called from almost anywhere else within the dialplan using the Macro application or else via flags and arguments for other applications that allow calling macros.

Other dialplan applications, such as Dial and Queue make use of Macro functionality from within their applications. That means, they allow you to perform actions like calling Macro (or Gosub) on the called party's channel from a Dial, or on a Queue member's channel after they answer. See the Pre-Dial Handlers and Pre-Bridge Handlers sections for more information.

Variables and arguments available within a Macro

The calling extension, context, and priority are stored in **MACRO_EXTEN**, **MACRO_CONTEXT** and **MACRO_PRIORITY** respectively. Arguments become **ARG1**, **ARG2**, etc in the macro context. If you Goto out of the Macro context, the Macro will terminate and control will be returned at the location of the Goto. If **MACRO_OFFSET** is set at termination, Macro will attempt to continue at priority **MACRO_OFFSET + N + 1** if such a step exists, and **N + 1** otherwise.

Defining a dialplan context for use with Macro

Macros look like a typical dialplan context, except for two factors:

- Macros must be named with the 'macro-' prefix.
- Macros must use the 's' extension.

```
[macro-announcement]
exten = s,1,NoOp()
 same = n,Playback(tt-weasels)
```

Calling a Macro

Macro syntax is simple, you only need to specify the priority, and then optionally the context and extension plus any arguments you wish to use.

```
Macro([[context,]exten,]priority[(arg1[,...][,argN])])
```

Here is an example within Asterisk dialplan.

```
[somecontext]
exten = 7000,1,Verbose("We are going to run a Macro before Dialing!")
same = n,Macro(announcement,s,1)
same = n,Dial(PJSIP/ALICE)
```

As you can see we are calling the 'announcement' macro at context 'macro-announcement', extension 's' , priority '1'.

Calling Macro with arguments

Other than the predefined variables mentioned earlier on this page, if you want to pass information into your Macro routine then you need to use arguments.

Here is how we call Macro with an argument. We are substituting the EXTEN channel variable for the first argument field (ARG1).

```
[somecontext]
exten = 7000,1,Verbose("We are going to run a Macro before Dialing!")
same = n,Gosub(announcement,s,1(${EXTEN}))
same = n,Dial(PJSIP/ALICE)
```

Below notice that make use of ARG1 in the Verbose message we print during the subroutine execution.

```
[macro-announcement]
exten = s,1,Verbose("Here we are in a subroutine! This subroutine was called from
extension ${ARG1}")
same = s,n,Playback(tt-weasels)
same = s,n,Return()
```

To use multiple arguments, simply separate them via commas when defining them in the Macro call. Then within the Macro reference them as ARG1, ARG2, ARG3, etc.

Party ID Interception Macros and Routines

Interception routines

> ⚠ As Interception routines are implemented internally using the Gosub application, all routines should end with an explicit call to the Return application.

The interception routines give the administrator an opportunity to alter connected line and redirecting information before the channel driver is given the information. If the routine does not change a value then that is what is going to be passed to the channel driver.

The tag string available in CALLERID, CONNECTEDLINE, and REDIRECTING is useful for the interception routines to provide some information about where the information originally came from.

The 'i' option of the CONNECTEDLINE dialplan function should always be used in the CONNECTED_LINE interception routines. The interception routine always passes the connected line information on to the channel driver when the routine returns. Similarly, the 'i' option of the REDIRECTING dialplan function should always be used in the REDIRECTING interception routines.

> ⓘ Note that Interception routines do not attempt to draw a distinction between caller/callee. As it turned out, it was not a good thing to distinguish since transfers make a mockery of caller/callee.

- ${REDIRECTING_SEND_SUB}
  Subroutine to call before sending a redirecting update to the party.
- ${REDIRECTING_SEND_SUB_ARGS}
  Arguments to pass to ${REDIRECTING_CALLEE_SEND_SUB}.
- ${CONNECTED_LINE_SEND_SUB}
  Subroutine to call before sending a connected line update to the party.
- ${CONNECTED_LINE_SEND_SUB_ARGS}
  Arguments to pass to ${CONNECTED_LINE_SEND_SUB}.

Interception macros

> ⓘ **WARNING**
> Interception macros have been deprecated in Asterisk 11 due to deprecation of Macro. Users of the interception functionality should plan to migrate to Interception routines.

The interception macros give the administrator an opportunity to alter connected line and redirecting information before the channel driver is given the information. If the macro does not change a value then that is what is going to be passed to the channel driver.

The tag string available in CALLERID, CONNECTEDLINE, and REDIRECTING is useful for the interception macros to provide some information about where the information originally came from.

The 'i' option of the CONNECTEDLINE dialplan function should always be used in the CONNECTED_LINE interception macros. The interception macro always passes the connected line information on to the channel driver when the macro exits. Similarly, the 'i' option of the REDIRECTING dialplan function should always be used in the REDIRECTING interception macros.

- ${REDIRECTING_CALLEE_SEND_MACRO}
  Macro to call before sending a redirecting update to the callee. This macro may never be needed since the redirecting updates should only go from the callee to the caller direction. It is available for completeness.
- ${REDIRECTING_CALLEE_SEND_MACRO_ARGS}
  Arguments to pass to ${REDIRECTING_CALLEE_SEND_MACRO}.
- ${REDIRECTING_CALLER_SEND_MACRO}
  Macro to call before sending a redirecting update to the caller.
- ${REDIRECTING_CALLER_SEND_MACRO_ARGS}
  Arguments to pass to ${REDIRECTING_CALLER_SEND_MACRO}.
- ${CONNECTED_LINE_CALLEE_SEND_MACRO}
  Macro to call before sending a connected line update to the callee.
- ${CONNECTED_LINE_CALLEE_SEND_MACRO_ARGS}
  Arguments to pass to ${CONNECTED_LINE_CALLEE_SEND_MACRO}.
- ${CONNECTED_LINE_CALLER_SEND_MACRO}
  Macro to call before sending a connected line update to the caller.

- ${CONNECTED_LINE_CALLER_SEND_MACRO_ARGS}
  Arguments to pass to ${CONNECTED_LINE_CALLER_SEND_MACRO}.

Pre-Bridge Handlers

Overview

Pre-bridge handlers allow you to execute dialplan subroutines on a channel after the call has been initiated and the channels have been created, but before connecting the caller to the callee. Handlers for the Dial and queue applications allow both the older **macro** and the newer **gosub** routines to be executed. These handlers are executed on the **called party** channel, after it is **answered**, but **pre-bridge** before the calling and called party are connected.

If you want to execute routines earlier in the call lifetime then check out the Pre-Dial Handlers section.

Pre-bridge handlers are invoked using flags or arguments for a particular dialplan application. The dialplan application help documentation within Asterisk goes into detail on the various arguments, options and flags, however we will provide some examples below. You should always check the CLI or wiki application docs for any updates.

Dial application

There are two flags for the Dial application, **M** and **U**.

M flag

The M flag allows a macro and arguments to be specified. You must specify the macro name, leaving off the 'macro-' prefix.

```
M(macro[^arg[^...]])
```

The variable MACRO_RESULT can be set with certain options inside the specified macro to determine behavior when the macro finishes. The options are documented in the Dial application documentation.

U flag

The U flag allows a gosub and arguments to be specified. You must remember to call Return inside the gosub.

```
U(x[^arg[^...]])
```

The variable GOSUB_RESULT can be set within certain options inside the specified gosub to determine behavior when the gosub returns. The options are documented in the Dial application documentation.

Queue application

The Queue application, similar to Dial, has two options for handling pre-bridge subroutines. For Queue, both arguments have the same syntax.

```
Queue(queuename[,options[,URL[,announceoverride[,timeout[,AGI[,macro[,gosub[,rule[,positi
on]]]]]]]]])
```

macro and **gosub** can both be populated with the name of a macro or gosub routine to execute on the called party channel as described in the overview.

Examples

Example 1 - Executing a pre-bridge macro handler from Dial

BOB(6002) dials ALICE(6001) and Playback is executed from within the subroutine on the called party's channel after they answer.

Dialplan

```
[from-internal]
exten = 6001,1,Dial(PJSIP/ALICE,30,M(announcement))

[macro-announcement]
exten = s,1,NoOp()
 same = n,Playback(tt-weasels)
 same = n,Hangup()
```

CLI output

```
  -- Executing [6001@from-internal:1] Dial("PJSIP/BOB-00000014", "PJSIP/ALICE,30,M(announcement)") in new stack
      -- Called PJSIP/ALICE
      -- PJSIP/ALICE-00000015 is ringing
      -- PJSIP/ALICE-00000015 answered PJSIP/BOB-00000014
      -- Executing [s@macro-announcement:1] NoOp("PJSIP/ALICE-00000015", "") in new stack
      -- Executing [s@macro-announcement:2] Playback("PJSIP/ALICE-00000015", "tt-weasels") in new stack
      -- <PJSIP/ALICE-00000015> Playing 'tt-weasels.gsm' (language 'en')
      -- Channel PJSIP/BOB-00000014 joined 'simple_bridge' basic-bridge <612c2313-98bf-48ce-89b1-d530b06e44d7>
      -- Channel PJSIP/ALICE-00000015 joined 'simple_bridge' basic-bridge <612c2313-98bf-48ce-89b1-d530b06e44d7>
      -- Channel PJSIP/BOB-00000014 left 'native_rtp' basic-bridge <612c2313-98bf-48ce-89b1-d530b06e44d7>
      -- Channel PJSIP/ALICE-00000015 left 'native_rtp' basic-bridge <612c2313-98bf-48ce-89b1-d530b06e44d7>
    == Spawn extension (from-internal, 6001, 1) exited non-zero on 'PJSIP/BOB-00000014'
```

Example 2 - Executing a pre-bridge gosub handler from Dial

ALICE(6001) dials BOB(6002) and Playback is executed from within the subroutine on the called party's channel after they answer. Notice that since this subroutine is a gosub, we need to call Return.

Dialplan

```
[from-internal]
exten = 6002,1,Dial(PJSIP/BOB,30,U(sub-announcement))

[sub-announcement]
exten = s,1,NoOp()
 same = n,Playback(tt-weasels)
 same = n,Return()
```

CLI output

```
  -- Executing [6002@from-internal:1] Dial("PJSIP/ALICE-00000016", "PJSIP/BOB,30,U(sub-announcement)") in new stack
      -- Called PJSIP/BOB
      -- PJSIP/BOB-00000017 is ringing
      -- PJSIP/BOB-00000017 answered PJSIP/ALICE-00000016
      -- PJSIP/BOB-00000017 Internal Gosub(sub-announcement,s,1) start
      -- Executing [s@sub-announcement:1] NoOp("PJSIP/BOB-00000017", "") in new stack
      -- Executing [s@sub-announcement:2] Playback("PJSIP/BOB-00000017", "tt-weasels") in new stack
      -- <PJSIP/BOB-00000017> Playing 'tt-weasels.gsm' (language 'en')
      -- Executing [s@sub-announcement:3] Return("PJSIP/BOB-00000017", "") in new stack
    == Spawn extension (from-internal, , 1) exited non-zero on 'PJSIP/BOB-00000017'
      -- PJSIP/BOB-00000017 Internal Gosub(sub-announcement,s,1) complete GOSUB_RETVAL=
      -- Channel PJSIP/ALICE-00000016 joined 'simple_bridge' basic-bridge <16e76a40-4a24-441d-a2b2-5c9ddfb21d7a>
      -- Channel PJSIP/BOB-00000017 joined 'simple_bridge' basic-bridge <16e76a40-4a24-441d-a2b2-5c9ddfb21d7a>
      -- Channel PJSIP/BOB-00000017 left 'native_rtp' basic-bridge <16e76a40-4a24-441d-a2b2-5c9ddfb21d7a>
      -- Channel PJSIP/ALICE-00000016 left 'native_rtp' basic-bridge <16e76a40-4a24-441d-a2b2-5c9ddfb21d7a>
    == Spawn extension (from-internal, 6002, 1) exited non-zero on 'PJSIP/ALICE-00000016'
```

Example 3 - Executing a pre-bridge gosub handler from Queue

ALICE(6001) dials Queue 'sales' where BOB(6002) is a member. Once BOB answers the queue call, the Playback is executed from within the gosub.

Dialplan

```
[sub-announcement]
exten = s,1,NoOp()
 same = n,Playback(tt-weasels)
 same = n,Return()

[queues]
exten => 7002,1,Verbose(2,${CALLERID(all)} entering the sales queue)
same => n,Queue(sales,,,,,,,sub-announcement)
same => n,Hangup()
```

CLI output

```
  -- Executing [7002@from-internal:1] Verbose("PJSIP/ALICE-00000009", "2,"Alice" <ALICE> entering the sales queue") in new stack
    == "Alice" <ALICE> entering the sales queue
      -- Executing [7002@from-internal:2] Queue("PJSIP/ALICE-00000009", "sales,,,,,,,sub-announcement") in new stack
      -- Started music on hold, class 'default', on channel 'PJSIP/ALICE-00000009'
      -- Called PJSIP/BOB
      -- PJSIP/BOB-0000000a is ringing
         > 0x7f74d4039840 -- Probation passed - setting RTP source address to 10.24.18.138:4042
      -- PJSIP/BOB-0000000a answered PJSIP/ALICE-00000009
      -- Stopped music on hold on PJSIP/ALICE-00000009
      -- PJSIP/BOB-0000000a Internal Gosub(sub-announcement,s,1) start
      -- Executing [s@sub-announcement:1] NoOp("PJSIP/BOB-0000000a", "") in new stack
      -- Executing [s@sub-announcement:2] Playback("PJSIP/BOB-0000000a", "tt-weasels") in new stack
      -- <PJSIP/BOB-0000000a> Playing 'tt-weasels.gsm' (language 'en')
      -- Executing [s@sub-announcement:3] Return("PJSIP/BOB-0000000a", "") in new stack
    == Spawn extension (from-internal, 7002, 1) exited non-zero on 'PJSIP/BOB-0000000a'
      -- PJSIP/BOB-0000000a Internal Gosub(sub-announcement,s,1) complete GOSUB_RETVAL=
      -- Channel PJSIP/ALICE-00000009 joined 'simple_bridge' basic-bridge <cbc54ed6-1f51-4b10-be99-4994f52d851f>
      -- Channel PJSIP/BOB-0000000a joined 'simple_bridge' basic-bridge <cbc54ed6-1f51-4b10-be99-4994f52d851f>
         > Bridge cbc54ed6-1f51-4b10-be99-4994f52d851f: switching from simple_bridge technology to native_rtp
         > Remotely bridged 'PJSIP/BOB-0000000a' and 'PJSIP/ALICE-00000009' - media will flow directly between them
         > Remotely bridged 'PJSIP/BOB-0000000a' and 'PJSIP/ALICE-00000009' - media will flow directly between them
         > 0x7f74d400c620 -- Probation passed - setting RTP source address to 10.24.18.16:4040
      -- Channel PJSIP/BOB-0000000a left 'native_rtp' basic-bridge <cbc54ed6-1f51-4b10-be99-4994f52d851f>
      -- Channel PJSIP/ALICE-00000009 left 'native_rtp' basic-bridge <cbc54ed6-1f51-4b10-be99-4994f52d851f>
    == Spawn extension (from-internal, 7002, 2) exited non-zero on 'PJSIP/ALICE-00000009'
```

Example 4 - Executing a pre-bridge macro handler from Queue

BOB(6002) calls the queue 'support' where ALICE(6001) is a member. Once ALICE answers the queue call, the Playback is executed from within the macro.

Dialplan

```
[macro-announcement]
exten = s,1,NoOp()
 same = n,Playback(tt-weasels)

[queues]
exten => 7001,1,Verbose(2,${CALLERID(all)} entering the support queue)
same => n,Queue(support,,,,,,announcement)
same => n,Hangup()
```

CLI output

```
  -- Executing [7001@from-internal:1] Verbose("PJSIP/BOB-00000004", "2,"Bob" <BOB> entering the support queue") in new stack
    == "Bob" <BOB> entering the support queue
      -- Executing [7001@from-internal:2] Queue("PJSIP/BOB-00000004", "support,,,,,,announcement") in new stack
      -- Started music on hold, class 'default', on channel 'PJSIP/BOB-00000004'
      -- Called PJSIP/ALICE
      -- PJSIP/ALICE-00000005 is ringing
         > 0x7f8450039d40 -- Probation passed - setting RTP source address to 10.24.18.16:4048
      -- PJSIP/ALICE-00000005 answered PJSIP/BOB-00000004
      -- Stopped music on hold on PJSIP/BOB-00000004
      -- Executing [s@macro-announcement:1] NoOp("PJSIP/ALICE-00000005", "") in new stack
      -- Executing [s@macro-announcement:2] Playback("PJSIP/ALICE-00000005", "tt-weasels") in new stack
      -- <PJSIP/ALICE-00000005> Playing 'tt-weasels.gsm' (language 'en')
      -- Channel PJSIP/BOB-00000004 joined 'simple_bridge' basic-bridge <8283212f-b12d-4571-9653-0c8484e88980>
      -- Channel PJSIP/ALICE-00000005 joined 'simple_bridge' basic-bridge <8283212f-b12d-4571-9653-0c8484e88980>
         > Bridge 8283212f-b12d-4571-9653-0c8484e88980: switching from simple_bridge technology to native_rtp
         > Remotely bridged 'PJSIP/ALICE-00000005' and 'PJSIP/BOB-00000004' - media will flow directly between them
         > Remotely bridged 'PJSIP/ALICE-00000005' and 'PJSIP/BOB-00000004' - media will flow directly between them
         > 0x7f84500145d0 -- Probation passed - setting RTP source address to 10.24.18.138:4050
      -- Channel PJSIP/ALICE-00000005 left 'native_rtp' basic-bridge <8283212f-b12d-4571-9653-0c8484e88980>
      -- Channel PJSIP/BOB-00000004 left 'native_rtp' basic-bridge <8283212f-b12d-4571-9653-0c8484e88980>
    == Spawn extension (from-internal, 7001, 2) exited non-zero on 'PJSIP/BOB-00000004'
```

Pre-Dial Handlers

Pre-Dial Handlers were added in **Asterisk 11**

Overview

Pre-dial handlers allow you to execute a dialplan subroutine on a channel before a call is placed but after the application performing a dial action is invoked. This means that the handlers are executed after the creation of the caller/callee channels, but before any actions have been taken to actually dial the callee channels. You can execute a dialplan subroutine on the caller channel and on each callee channel dialed.

There are two ways in which a pre-dial handler can be invoked:

- The '**B**' option in an application executes a dialplan subroutine on the caller channel before any callee channels are created.
- The '**b**' option in an application executes a dialplan subroutine on each callee channel after it is created but before the call is placed to the end-device.

Pre-dial handlers are supported in the Dial application and the FollowMe application.

WARNINGS

- As pre-dial handlers are implemented using Gosub subroutines, they must be terminated with a call to Return.
- Taking actions in pre-dial handlers that would put the caller/callee channels into other applications will result in undefined behaviour. Pre-dial handlers should be short routines that do not impact the state that the dialing application assumes the channel will be in.

Syntax

For Dial or FollowMe, handlers are invoked using similar nomenclature as other options (such as **M** or **U** in Dial) that cause some portion of the dialplan to execute.

```
b([[context^]exten^]priority[(arg1[^...][^argN])])
B([[context^]exten^]priority[(arg1[^...][^argN])])
```

If context or exten are not supplied then the current values from the caller channel are used.

Examples

The examples illustrated below use the following channels:

- *SIP/foo* is calling either *SIP/bar*, *SIP/baz*, or both
- *SIP/foo* is the caller
- *SIP/bar* is a callee
- *SIP/baz* is another callee

Example 1 - Executing a pre-dial handler on the caller channel

```
[default]

exten => s,1,NoOp()
same => n,Dial(SIP/bar,,B(default^caller_handler^1))
same => n,Hangup()

exten => caller_handler,1,NoOp()
same => n,Verbose(0, In caller pre-dial handler!)
same => n,Return()
```

Example 1 CLI Output

```
<SIP/foo-123> Dial(SIP/bar,,B(default^caller_handler^1))
<SIP/foo-123> Executing default,caller_handler,1
<SIP/foo-123> In caller pre-dial handler!
<SIP/foo-123> calling SIP/bar-124
```

Example 2 - Executing a pre-dial handler on a callee channel

```
[default]

exten => s,1,NoOp()
same => n,Dial(SIP/bar,,b(default^callee_handler^1))
same => n,Hangup()

exten => callee_handler,1,NoOp()
same => n,Verbose(0, In callee pre-dial handler!)
same => n,Return()
```

Example 2 CLI Output

```
<SIP/foo-123> Dial(SIP/bar,,b(default^callee_handler^1))
<SIP/bar-124> Executing default,callee_handler,1
<SIP/bar-124> In callee pre-dial handler!
<SIP/foo-123> calling SIP/bar-124
```

Example 3 - Executing a pre-dial handler on multiple callee channels

```
[default]

exten => s,1,NoOp()
same => n,Dial(SIP/bar&SIP/baz,,b(default^callee_handler^1))
same => n,Hangup()

exten => callee_handler,1,NoOp()
same => n,Verbose(0, In callee pre-dial handler!)
same => n,Return()
```

Example 3 CLI Output

```
<SIP/foo-123> Dial(SIP/bar&SIP/baz,,b(default^callee_handler^1))
<SIP/bar-124> Executing default,callee_handler,1
<SIP/bar-124> In callee pre-dial handler!
<SIP/baz-125> Executing default,callee_handler,1
<SIP/baz-125> In callee pre-dial handler!
<SIP/foo-123> calling SIP/bar-124
<SIP/foo-123> calling SIP/baz-125
```

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.

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.

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 [ ] constructs (recursively, bottom- up), no is ever present when the contents of a [] is evaluated. Thus, tokens are usually already merged at evaluation time. But, in AEL, various exprs are evaluated raw, 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) 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.

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.

Expr2 Built-in 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!

- COS(x) x is in radians. Results vary from -1 to 1.
- SIN(x) x is in radians. Results vary from -1 to 1.
- TAN(x) x is in radians.
- ACOS(x) x should be a value between -1 and 1.
- ASIN(x) x should be a value between -1 and 1.
- ATAN(x) returns the arc tangent in radians; between -PI/2 and PI/2.
- 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.
- POW(x,y) returns the value of x raised to the power of y.
- SQRT(x) returns the square root of x.
- FLOOR(x) rounds x down to the nearest integer.
- CEIL(x) rounds x up to the nearest integer.
- ROUND(x) rounds x to the nearest integer, but round halfway cases away from zero.
- RINT(x) rounds x to the nearest integer, rounding halfway cases to the nearest even integer.
- TRUNC(x) rounds x to the nearest integer not larger in absolute value.
- 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.
- EXP(x) returns e to the x power.
- EXP2(x) returns 2 to the x power.
- LOG(x) returns the natural logarithm of x.
- LOG2(x) returns the base 2 log of x.
- LOG10(x) returns the base 10 log of x.

Expressions Examples

Expression	Result	Note
"One Thousand Five Hundred" =~ "(T[Expressions Examples^])"	Thousand	
"One Thousand Five Hundred" =~ "T[Expressions Examples^]"	8	
"One Thousand Five Hundred" : "T[Expressions Examples^]"	0	
"8015551212" : "(...)"	801	
"3075551212":"...(...)"	555	
! "One Thousand Five Hundred" =~ "T[Expressions Examples^]"	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[Expressions Examples^]+")	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	6	Because of operator precedence; the division is done first, then the addition
2+8/2	6	Spaces aren't necessary
(2+8)/2	5	
(3+8)/2	5.5	
TRUNC((3+8)/2)	5	
FLOOR(2.5)	2	
FLOOR(-2.5)	-3	
CEIL(2.5)	3	
CEIL(-2.5)	-2	
ROUND(2.5)	3	
ROUND(3.5)	4	
ROUND(-2.5)	-3	
RINT(2.5)	2	
RINT(3.5)	4	
RINT(-2.5)	-2	
RINT(-3.5)	-4	
TRUNC(2.5)	2	
TRUNC(3.5)	3	
TRUNC(-3.5)	-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.

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.

Expression Parsing 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-liar,s,1:s,7)
```

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_M
INUS or TOK_LP or TOKEN; Input:
 "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.

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.

Warnings about Expressions

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.

Expression Parser Incompatibilities

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, ${VAR1} : (WhoWhat)+ may have have worked before, but unless you wrap the pattern in double quotes now, look out for trouble! This is better: "${VAR1}" : "(WhoWhat*)+" 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.

Expression 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.

Conditional Applications

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)
```

Privilege Escalations with Dialplan Functions

Dialplan functions within Asterisk are incredibly powerful, which is wonderful for building applications using Asterisk. Dialplan functions can be 'read' or 'written'. But during the read or write execution, certain diaplan functions do much more. For example, reading the SHELL() function can execute arbitrary commands on the system Asterisk is running on. Writing to the FILE() function can change any file that Asterisk has write access to.

From the context of executing the dialplan defined in `extensions.conf`, this is not a problem. From other contexts, however, it could be a problem.

Channel variables can be get or set via external mechanisms (like AMI or ARI), during which time dialplan functions are evaluated. These external protocols have permission levels associated with them, so the fact that executing a read on a certain function could effect a change on the system results in a privilege escalation.

In order to avoid this security issue, Asterisk can now inhibit the execution of privilege escalating functions from external protocols. These functions will continue to execute normally when invoked from the dialplan. For legacy configurations where the less secure behavior is desired, a new flag called `live_dangerously` has been added to `asterisk.conf`. When set to `yes`, Asterisk will allow privilege escalating functions to execute, even from external protocols.

For Asterisk 11 and earlier, in order to maintain backward compatibility, `live_dangerously` defaults to `yes`. In Asterisk 12, that default was changed to `no`.

Asterisk Extension Language (AEL)

Top-level page for all things AEL

Introduction to AEL

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:

1. The first and most obvious is the AEL syntax itself. A BNF is provided near the end of this document.
2. 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.
3. 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.
4. 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.

AEL and 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.

AEL about 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.

AEL about 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.

AEL about 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.

AEL about 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!

Getting Started with AEL

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
```

AEL 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.

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:

1. -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.
2. -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.

General Notes about AEL 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;
}
```

AEL 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

AEL 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.

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

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

AEL Example Usages

Example usages of AEL

AEL 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.

AEL 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);
}
```

AEL 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!

AEL 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;
    }
}
```

AEL including other files

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).

AEL 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};
    }
}
```

AEL 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;
}
```

AEL 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});
}
```

AEL 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});
}
```

AEL 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 (14:00-23:59|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!);
        }
    }
}
```

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

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

⚠ 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.

⚠ 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

⚠ 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.

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

⚠ 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.

AEL 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 }
}
```

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...

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

AEL 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 commas.

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

⚠ **Dialplan Macros Channel Variables**
Due to AEL macro reimplementation the following variables will **not** be set:

- ${MACRO_EXTEN}
- ${MACRO_CONTEXT}
- ${MACRO_PRIORITY}
- ${MACRO_OFFSET}

AEL 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.

AEL Break, Continue, and Return

Three keywords:

1. break
2. continue
3. 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.

AEL 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);
}
```

AEL 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.

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 rangedays of weekdays of monthmonths) 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.

AEL 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.

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.

Lua Dialplan Configuration

Asterisk supports the ability to write dialplan instructions in the Lua programming language. This method can be used as an alternative to or in combination with extensions.conf and/or AEL. PBX lua allows users to use the full power of lua to develop telephony applications using Asterisk. Lua dialplan configuration is done in the `extensions.lua` file.

> **Dependencies**
> To use pbx_lua, the lua development libraries must be installed before Asterisk is configured and built. You can get these libraries directly from http://lua.org, but it is easier to install them using your distribution's package management tool. The package is probably named liblua5.1-dev, liblua-dev, or lua-devel depending on your linux distribution.

PBX Lua Basics

The `extensions.lua` file is used to configure PBX lua and is a lua script (as opposed to being a standard asterisk configuration file). Any thing that is proper lua code is allowed in this file. Asterisk expects to find a global table named '`extensions`' when the file is loaded. This table can be generated however you wish. The simplest way is to define all of the extensions in line, but for more complex dialplans alternative methods may be necessary.

Each extension is a lua function that is executed when a channel lands on that extension. The extension function is passed the current context and extension as the first two arguments. These can be safely ignored if desired. There are no priorities (each extension function is treated as priority 1 by the rest of Asterisk). Patterns are allowed just as in `extensions.conf` and the matching order is identical.

extensions.lua

```
extensions = {
   default = {
      ["100"] = function(context, extension)
         app.playback("please-hold")
         app.dial("SIP/100", 60)
      end;

      ["101"] = function(c, e)
         app.dial("SIP/101", 60)
      end;
}
```

The `extensions.lua` file can be reloaded by reloading the pbx_lua module.

```
*CLI> module reload pbx_lua
```

If there are errors in the file, the errors will be reported and the existing extensions.lua file will remain in use. Channels that existed before the reload command was issued will also continue to use the existing extensions.lua file.

> Runtime errors are logged and the channel on which the error occurred is hung up.

Dialplan to Lua Reference

Below is a quick reference that can be used to translate traditional `extensions.conf` dialplan concepts to their analog in `extensions.lua`.

Extension Patterns

Extension pattern matching syntax on logic works the same for `extensions.conf` and `extensions.lua`.

extensions.conf

```
[users]
exten =>
_1XX,1,Dial(SIP/${EXTEN})

exten =>
_2XX,1,Voicemail(${EXTEN:1})
```

extensions.lua

```
extensions = {}
extensions.users = {}

extensions.users["_1XX"] =
function(c, e)
   app.dial("SIP/" .. e)
end

extensions.users["_2XX"] =
function(c, e)
  app.voicemail("1" .. e:sub(2))
end
```

Context Includes

extensions.conf

```
[users]
exten => 100,1,Noop
exten => 100,n,Dial("SIP/100")

[demo]
exten => s,1,Noop
exten =>
s,n,Playback(demo-congrats)

[default]
include => demo
include => users
```

extensions.lua

```
extensions = {
   users = {
      [100] = function()
         app.dial("SIP/100")
      end;
   };

   demo = {
      ["s"] = function()

app.playback(demo-congrats)
      end;
   };

   default = {
      include = {"demo",
"users"};
   };
}
```

Loops

extensions.conf

```
exten => 100,1,Noop
exten => 100,n,Set(i=0)
exten => 100,n,While($[i < 10])
exten => 100,n,Verbose(i = ${i})
exten => 100,n,EndWhile
```

extensions.lua

```
i = 0
while i < 10 do
  app.verbose("i = " .. i)
end
```

Variables

extensions.conf

```
exten =>
100,1,Set(my_variable=my_value)
exten =>
100,n,Verbose(my_variable =
${my_variable})
```

extensions.lua

```
channel.my_variable = "my_value"
app.verbose("my_variable = " ..
channel.my_variable:get())
```

Applications

extensions.conf

```
exten =>
100,1,Dial("SIP/100",,m)
```

extensions.lua

```
app.dial("SIP/100", nil, "m")
```

Macros/GoSub

Macros can be defined in pbx_lua by naming a context 'macro-' just as in `extensions.conf`, but generally where you would use macros or gosub in `extensions.conf` you would simply use a function in lua.*

extensions.conf

```
[macro-dial]
exten => s,1,Noop
exten => s,n,Dial(${ARG1})

[default]
exten =>
100,1,Macro(dial,SIP/100)
```

extensions.lua

```
extensions = {}
extensions.default = {}

function dial(resource)
   app.dial(resource)
end

extensions.default[100] =
function()
   dial("SIP/100")
end
```

Goto

While `Goto` is an extenstions.conf staple, it should generally be avoided in pbx_lua in favor of functions.

extensions.conf

```
[default]
exten => 100,1,Goto(102,1)

exten =>
102,1,Playback("demo-thanks")
exten => 102,n,Hangup
```

extensions.lua

```
extensions = {}
extensions.default = {}

function do_hangup()
   app.playback("demo-thanks")
   app.hangup()
end

extensions.default[100] =
function()
   do_hangup()
end
```

The `app.goto()` function will not work as expected in pbx_lua in Asterisk 1.8. If you must use `app.goto()` you must manually return control back to asterisk using `return` from the dialplan extension function, otherwise execution will continue after the call to `app.goto()`. Calls to `app.goto()` should work as expected in Asterisk 10 but still should not be necessary in most cases.

In Asterisk 1.8, use return

```
function extension_function(c, e)
   return app.goto("default", "100", 1)

   -- without that 'return' the rest of the function would execute normally
   app.verbose("Did you forget to use 'return'?")
end
```

Interacting with Asterisk from Lua (apps, variables, and functions)

Interaction with is done through a series of predefined objects provided by pbx_lua. The `app` table is used to access dialplan applications. Any asterisk application can be accessed and executed as if it were a function attached to the `app` table. Dialplan variables and functions are accessed and executed via the `channel` table.

⚠ **Naming Conflicts Between Lua and Asterisk**
Asterisk applications, variables or functions whose names conflict with Lua reserved words or contain special characters must be referenced using the `[]` operator. For example, Lua 5.2 introduced the `goto` control statement which conflicts with the Asterisk `goto` dialplan application. So...

> The following will cause pbx_lua.so to fail to load with Lua 5.2 or later because `goto` is a reserved word.
>
> ```
> app.goto("default", 1000, 1)
> ```

> The following will work with all Lua versions...
>
> ```
> app["goto"]("default", 1000, 1)
> ```

Dialplan Applications

extensions.lua

```
app.playback("please-hold")
app.dial("SIP/100", nil, "m")
```

Any dialplan application can be executed using the `app` table. Application names are case insensitive. Arguments are passed to dialplan applications just as arguments are passed to functions in lua. String arguments must be quoted as they are lua strings. Empty arguments may be passed as `nil` or as empty strings.

Channel Variables

Set a Variable

```
channel.my_variable = "my_value"
```

After this the channel variable `${my_variable}` contains the value "my_value".

Read a Variable

```
value = channel.my_variable:get()
```

Any channel variable can be read and set using the `channel` table. Local and global lua variables can be used as they normally would and are completely unrelated to channel variables.

> The following construct will NOT work.
>
> ```
> value = channel.my_variable -- does not work as expected (value:get() could be used
> to get the value after this line)
> ```

If the variable name is an Lua reserved word or contains characters that Lua considers special use the `[]` operator to access them.

```
channel["my_variable"] = "my_value"
value = channel["my_variable"]:get()
```

Dialplan Functions

Write a Dialplan Function

```
channel.FAXOPT("modems"):set("v17,v27,v29")
```

Read a Dialplan Function

```
value = channel.FAXOPT("modems"):get()
```

Note the use of the `:` operator with the `get()` and `set()` methods.

If the function name is an Lua reserved word or contains characters that Lua considers special use the `[]` operator to access them.

```
channel["FAXOPT(modems)"] = "v17,v27,v29"
value = channel["FAXOPT(modems)"]:get()
```

The following constructs will NOT work.

```
channel.FAXOPT("modems") = "v17,v27,v29" -- syntax error
value = channel.FAXOPT("modems")         -- does not work as expected (value:get()
could be used to get the value after this line)
```

Dialplan function names are case sensitive.

Lua Dialplan Tips and Tricks

Long Running Operations (Autoservcie)

Before starting long running operations, an autoservice should be started using the `autoservice_start()` function. An autoservice will ensure that the user hears a continuous stream of audio while your lua code works in the background. This autoservice will automatically be stopped before executing applications and dialplan functions and will be restarted afterwards. The autoservice can be stopped using autoservice_stop() and the autoservice_status() function will return `true` if an autoservice is currently running.

```
app.startmusiconhold()

autoservice_start()
do_expensive_db_query()
autoservice_stop()

app.stopmusiconhold()
```

> ⓘ In Asterisk 10 an autoservice is automatically started for you by default.

Defining Extensions Dynamically

Since extensions are functions in pbx_lua, any function can be used, including closures. A function can be defined that returns extension functions and used to populate the extensions table.

extensions.lua

```
extensions = {}
extensions.default = {}

function sip_exten(e)
   return function()
      app.dial("SIP/" .. e)
   end
end

extensions.default[100] = sip_exten(100)
extensions.default[101] = sip_exten(101)
```

Creating Custom Aliases for Built-in Constructs

If you don't like the `app` table being named 'app' or if you think typing 'channel' to access the `channel` table is too much work, you can rename them.

I prefer less typing

```
function my_exten(context, extensions)
   c = channel
   a = app

   c.my_variable = "my new channel variable"
   a.dial("SIP/100")
end
```

Re-purposing The `print` Function

Lua has a built in "print" function that outputs things to stdout, but for Asterisk, we would rather have the output go in the verbose log. To do so, we could rewrite the `print` function as follows.

```
function print(...)
   local msg = ""
   for i=1,select('#', ...) do
      if i == 1 then
         msg = msg .. tostring(select(i, ...))
      else
         msg = msg .. "\t" .. tostring(select(i, ...))
      end
   end

   app.verbose(msg)
end
```

Splitting Configuration into Multiple Files

The `require` method can be used to load lua modules located in LUA_PATH. The `dofile` method can be used to include any file by path name.

Using External Modules

Lua modules can be loaded using the standard `require` lua method. Some of the functionality provided by various lua modules is already included in Asterisk (e.g. func_odbc provides what LuaSQL provides). It is generally better to use code built-in to Asterisk over external lua modules. Specifically, the func_odbc module uses a connection pool to provide database resources, where as with LuaSQL each channel would have to make a new connection to the database on its own.

Compile extensions.lua

The `luac` program can be used to compile your `extensions.lua` file into lua bytecode. This will slightly increase performance as pbx_lua will no longer need to parse `extensions.lua` on load. The `luac` compiler will also detect and report any syntax errors. To use `luac`, rename your `extensions.lua` file and then run `luac` as follows.

Assume you name your extensions.lua file extensions.lua.lua

```
luac -o extensions.lua extensions.lua.lua
```

The pbx_lua module automatically knows the difference between a lua text file and a lua bytecode file.

Lua Dialplan Hints

In Asterisk 10 dialplan hints can be specified in `extensions.lua` in a manner similar to the way extensions are specified.

extensions.lua

```
hints = {
   default = {
      ["100"] = "SIP/100";
   };

   office = {
      ["500"] = "SIP/500";
   };

   home = {
      ["200"] = "SIP/200";
      ["201"] = "SIP/201";
   };
}
```

Lua Dialplan Examples

Some example `extensions.lua` files can be found below. They demonstrate various ways to organize extensions.

Less Clutter

Instead of defining every extension inline, you can use this method to create a neater `extensions.lua` file. Since the extensions table and each context are both normal lua tables, you can treat them as such and build them piece by piece.

extensions.lua

```
-- this function serves as an extension function directly
function call_user(c, user)
   app.dial("SIP/" .. user, 60)
end

-- this function returns an extension function
function call_sales_queue(queue)
  return function(c, e)
      app.queue(queue)
   end
end

e = {}

e.default = {}
e.default.include = {"users", "sales"}

e.users = {}
e.users["100"] = call_user
e.users["101"] = call_user

e.sales = {}
e.sales["5000"] = call_sales_queue("sales1")
e.sales["6000"] = call_sales_queue("sales2")

extensions = e
```

Less Clutter v2

In this example, we use a fancy function to register extensions.

extensions.lua

```
function register(context, extension, func)
   if not extensions then
      extensions = {}
   end

   if not extensions[context] then
      extensions[context] = {}
   end

   extensions[context][extension] = func
end

function include(context, included_context)
   if not extensions then
      extensions = {}
   end

   if not extensions[context] then
      extensions[context] = {}
   end

   if not extensions[context].include then
      extensions[context].include = {}
   end

   table.insert(extensions[context].include, included_context)
end

-- this function serves as an extension function directly
function call_user(c, user)
   app.dial("SIP/" .. user, 60)
end

-- this function returns an extension function
function call_sales_queue(queue)
  return function(c, e)
      app.queue(queue)
   end
end

include("default", "users")
include("default", "sales")

register("users", "100", call_user)
register("users", "101", call_user)

register("sales", "5000", call_sales_queue("sales1"))
register("sales", "6000", call_sales_queue("sales2"))
register("sales", "7000", function()
   app.queue("sales3")
end)
```

Advanced pbx_lua Topics

Behind the scenes, a number of things happen to make the integration of lua into Asterisk as seamless as possible. Some details of how this integration works can be found below.

extensions.lua Load Process

The `extensions.lua` file is loaded into memory once when the pbx_lua module is loaded or reloaded. The file is then read from memory and executed once for each channel that looks up or executes a lua based extension. Since the file is executed once for each channel, it may not be wise to do things like connect to external services directly from the main script or build your extensions table from a webservice or database.

This is probably a bad idea.

```
-- my fancy extensions.lua

extensions = {}
extensions.default = {}

-- might be a bad idea, this will run each time a channel is created
data = query_webservice_for_extensions_list("site1")

for _, e in ipairs(data) do
   extensions.default[e.exten] = function()
      app.dial("SIP/" .. e.sip_peer, e.dial_timeout)
   end
end
```

The extensions Table

The `extensions` table is a standard lua table and can be defined however you like. The pbx_lua module loads and sorts the table when it is needed. The keys in the table are context names and each value is another lua table containing extensions. Each key in the context table is an extension name and each value is an extension function.

```
extensions = {
   context_table = {
      extension1 = function()
      end;
      extension2 = function()
      end;
   };
}
```

Where did the priorities go?

There are no priorities. Asterisk uses priorities to define the order in which dialplan operations occur. The pbx_lua module uses functions to define extensions and execution occurs within the lua interpreter, priorities don't make sense in this context. To Asterisk, each pbx_lua extension appears as an extension with one priority. Lua extensions can be referenced using the context name, extension, and priority 1, e.g. `Goto(default,1234,1)`. You would only reference extensions this way from outside of pbx_lua (i.e. from `extensions.conf` or `extensions.ael`). From with in pbx_lua you can just execute that extension's function.

```
extensions.default["1234"]("default", "1234")
```

Lua Script Lifetime

The same lua state is used for the lifetime of the Asterisk channel it is running on, so effectively, the script has the lifetime of the channel. This means you can set global variables in the lua state and retrieve them later from a different extension if necessary.

Apps, Functions, and Variables

Details on accessing dialplan applications and functions and channel variables can be found in the Interacting with Asterisk from Lua (apps, variables, and *functions) page.*

When accessing a dialplan application or function or a channel variable, a placeholder object is generated that provides the `:get()` and `:set()` methods.

channel variable: var is the placeholder object

```
var = channel.my_variable
var:set("my value")
value = var:get("my value")
```

dialplan function: fax_modems is the placeholder object

```
fax_modems = channel.FAXOPT("module")

-- the function arguments are stored in the placeholder

fax_modems:set("v17")
value = fax_modems:get()
```

dialplan application: dial is the placeholder object

```
dial = app.dial

-- the only thing we can do with it is execute it
dial("SIP/100")
```

There is a small cost in creating the placeholder objects so storing frequently used placeholder objects can be used as a micro optimization. This should never be necessary though and only provides benefits if you are running micro benchmarks.

Features

The Asterisk core provides a set of features that once enabled can be activated through DTMF codes (also known as feature codes).

Features are configured in features.conf and most require additional configuration via arguments or options to applications that invoke channel creation.

> ✓ Versions of Asterisk older than 12 included parking configuration inside features.conf. In Asterisk 12 parking configuration was moved out into res_parking.conf.

The core features discussed in this section are:

- Feature Code Call Transfers
  - Blind transfers
  - Attended transfers and variations.
- One-Touch Features
  - This includes instructions for call recording, disconnect and quick parking.
- Call Pickup
  - Feature code call pickup as well as dialplan application-based call pickup.
- Built-in Dynamic Features
  - How to use a couple of functions to set built-in feature codes on a per-channel basis.
- Custom Dynamic Features
  - How to define custom features and set them on a per-channel basis using a channel variable.
- Call Parking
  - Instructions for how to implement parking lots (with examples).

The only features discussed in this section are those that have some relation to features.conf. Features in a broader sense - that is features that your application built with Asterisk may have - are implemented through usage of Applications, Functions and Interfaces or Dialplan.

Feature Code Call Transfers

Overview of Feature Code Call Transfers

A call transfer is when one party of a call directs Asterisk to connect the other party to a new location on the system.

Transfer types supported by the Asterisk core:

- Blind transfer
- Attended transfer
  - Variations on attended transfer behavior

Transfer features provided by the Asterisk core are configured in features.conf and accessed with feature codes.

Channel driver technologies such as chan_sip and chan_pjsip have native capability for various transfer types. That native transfer functionality is independent of this core transfer functionality. The core feature code transfer functionality is channel agnostic.

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Blind transfer

A blind or unsupervised transfer is where the initiating party is blind to what is happening after initiating the transfer. They are removed from the process as soon as they initiate the transfer. It is a sort of "fire and forget" transfer.

Attended transfer

An attended or supervised transfer happens when one party transfers another party to a new location by first dialing the transfer destination and only completing the transfer when ready. The initiating party is attending or supervising the transfer process by contacting the destination before completing the transfer. This is helpful if the transfer initiator wants to make sure someone answers or is ready at the destination.

Configuring Transfer Features

There are three primary requirements for the use of core transfer functionality.

- The transfer type must be enabled and assigned a DTMF digit string in features.conf or per channel - see (((Dynamic DTMF Features)))
- The channel must allow the type of transfer attempted. This can be configured via the Application invoking the channel such as Dial or Queue.
- The channels involved must be answered and bridged.

Enabling blind or attended transfers

In features.conf you must configure the blindxfer or atxfer options in the featuremap section. The options are configured with the DTMF character string you want to use for accessing the feature.

```
[featuremap]
blindxfer = #1
atxfer = *2
```

Now that you have the feature enabled you need to configure the dialplan such that a particular channel will be allowed to use the feature.

As an example if you want to allow transfers via the Dial application you can use two options, "t" or "T".

- t - Allow the called party to transfer the calling party by sending the DTMF sequence defined in features.conf. This setting does not perform policy enforcement on transfers initiated by other methods
- T - Allow the calling party to transfer the called party by sending the DTMF sequence defined in features.conf. This setting does not perform policy enforcement on transfers initiated by other methods.

Setting these options for Dial in extensions.conf would look similar to the following:

```
exten = 102,1,Dial(PJSIP/BOB,30,T)
```

Asterisk should be restarted or relevant modules should be reloaded for changes to take effect.

The same arguments ("t" and "T") work for the Queue and Dial applications!

Feature codes for attended transfer control

There are a few additional feature codes related to attended transfers. These features allow you to vary the behavior of an attended transfer on command. They are all configured in the 'general' section of features.conf

Aborting an attended transfer

Dialing the **atxferabort** code aborts an attended transfer. Otherwise there is no way to abort an attended transfer.

Completing an attended transfer

Dialing the **atxfercomplete** code completes an attended transfer and drops out of the call without having to hang up.

Completing an attended transfer as a three-way bridge

Dialing the **atxferthreeway** code completes an attended transfer and enters a bridge with both of the other parties.

Swapping between the transferee and destination

Dialing the **atxferswap** code swaps you between bridges with either party before the transfer is complete. This allows you to talk to either party one at a time before finalizing the attended transfer.

Example configuration

```
[general]
atxferabort = *3
atxfercomplete = *4
atxferthreeway = *5
atxferswap = *6
```

Configuring attended transfer callbacks

By default Asterisk will call back the initiator of the transfer if they hang up before the target answers and the answer timeout is reached. There are a few options for configuring this behavior.

No answer timeout

atxfernoanswertimeout allows you to define the timeout for attended transfers. This is the amount of time (in seconds) Asterisk will attempt to ring the target before giving up.

Dropped call behavior

atxferdropcall allows you to change the default callback behavior. The default is 'no' which results in Asterisk calling back the initiator of a transfer when they hang up early and the attended transfer times out. If set to 'yes' then the transfer target channel will be immediately transferred to the channel being transferred as soon as the initiator hangs up.

Loop delay timing

atxferloopdelay sets the number of seconds to wait between callback retries. This option is only relevant when atxferdropcall=no (or is undefined).

Number of retries for callbacks

atxfercallbackretries sets the number of times Asterisk will try to send a failed attended transfer back to the initiator. The default is 2.

Example Configuration

```
[general]
atxfernoanswertimeout = 15
atxferdropcall = no
atxferloopdelay = 10
atxfercallbackretries = 2
```

Behavior Options

These options are configured in the "[general]" section of features.conf

General transfer options

```
;transferdigittimeout = 3 ; Number of seconds to wait between digits when transferring a call
; (default is 3 seconds)
```

Attended transfer options

```
;xfersound = beep ; to indicate an attended transfer is complete
;xferfailsound = beeperr ; to indicate a failed transfer
;transferdialattempts = 3 ; Number of times that a transferer may attempt to dial an extension before
; being kicked back to the original call.
;transferretrysound = "beep" ; Sound to play when a transferer fails to dial a valid extension.
;transferinvalidsound = "beeperr" ; Sound to play when a transferer fails to dial a valid extension and is out of retries.
```

Basic Transfer Examples

In the previous section we configured #1 and *2 as our features codes. We also passed the "T" argument in the Dial application at 102 to allow transfers by the calling party.

Our hypothetical example includes a few devices:

- PJSIP/ALICE at extension 101
- PJSIP/BOB at extension 102
- PJSIP/CATHY at extension 103

Making a blind transfer

For blind transfers we configured the #1 feature code.

An example call flow:

- ALICE dials extension 102 to call BOB
- ALICE decides to transfer BOB to extension 103, so she dials #1. Asterisk will play the audio prompt "transfer".
- ALICE enters the digits 103 for the destination extension.
- Asterisk immediately hangs up the channel between ALICE and BOB. Asterisk creates a new channel for BOB that is dialing extension 103.

Making an attended transfer

For attended transfers we configured *2 as our feature code.

An example call flow:

- ALICE dials extension 102 to call BOB and BOB answers.
- ALICE decides to transfer BOB to extension 103, so she dials *2. Asterisk plays the audio prompt "transfer".
- ALICE enters the digits 103 for the destination extension. Asterisk places BOB on hold and creates a channel for ALICE to dial CATHY.
- CATHY answers - ALICE and CATHY talk. ALICE decides to complete the transfer and hangs up the phone.
- Asterisk immediately hangs up the channel between ALICE and BOB. Asterisk plays a short beep tone to CATHY and then bridges the channels for BOB and CATHY.

One-Touch Features

Overview

Once configured these features can be activated with only a few or even one keypress on a user's phone. They are often called "one-touch" or "one-step" features.

All of the features are configured via options in the featuremap section of features.conf and require arguments to be passed to the applications invoking the target channel.

Available Features

- **automon** - (One-touch Recording) Asterisk will invoke Monitor on the current channel.
- **automixmon** - (One-touch Recording) Has the same behavior as automon, but uses MixMonitor instead of Monitor.
- **disconnect** - (One-touch Disconnect) When this code is detected on a channel that channel will be immediately hung up.
- **parkcall** - (One-touch Parking) Sets a feature code for quickly parking a call.
    - Most parking options and behavior are configured in res_parking.conf in Asterisk 12 and newer.

Enabling the Features

Configuration of features.conf

The options are configured in features.conf in the featuremap section. They use typical Asterisk configuration file syntax.

features.conf

```
 [featuremap]
automon = *1
automixmon = *3
disconnect = *0
parkcall = #72
```

Assign each option the DTMF character string that you want users to enter for invoking the feature.

Dialplan application options

For each feature there are a pair of options that can be set in the Dial or Queue applications. The two options enable the feature on either the calling party channel or the called party channel.

> ⚠ If neither option of a pair are set then you will not be able to use the related feature on the channel.

automon

- W - Allow the calling party to enable recording of the call.
- w - Allow the called party to enable recording of the call.

automixmon

- X - Allow the calling party to enable recording of the call.
- x - Allow the called party to enable recording of the call.

disconnect

- H - Allow the calling party to hang up the channel.
- h - Allow the called party to hang up the channel.

parkcall

- K - Allow the calling party to enable parking of the call.
- k - Allow the called party to enable parking of the call.

Example usage

Set the option as you would any application option.

extensions.conf

```
exten = 101,1,Dial(PJSIP/ALICE,30,X)
```

This would allow the calling party, the party dialing PJSIP/ALICE, to invoke recording on the channel.

Using the Feature

One you have configured features.conf and the options in the application then you only have to enter the feature code on your phone's keypad during a call!

Call Pickup

Call pickup support added in Asterisk 11

Overview

Call pickup allows you to answer a call while it is ringing another phone or group of phones(other than the phone you are sitting at).

Requesting to pickup a call is done by two basic methods.

1. by dialplan using the Pickup or PickupChan applications.
2. by dialing the extension defined for pickupexten configured in features.conf.

Which calls can be picked up is determined by configuration and dialplan.

Dialplan Applications and Functions

Pickup Application

The Pickup application has three ways to select calls for pickup.

PickupChan Application

The PickupChan application tries to pickup the specified channels given to it.

CHANNEL Function

The CHANNEL function allows the pickup groups set on a channel to be changed from the defaults set by the channel driver when the channel was created.

callgroup/namedcallgroup

The CHANNEL(callgroup) option specifies which numeric pickup groups that this channel is a member.

```
same => n,Set(CHANNEL(callgroup)=1,5-7)
```

The CHANNEL(namedcallgroup) option specifies which named pickup groups that this channel is a member.

```
same => n,Set(CHANNEL(namedcallgroup)=engineering,sales)
```

For this option to be effective, you must set it on the outgoing channel. There are a couple of ways:

- You can use the setvar option available with several channel driver configuration files to set the pickup groups.
- You can use a pre-dial handler.

pickupgroup/namedpickupgroup

The CHANNEL(pickupgroup) option specifies which numeric pickup groups this channel can pickup.

```
same => n,Set(CHANNEL(pickupgroup)=1,6-8)
```

The CHANNEL(namedpickupgroup) option specifies which named pickup groups this channel can pickup.

```
same => n,Set(CHANNEL(namedpickupgroup)=engineering,sales)
```

For this option to be effective, you must set it on the channel before executing the Pickup application or calling the pickupexten.

- You can use the setvar option available with several channel driver configuration files to set the pickup groups.

Configuration Options

The pickupexten request method selects calls using the numeric and named call groups. The ringing channels have the callgroup assigned when the channel is created by the channel driver or set by the CHANNEL(callgroup) or CHANNEL(namedcallgroup) dialplan function.

Calls picked up using pickupexten can hear an optional sound file for success and failure.

> ⚠ The current channel drivers that support calling the pickupexten to pickup a call are: chan_dahdi/analog, chan_mgcp, chan_misdn, chan_sip, chan_unistim and chan_pjsip.

features.conf

```
pickupexten = *8               ; Configure the pickup extension. (default is *8)
pickupsound = beep             ; to indicate a successful pickup (default: no sound)
pickupfailsound = beeperr      ; to indicate that the pickup failed (default: no sound)
```

Numeric call pickup groups

A numeric callgroup and pickupgroup can be set to a comma separated list of ranges (e.g., 1-4) or numbers that can have a value of 0 to 63. There can be a maximum of 64 numeric groups.

SYNTAX

```
callgroup=[number[-number][,number[-number][,...]]]
pickupgroup=[number[-number][,number[-number][,...]]]
```

- callgroup - specifies which numeric pickup groups that this channel is a member.
- pickupgroup - specifies which numeric pickup groups this channel can pickup.

Configuration example

```
callgroup=1,5-7
pickupgroup=1
```

Configuration should be supported in several channel drivers, including:

- chan_dahdi.conf
- misdn.conf
- mgcp.conf
- sip.conf
- unistim.conf
- pjsip.conf

pjsip.conf uses snake case:

Configuration in pjsip.conf

```
call_group=1,5-7
pickup_group=1
```

Named call pickup groups

A named callgroup and pickupgroup can be set to a comma separated list of case sensitive name strings. The number of named groups is unlimited. The number of named groups you can specify at once is limited by the line length supported.

SYNTAX

```
namedcallgroup=[name[,name[,...]]]
namedpickupgroup=[name[,name[,...]]]
```

- namedcallgroup - specifies which named pickup groups that this channel is a member.
- namedpickupgroup - specifies which named pickup groups this channel can pickup.

Configuration Example

```
namedcallgroup=engineering,sales,netgroup,protgroup
namedpickupgroup=sales
```

Configuration should be supported in several channel drivers, including:

- chan_dahdi.conf
- misdn.conf
- sip.conf
- pjsip.conf

pjsip.conf uses snake case:

```
named_call_group=engineering,sales,netgroup,protgroup
named_pickup_group=sales
```

⚠ You can use named pickup groups in parallel with numeric pickup groups. For example, the named pickup group '4' is not the same as the numeric pickup group '4'.

Named pickup groups are new with Asterisk 11.

Built-in Dynamic Features

The FEATURE and FEATUREMAP dialplan functions allow you to set some features.conf options on a per channel basis.

> To see what options are currently supported, look at the FEATURE and FEATUREMAP function descriptions. **These functions were added in Asterisk 11.**
>
> At this time the functions do not work with custom features. Those are set with a channel variable as described in the Custom Dynamic Features section.

Set the parking time of this channel to be 100 seconds if it is parked.

```
exten => s,1,Set(FEATURE(parkingtime)=100)
same => n,Dial(SIP/100)
same => n,Hangup()
```

Set the DTMF sequence for attended transfer on this channel to *9.

```
exten => s,1,Set(FEATUREMAP(atxfer)=*9)
same => n,Dial(SIP/100,,T)
same => n,Hangup()
```

Custom Dynamic Features

Overview

Asterisk allows you to define custom features mapped to Asterisk applications. You can then enable these features dynamically, on a per-channel basis by using a channel variable.

Defining the Features

Custom features are defined in the **applicationmap** section of the features.conf file.

Syntax:

```
[applicationmap]
<FeatureName> = <DTMF_sequence>,<ActivateOn>[/<ActivatedBy>],<Application>[,<AppArguments>[,MOH_Class]]
<FeatureName> = <DTMF_sequence>,<ActivateOn>[/<ActivatedBy>],<Application>[,"<AppArguments>"[,MOH_Class]]
<FeatureName> = <DTMF_sequence>,<ActivateOn>[/<ActivatedBy>],<Application>([<AppArguments>])[,MOH_Class]
```

Syntax Fields:

Field Name	Description
FeatureName	This is the name of the feature used when setting the DYNAMIC_FEATURES variable to enable usage of this feature.
DTMF_sequence	This is the key sequence used to activate this feature.
ActivateOn	This is the channel of the call that the application will be executed on. Valid values are "self" and "peer". "self" means run the application on the same channel that activated the feature. "peer" means run the application on the opposite channel from the one that has activated the feature.
ActivatedBy	ActivatedBy is no longer honored. The feature is activated by which channel DYNAMIC_FEATURES includes the feature is on. Use a pre-dial handler to set different values for DYNAMIC_FEATURES on the channels. Historic values are: "caller", "callee", and "both".
Application	This is the application to execute.
AppArguments	These are the arguments to be passed into the application. If you need commas in your arguments, you should use either the second or third syntax, above.
MOH_Class	This is the music on hold class to play while the idle channel waits for the feature to complete. If left blank, no music will be played.

Application Mapping

The applicationmap is not intended to be used for all Asterisk applications. When applications are used in extensions.conf, they are executed by the PBX core. In this case, these applications are executed outside of the PBX core, so it does *not* make sense to use any application which has any concept of dialplan flow. Examples of this would be things like Goto, Background, WaitExten, and many more. The exceptions to this are Gosub and Macro routines which must complete for the call to continue.

Enabling these features means that the PBX needs to stay in the media flow and media will not be re-directed if DTMF is sent in the media stream.

Example Feature Definitions:

Here we have defined a few custom features to give you an idea of how the configuration looks.

features.conf

```
 [applicationmap]
playmonkeys => #9,peer,Playback,tt-monkeys
retrieveinfo => #8,peer,Set(ARRAY(CDR(mark),CDR(name))=${ODBC_FOO(${CALLERID(num)})})
pauseMonitor   => #1,self/callee,Pausemonitor
unpauseMonitor => #3,self/callee,UnPauseMonitor
```

Example feature descriptions:

- playmonkeys - Allow both the caller and callee to play tt-monkeys to the bridged channel.
- retrieveinfo - Set arbitrary channel variables, based upon CALLERID number (Note that the application argument contains commas)
- pauseMonitor - Allow the callee to pause monitoring on their channel.

- unpauseMonitor - Allow the callee to unpause monitoring on their channel.

Enabling Features

After you define a custom feature in features.conf you must enable it on a channel by setting the DYNAMIC_FEATURES channel variable.

DYNAMIC_FEATURES accepts as an argument a list of hash-sign delimited feature names.

Example Usage:

extensions.conf

```
Set(__DYNAMIC_FEATURES=playmonkeys#pauseMonitor#unpauseMonitor)
```

Tip: Variable Inheritance
The two leading underscores allow these feature settings to be set on the outbound channels, as well. Otherwise, only the original channel will have access to these features.

Call Parking

Overview

Some organizations have the need to facilitate calls to employees who move around the office a lot or who don't necessarily sit at a desk all day. In Asterisk, it is possible to allow a call to be put on hold at one location and then picked up from a different location such that the conversation can be continued from a device other than the one from which call was originally answered. This concept is known as call parking.

Call parking is a feature that allows a participant in a call to put the other participants on hold while they themselves hang up. When parking, the party that initiates the park will be told a parking space, which under standard configuration doubles as an extension. This extension, or parking space, serves as the conduit for accessing the parked call. At this point, as long as the parking space is known, the parked call can be retrieved from a different location/device from where it was originally answered.

Call Parking Configuration Files and Module

In versions of Asterisk prior to Asterisk 12, call parking was considered an Asterisk core feature and was configured using `features.conf` . However, Asterisk 12 underwent vast architectural changes, several of which were directed at call parking support. Because the amount of changes introduced in Asterisk 12 was quite extensive, they have been omitted from this document. For reference, you can find a comprehensive list of these changes here: New in 12 .

In a nutshell, Asterisk 12 relocated its support for call parking from the Asterisk core into a separate, loadable module, `res_parking` . As a result, configuration for call parking was also moved to `res_parking.conf` . Configuration for call parking through `features.conf` for versions of Asterisk 12 and beyond, is no longer supported. Additionally, support for the `ParkAndAnnounce` application was relocated to the `res_parking` module and the `app_parkandannounce` module was removed.

Before we move any further, there is one more rather important detail to address regarding configuration for `res_parking` :

> ⚠ `res_parking` uses the configuration framework. If an invalid configuration is supplied, `res_parking` will fail to load or fail to reload. Previously, invalid configurations would generally be accepted, with certain errors resulting in individually disabled parking lots.

Now that we've covered all of that, let's look at some examples of how all this works.

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Example Configurations

Basic Call Parking/Retrieval Scenario

This is a basic scenario that only requires minimal adjustments to the following configuration files: **`res_parking.conf`** , **`features.conf`** , and **`extensions.conf`** .

In this scenario, our dialplan contains an extension to accept calls from the outside. Let's assume that the extension the caller dialed was: **`5555001`** . The handler will then attempt to dial the **`alice`** extension, using the **`k`** option.

Sadly for our caller, the **`alice`** extension answers the call and immediately after saying, "Hello world!", sends the DTMF digits to invoke the call parking feature without giving the caller a chance to speak. The **`alice`** extension quickly redeems itself by using the **`GoTo`** application to navigate to the **`701`** extension in the **`parkedcalls`** context to retrieve the parked call. But, since the next thing the **`alice`** extension does is hangup on our caller, I am beginning to think the **`alice`** extension doesn't want to be bothered.

In summary:

- Outside caller dials **`5555001`**
- Alice picks up the device and says "Hello World!"
- Alice presses the one touch parking DTMF combination
- Alice then dials the extension that the call was parked to (**`701`**) to retrieve the call
- Alice says, "Goodbye", and disconnects the caller

res_parking.conf

```
[general]
parkext => 700                           ; Sets the default extension used to park calls.
Note: This option
                                         ; can take any alphanumeric string.

parkpos => 701-709                       ; Sets the range of extensions used as the
parking lot. Parked calls
                                         ; may be retrieved by dialing the numbers in
this range. Note: These
                                         ; need to be numeric, as Asterisk starts from
the start position and
                                         ; increments with one for the next parked call.

context => parkedcalls                   ; Sets the default dialplan context where the
parking extension and
                                         ; the parking lot extensions are created. These
will be automatically
                                         ; generated since we have specified a value for
the 'parkext' option
                                         ; above. If you need to use this in your
dialplan (extensions.conf),
                                         ; just include it like: include => parkedcalls.

parkingtime => 300                       ; Specifies the number of seconds a call will
wait in the parking
                                         ; lot before timing out. In this example, a
parked call will time out
                                         ; if it is not un-parked before 300 seconds (5
minutes) elapses.

findslot => next                         ; Configures the parking slot selection
behavior. For this example,
                                         ; the next free slot will be selected when a
call is parked.
```

features.conf

```
[featuremap]
parkcall => #72                          ; Parks the call (one-step parking). For this
example, a call will be
                                         ; automatically parked when an allowed party
presses the DTMF digits,
                                         ; #·7·2. A party is able to make use of this
when the the K/k options
                                         ; are used when invoking the Dial() application.
For convenience, the
                                         ; values of this option are defined below:
                                         ; K - Allow the calling party to enable parking
of the call.
                                         ; k - Allow the called party to enable parking
of the call.
```

extensions.conf

```
[globals]
; Extension Maps
5001=alice                              ; Maps 5001 to a local extension that will
emulate
                                        ; a party pressing DTMF digits from a device.
;5001=PJSIP/sip:alice@127.0.0.1:5060     ; What a realistc mapping for the alice device
would look like.

; Realistically, 'alice' would map to a channel for a local device that would receive the
call, therefore
; rendering this extension unnecessary. However, for the purposes of this demonstration,
the extension is
; presented to you to show that sending the sequence of DTMF digits defined in the
'parkcall' option in
; 'features.conf' is the trigger that invokes the one-step parking feature.

[parking-example]
include => parkedcalls

exten => alice,1,NoOp(Handles calls to alice.)
  same => n,Answer()
  same => n,Playback(hello-world)
  same => n,SendDTMF(#72w)
  same => n,Goto(parkedcalls,701,1)
  same => n,Playback(vm-goodbye)
  same => n,Hangup()

[from-outside]
exten => 5555001,1,NoOp(Route to a local extension.)
  ; Dials the device that is mapped to the local resource, alice, giving the recipient of
the call the ability
  ; to park it. Assuming the value of LocalExtension is 5001, the Dial() command will
look like: Dial(alice,,k)
  same => n,Dial(PJSIP/alice)
  same => n,Hangup()
```

Basic Handling for Call Parking Timeouts

Next we will move on to explain how to handle situations where a call is parked but is not retrieved before the value specified as the **parkingtime** option elapses. Just like the scenario above, this is a basic scenario that only requires minimal adjustments to the following configuration files: **res_parking.conf** , **features.conf** , and **extensions.conf** .

Like before, our dialplan contains an extension to accept calls from the outside. Again, let's assume that the extension the caller dialed was: **5555001** . The handler will then attempt to dial the **alice** extension, using the **k** option.

Sadly for our caller, the **alice** extension answers the call and immediately sends the DTMF digits to invoke the call parking feature without giving the caller a chance to speak. Unlike in the previous scenario, however, the **alice** extension does not retrieve the parked call. Our sad caller is now even more sad.

After a period of **300 seconds** , or **5 minutes** (as defined in the **parkingtime** option in **res_parking.conf**), the call will time out. Because we told Asterisk to return a timed-out parked call to the party that originally parked the call (**comebacktoorigin=yes**), Asterisk will attempt to call **alice** using an extension automagically created in the special context, **park-dial** .

Unfortunately, the **alice** extension has no time to be bothered with us at this moment, so the call is not answered. After a period of **20 seconds** elapses (the value specified for the **comebackdialtime** option in **res_parking.conf**), Asterisk finally gives up and the **t** extension in the **park-dial** context is executed. Our caller is then told "Goodbye" before being disconnected.

In summary:

- Outside caller dials **5555001**
- Alice picks up the device and says "Hello World!"

- Alice presses the one touch parking DTMF combination
- The parked call times out after 300 seconds
- Asterisk sends the call to the origin, or the **`alice`** extension
- A period of **`20 seconds`** elapses without an answer
- Asterisk sends the call to **`t`** extension in the **`park-dial`** context
- Our caller hears, "Goodbye", before being disconnected

res_parking.conf

```
[general]
parkext => 700                           ; Sets the default extension used to park calls.
Note: This option
                                         ; can take any alphanumeric string.

parkpos => 701-709                       ; Sets the range of extensions used as the
parking lot. Parked calls
                                         ; may be retrieved by dialing the numbers in
this range. Note: These
                                         ; need to be numeric, as Asterisk starts from
the start position and
                                         ; increments with one for the next parked call.

context => parkedcalls                   ; Sets the default dialplan context where the
parking extension and
                                         ; the parking lot extensions are created. These
will be automatically
                                         ; generated since we have specified a value for
the 'parkext' option
                                         ; above. If you need to use this in your
dialplan (extensions.conf),
                                         ; just include it like: include => parkedcalls.

parkingtime => 300                       ; Specifies the number of seconds a call will
wait in the parking
                                         ; lot before timing out. In this example, a
parked call will time out
                                         ; if it is not un-parked before 300 seconds (5
minutes) elapses.

findslot => next                         ; Configures the parking slot selection
behavior. For this example,
                                         ; the next free slot will be selected when a
call is parked.

comebackdialtime=20                      ; When a parked call times out, this is the
number of seconds to dial
                                         ; the device that originally parked the call, or
the PARKER
                                         ; channel variable. The value of
'comebackdialtime' is available as
                                         ; the channel variable 'COMEBACKDIALTIME' after
a parked call has
                                         ; timed out. For this example, when a parked
call times out, Asterisk
                                         ; will attempt to call the PARKER for 20
seconds, using an extension
                                         ; it will automatically create in the
'park-dial' context. If the
                                         ; party does not answer the call during this
period, Asterisk will
```

```
                                         ; continue executing any remaining priorities in
the dialplan.

comebacktoorigin=yes                     ; Determines what should be done with a parked
call if it is not
                                         ; retrieved before the time specified in the
'parkingtime' option
                                         ; elapses. In the case of this example where
'comebacktoorigin=yes',
                                         ; Asterisk will attempt to return the parked
call to the party that
                                         ; originally parked the call, or the PARKER
channel variable, using
                                         ; an extension it will automatically create in
```

```
the 'park-dial'
                                         ; context.
```

features.conf

```
[featuremap]
parkcall => #72                          ; Parks the call (one-step parking). For this
example, a call will be
                                         ; automatically parked when an allowed party
presses the DTMF digits,
                                         ; #·7·2. A party is able to make use of this
when the the K/k options
                                         ; are used when invoking the Dial() application.
For convenience, the
                                         ; values of this option are defined below:
                                         ; K - Allow the calling party to enable parking
of the call.
                                         ; k - Allow the called party to enable parking
of the call.
```

extensions.conf

```
[globals]
; Extension Maps
5001=alice                               ; Maps 5001 to a local extension that will
emulate
                                         ; a party pressing DTMF digits from a device.
;5001=PJSIP/sip:alice@127.0.0.1:5060     ; What a realistc mapping for the alice device
would look like.

; Realistically, 'alice' would map to a channel for a local device that would receive the
call, therefore
; rendering this extension unnecessary. However, for the purposes of this demonstration,
the extension is
; presented to you to show that sending the sequence of DTMF digits defined in the
'parkcall' option in
; 'features.conf' is the trigger that invokes the one-step parking feature.

[parking-example]
include => parkedcalls

exten => alice,1,NoOp(Handles calls to alice.)
  same => n,Answer()
  same => n,Playback(hello-world)
  same => n,SendDTMF(#72w)
  same => n,Wait(300)
  same => n,Hangup()

[from-outside]
exten => 5555001,1,NoOp(Route to a local extension.)
  ; Dials the device that is mapped to the local resource, alice, giving the recipient of
the call the ability
  ; to park it. Assuming the value of LocalExtension is 5001, the Dial() command will
look like: Dial(alice,,k)
  same => n,Dial(PJSIP/alice)
  same => n,Hangup()

[park-dial]
; Route here if the party that initiated the call parking cannot be reached after a
period of time equaling the
; value specified in the 'comebackdialtime' option elapses.
exten => t,1,NoOp(End of the line for a timed-out parked call.)
  same => n,Playback(vm-goodbye)
  same => n,Hangup()
```

Custom Handling for Call Parking Timeouts

Finally, we will move on to explain how to handle situations where upon a parked call session timing out, it is not desired to return to the parked call to the device from where the call was originally parked. (This might be handy for situations where you have a dedicated receptionist or service desk extension to handle incoming call traffic.) Just like the previous two examples, this is a basic scenario that only requires minimal adjustments to the following configuration files: **`res_parking.conf`** , **`features.conf`** , and **`extensions.conf`** .

Like before, our dialplan contains an extension to accept calls from the outside. Again, let's assume that the extension the caller dialed was: **`5555001`** . The handler will then attempt to dial the **`alice`** extension, using the **`k`** option.

Sadly for our caller, the **`alice`** extension answers the call and immediately sends the DTMF digits to invoke the call parking feature without giving the caller a chance to speak. Just like in the previous scenario, the **`alice`** extension does not retrieve the parked call. Maybe the **`alice`** extension is having a bad day.

After a period of **`300 seconds`** , or **`5 minutes`** (as defined in the **`parkingtime`** option in **`res_parking.conf`**), the call will time out. Because we told Asterisk to send a timed-out parked call to the **`parkedcallstimeout`** context (**`comebacktoorigin=no`**), we are able to bypass the default logic that directs Asterisk to returning the call to the person who initiated the park. In our example, when a parked call enters our **`s`** extension in our **`parkedcallstimeout`** context, we only play a sound file to the caller and hangup the call, but this is where you could do any custom logic like returning the call to a different extension, or performing a lookup of some sort.

In summary:

- Outside caller dials **`5555001`**
- Alice picks up the device and says "Hello World!"
- Alice presses the one touch parking DTMF combination
- The parked call times out after 300 seconds
- Asterisk sends the call to the **`s`** extension in our **`parkedcallstimeout`**
- Our caller hears, "Goodbye", before being disconnected

res_parking.conf

```
[general]

[default]
parkext => 700                           ; Sets the default extension used to park calls.
Note: This option
                                         ; can take any alphanumeric string.

parkpos => 701-709                       ; Sets the range of extensions used as the
parking lot. Parked calls
                                         ; may be retrieved by dialing the numbers in
this range. Note: These
                                         ; need to be numeric, as Asterisk starts from
the start position and
                                         ; increments with one for the next parked call.

context => parkedcalls                   ; Sets the default dialplan context where the
parking extension and
                                         ; the parking lot extensions are created. These
will be automatically
                                         ; generated since we have specified a value for
the 'parkext' option
                                         ; above. If you need to use this in your
dialplan (extensions.conf),
                                         ; just include it like: include => parkedcalls.

parkingtime => 300                       ; Specifies the number of seconds a call will
wait in the parking
                                         ; lot before timing out. In this example, a
parked call will time out
                                         ; if it is not un-parked before 300 seconds (5
minutes) elapses.

findslot => next                         ; Configures the parking slot selection
behavior. For this example,
                                         ; the next free slot will be selected when a
call is parked.

comebacktoorigin=no                      ; Determines what should be done with a parked
call if it is not
                                         ; retrieved before the time specified in the
'parkingtime' option
                                         ; elapses.
                                         ;
                                         ; Setting 'comebacktoorigin=no' (like in this
example) is for cases
                                         ; when you want to perform custom dialplan logic
```

```
to gracefully handle
                                          ; the remainder of the parked call when it times
out.

comebackcontext=parkedcallstimeout        ; The context that a parked call will be routed
to in the event it
                                          ; times out. Asterisk will first attempt to
route the call to an
                                          ; extension in this context that matches the
flattened peer name. If
                                          ; no such extension exists, Asterisk will next
attempt to route the
                                          ; call to the 's' extension in this context.
Note: If you set
                                          ; 'comebacktoorigin=no' in your configuration
but do not define this
                                          ; value, Asterisk will route the call to the 's'
```

```
extension in the
                                         ; default context.
```

features.conf

```
[featuremap]
parkcall => #72                          ; Parks the call (one-step parking). For this
example, a call will be
                                         ; automatically parked when an allowed party
presses the DTMF digits,
                                         ; #·7·2. A party is able to make use of this
when the the K/k options
                                         ; are used when invoking the Dial() application.
For convenience, the
                                         ; values of this option are defined below:
                                         ; K - Allow the calling party to enable parking
of the call.
                                         ; k - Allow the called party to enable parking
of the call.
```

extensions.conf

```
[globals]
; Extension Maps
5001=alice                               ; Maps 5001 to a local extension that will
emulate
                                         ; a party pressing DTMF digits from a device.
;5001=PJSIP/sip:alice@127.0.0.1:5060     ; What a realistc mapping for the alice device
would look like.

; Realistically, 'alice' would map to a channel for a local device that would receive the
call, therefore
; rendering this extension unnecessary. However, for the purposes of this demonstration,
the extension is
; presented to you to show that sending the sequence of DTMF digits defined in the
'parkcall' option in
; 'features.conf' is the trigger that invokes the one-step parking feature.

[parking-example]
include => parkedcalls

exten => alice,1,NoOp(Handles calls to alice.)
  same => n,Answer()
  same => n,Playback(hello-world)
  same => n,SendDTMF(#72w)
  same => n,Wait(300)
  same => n,Hangup()

[from-outside]
exten => 5555001,1,NoOp(Route to a local extension.)
  ; Dials the device that is mapped to the local resource, alice, giving the recipient of
the call the ability
  ; to park it. Assuming the value of LocalExtension is 5001, the Dial() command will
look like: Dial(alice,,k)
  same => n,Dial(PJSIP/alice)
  same => n,Hangup()

[parkedcallstimeout]
exten => s,1,NoOp(This is all that happens to parked calls if they time out.)
  same => n,Playback(vm-goodbye)
  same => n,Hangup()
```

Applications

Asterisk Dialplan Applications

In this section we'll discuss Asterisk dialplan applications. The concept of Asterisk applications shouldn't be confused with the more general idea of building an "application" with Asterisk, where one is talking about what Asterisk is doing in relation to the outside world.

For example in the general sense; an application of Asterisk could be a voicemail server or PBX. In contrast, an **Asterisk dialplan application** is a unit of functionality provided via an Asterisk module that can be called via dialplan or one of Asterisk's APIs. Dialplan applications do some work on the channel, such as answering a call or playing back a sound prompt. There are a wide variety of dialplan applications available for your use.

The term **application** in Asterisk documentation and on Asterisk discussion forums is usually referring to dialplan applications.

Available applications

Many applications come with Asterisk by default. For a complete list of the dialplan applications available to your installation of Asterisk, type **core show applications** at the Asterisk CLI. Not all applications are compiled with Asterisk by default so if you have the source available then you may want to browse the applications listed in menuselect.

Since anyone can write an Asterisk application they can also be obtained from sources outside the Asterisk distribution. Pre-packaged community or commercial Asterisk distributions that have a special purpose may include a custom application or two.

General application syntax

Each extension priority in the dialplan calls an application. Most applications take one or more parameters, which provide additional information to the application or change its behavior. Parameters should be separated by commas.

You can find examples of Asterisk extensions and priorities in the Contexts, Extensions, and Priorities section. You'll notice that most extensions look similar to this example:

```
exten =>
6123,1,ApplicationName(ParamOne,ParamTwo,ParamThree)
```

ApplicationName is where you put the name of an application you want to call.

ParamOne, two and three are the parameters passed to the application. Each is separated by a comma. Some applications don't require any parameters, but most do.

> Using the pipe character or vertical bar character (|) to delimit parameters is deprecated syntax. Use commas instead!

Help for specific applications

The wiki section CLI Syntax and Help Commands details how to use the CLI-accessible documentation. This will allow you to access the syntax and usage info for each application including detail on all the options for each application.

Wikibot also publishes the same documentation on the wiki. You can find applications docs in the version specific top level sections; such as Asterisk 13 Dialplan Applications.

Commonly used applications

There are many Asterisk applications, but several are very commonly used and essential to almost every Asterisk system. The sub-pages in the section will provide some further detail and usage of these common Asterisk applications.

On This Page

Topics

Answer, Playback, and Hangup Applications

As its name suggests, the **Answer()** application answers an incoming call. The **Answer()** application takes a delay (in milliseconds) as its first parameter. Adding a short delay is often useful for ensuring that the remote endpoing has time to begin processing audio before you play a sound prompt. Otherwise, you may not hear the very beginning of the prompt.

Knowing When to Answer a Call

When you're first learning your way around the Asterisk dialplan, it may be a bit confusing knowing when to use the **Answer()** application, and when not to.

If Asterisk is simply going to pass the call off to another device using the **Dial()** application, you probably don't want to call the answer the call first. If, on the other hand, you want Asterisk to play sound prompts or gather input from the caller, it's probably a good idea to call the **Answer()** application before doing anything else.

The **Playback()** application loads a sound prompt from disk and plays it to the caller, ignoring any touch tone input from the caller. The first parameter to the dialplan application is the filename of the sound prompt you wish to play, without a file extension. If the channel has not already been answered, **Playback()** will answer the call before playing back the sound prompt, unless you pass **noanswer** as the second parameter.

To avoid the first few milliseconds of a prompt from being cut off you can play a second of silence. For example, if the prompt you wanted to play was hello-world which would look like this in the dialplan:

```
exten => 1234,1,Playback(hello-world)
```

You could avoid the first few seconds of the prompt from being cut off by playing the silence/1 file:

```
exten => 1234,1,Playback(silence/1)
exten => 1234,n,Playback(hello-world)
```

Alternatively this could all be done on the same line by separating the filenames with an ampersand (&):

```
exten => 1234,1,Playback(silence/1&hello-world)
```

The **Hangup()** application hangs up the current call. While not strictly necessary due to auto-fallthrough (see the note on Priority numbers above), in general we recommend you add the **Hangup()** application as the last priority in any extension.

Now let's put **Answer()**, **Playback()**, and **Hangup()** together to play a sample sound file.

```
exten => 6000,1,Answer(500)
exten => 6000,n,Playback(hello-world)
exten => 6000,n,Hangup()
```

Bridge Application

Overview of the Bridge Application

The Bridge application takes two channels and attempts to put them into a Bridge. Both channels and bridges are very common elements of Asterisk operation, so this is a really useful application to learn.

For Bridge to work, two channels are required to exist. That is the channel executing the Bridge application and a target channel that you want to bridge with. If the operation is successful the two channels will be bridged and media exchanged among the channels.

To help out let's contrast Dial and Bridge:

- When a channel executes Dial a new channel is created for the target device. If the new channel is answered then both channels are bridged.
- When a channel executes Bridge then Asterisk will attempt to bridge the two existing channels; the executing channel and the target channel.

Note that bridge provides several options to tweak behavior and upon completion Bridge will return status on a channel variable. These are detailed in the app documentation.

Using the Bridge application

Read the Bridge documentation for your version of Asterisk (e.g. Asterisk 13 - Asterisk 13 Application_Bridge) and the Key Concepts section on Bridges to get a good start.

See Also

- Bridging Modules
- Pre-Bridge Handlers
- Introduction to ARI and Bridges
- Asterisk 13 Application_BridgeWait
- Conferencing Applications

Conferencing Applications

Conferencing with Asterisk

Up until about Asterisk 1.6; app_meetme was the main application providing conferencing style features. In Asterisk 1.6.2 the ConfBridge module was added and then rewritten in Asterisk 10.

Both MeetMe and ConfBridge still exist in the latest Asterisk versions and provide different feature sets, but with plenty of overlap. Development attention is primarily given to ConfBridge these days and it is the recommended option for modern deployments when using a pre-built application.

There is a detailed description of ConfBridge functionality on the wiki as well as MeetMe application usage notes.

Building your own conferencing application

Conferencing needs can be very specific to your business application. The conferencing applications included with Asterisk provide basic features that will work for many users.

If the included applications don't work for you then you might consider building your own application using the Asterisk REST Interface. This will give you access to all the communication primitives needed and then you can write the logic you need in a language you are comfortable with.

ConfBridge

Overview

Asterisk, since its early days, has offered a conferencing application called MeetMe (app_meetme.so). MeetMe provides DAHDI-mixed software-based bridges for multi-party audio conferencing. MeetMe is used by nearly all Asterisk implementations - small office, call center, large office, feature-server, third-party application, etc. It has been extremely successful as an audio bridge.

Over time, several significant limitations of MeetMe have been encountered by its users. Among these are two of distinction: MeetMe requires DAHDI for mixing, and is thus limited to 8kHz (PSTN) audio sampling rates; and MeetMe is delivered in a fairly static form, it does not provide extensive configuration options.

To address these limitations, a new conferencing application, based upon the ConfBridge application introduced in Asterisk 1.6.0, is now available with Asterisk 10. This new ConfBridge application replaces the older ConfBridge application. It is not intended to be a direct replacement for MeetMe, it will not provide feature parity with the MeetMe application. Instead, the new ConfBridge application delivers a completely redesigned set of functionality that most users will find more than sufficient, and in many ways better, for their conferencing needs.

ConfBridge Concepts

ConfBridge provides four internal concepts:

1. Conference Number
2. Bridge Profile
3. User Profile
4. Conference Menu

A **Conference Number** is a numerical representation for an instance of the bridge. Callers joined to the same conference number will be in the same conference bridge; they're connected. Callers joined to different conference numbers are not in the same conference bridge; they're separated. Conference Numbers are assigned in the dialplan. Unlike MeetMe, they're not pre-reserved.

A **Bridge Profile** is a named set of options that control the behavior of a particular conference bridge. Each bridge must have its own profile. A single bridge cannot have more than one Bridge Profile.

A **User Profile** is a named set of options that control the user's experience as a member of a particular bridge. Each user participating in a bridge can have their own individual User Profile.

A **Conference Menu** is a named set of options that are provided to a user when they present DTMF keys while connected to the bridge. Each user participating in a bridge can have their own individual Conference Menu.

ConfBridge Application Syntax

The ConfBridge application syntax and usage can be found at Asterisk 13 Application_ConfBridge

ConfBridge Application Examples

Example 1
In this example, callers will be joined to conference number 1234, using the default Bridge Profile, the default User Profile, and no Conference Menu.

```
exten => 1,1,Answer()
exten => 1,n,ConfBridge(1234)
```

Example 2
In this example, callers will be joined to conference number 1234, with the default Bridge Profile, a User Profile called "1234_participants" and a Conference Menu called "1234_menu."

```
exten => 1,1,Answer()
exten => 1,n,ConfBridge(1234,,1234_participants,1234_menu)
```

Usage Notes, FAQ and Other

There are many points to consider when using the new ConfBridge appliation. Some will be examined here.

Video Conferencing

It is imperative that a video conference not have participants using disparate video codecs or encoding profiles. Everyone **must** use the same codec and profile. Otherwise, the video sessions won't work - you'll likely experience frozen video as the conference switches from one video stream using a codec your client has negotiated, to a video stream using a codec your client hasn't negotiated or doesn't support.

Video Endpoints

ConfBridge has been tested against a number of video-capable SIP endpoints. Success, and your mileage will vary.

Endpoints that work:

- Jitsi - Jitsi works well for both H.264 and H.263+1998 video calling on Mac, Linux and Windows machines. Currently, Jitsi seems to be the best-working, free, H.264-capable SIP video client.
- Linphone - Linphone works well for H.263+1998 and H.263 video calling on Linux - the Mac port and mobile ports do not support video. Currently, Linphone seems to be the best-working, free, H.263-capable SIP video client, when Jitsi or H.263+1998 aren't an option.
- Empathy - Empathy works for H.264 calling, but is amazingly difficult to configure (why one has to make two SIP accounts just to make a SIP call is a mystery).
- Lifesize - The Lifesize client supports H.264 and runs on Windows only. It works very well, but it isn't free.
- Polycom VVX 1500 - The Polycom VVX 1500 works well for H.264 calling. If you're connecting it to Jitsi, you may have to configure Jitsi to use the Baseline H.264 profile instead of the Main profile.

Endpoints that don't or weren't tested:

- Xlite - Xlite works in some cases, but also seems to crash, regardless of operating system, at odd times. In other cases, Xlite isn't able to decode video from clients.
- Ekiga - Ekiga wasn't tested, because our test camera wasn't supported by the client. The same camera was supported by other soft clients.
- SIPDroid - SIPDroid doesn't seem to work.
- OfficeSIP Messenger - OfficeSIP Messenger didn't seem capable of performing a SIP registration. On this basis alone, no one should recommend its use.

Mixing Interval

The mixing interval for a conference is defined in its Bridge Profile. The allowable options are 10, 20, 40, and 80, all in milliseconds. Usage of 80ms mixing intervals is only supported for conferences that are sampled at 8, 12, 16, 24, 32, and 48kHz. Usage of 40ms intervals includes all of the aforementioned sampling rates as well as 96kHz. 192kHz sampled conferences are only supported at 10 and 20ms mixing intervals. These limitations are imposed because higher mixing intervals at the higher sampling rates causes large increases in memory consumption. Adventurous users may, through changing of the MAX_DATALEN define in bridge_softmix.c allow 96kHz and 192kHz sampled conferences to operate at longer intervals - set to 16192 for 96kHz at 80ms or 32384 for 192kHz at 80ms, recompile, and restart.

Maximizing Performance

In order to maximize the performance of a given machine for ConfBridge purposes, there are several steps one should take.

- Enable dsp_drop_silence is enabled in the User Profile.
  - This is the **single most** important step one can take when trying to increase the number of bridge participants that a single machine can handle. Enabling this means that the audio of users that aren't speaking isn't mixed in with the bridge.
- Lengthen mixing_interval in the Bridge Profile.
  - The default interval is 20ms. Other options are 10, 40, and 80ms. Lower values provide a "tighter" sound, but require substantially more CPU. Higher values provider a "looser" sound, and consume substantially less CPU. Setting the value to 80 provides the highest number of possible participants.
- Connect clients at the same sampling rate.
  - Requiring the bridge to resample between clients that use codecs with different sampling rates is an expensive operation. If all clients are dialed in to the bridge at the same sampling rate, and the bridge operates at that same rate, e.g. 16kHz, then the number of possible clients will be maximized.
- Run Asterisk with a higher priority.
  - By default, Asterisk operates at a relatively normal priority, as compared to other processes on the system. To maximize the number of possible clients, Asterisk should be started using the **-p** (realtime) flag. If the load becomes too large, this can negatively impact the performance of other processes, including the console itself - making it difficult to remotely administer a fully loaded system.

As the number of clients approaches the maximum possible on the given machine, given its processing capabilities, audio quality will suffer. Following the above guidelines will increase the number of connected clients before audio quality suffers.

ConfBridge AMI Actions

On This Page

ConfbridgeList

Lists all users in a particular ConfBridge conference. ConfbridgeList will follow as separate events, followed by a final event called ConfbridgeListComplete

Example

```
Action: ConfbridgeList
Conference: 1111

Response: Success
EventList: start
Message: Confbridge user list will follow

Event: ConfbridgeList
Conference: 1111
CallerIDNum: malcolm
CallerIDName: malcolm
Channel: SIP/malcolm-00000000
Admin: No
MarkedUser: No

Event: ConfbridgeListComplete
EventList: Complete
ListItems: 1
```

ConfbridgeListRooms

Lists data about all active conferences. ConfbridgeListRooms will follow as separate events, followed by a final event called ConfbridgeListRoomsComplete.

Example

```
Action: ConfbridgeListRooms

Response: Success
EventList: start
Message: Confbridge conferences will follow

Event: ConfbridgeListRooms
Conference: 1111
Parties: 1
Marked: 0
Locked: No

Event: ConfbridgeListRoomsComplete
EventList: Complete
ListItems: 1
```

ConfbridgeMute

Mutes a specified user in a specified conference.

Example

```
Action: ConfbridgeMute
Conference: 1111
Channel: SIP/mypeer-00000001

Response: Success
Message: User muted
```

ConfbridgeUnmute

Unmutes a specified user in a specified conference.

Example

```
Action: ConfbridgeUnmute
Conference: 1111
Channel: SIP/mypeer-00000001

Response: Success
Message: User unmuted
```

ConfbridgeKick

Removes a specified user from a specified conference.

Example

```
Action: ConfbridgeKick
Conference: 1111
Channel: SIP/mypeer-00000001

Response: Success
Message: User kicked
```

ConfbridgeLock

Locks a specified conference.

Example

```
Action: ConfbridgeLock
Conference: 1111

Response: Success
Message: Conference locked
```

ConfbridgeUnlock

Unlocks a specified conference.

Example

```
Action: ConfbridgeUnlock
Conference: 1111

Response: Success
Message: Conference unlocked
```

ConfbridgeStartRecord

Starts recording a specified conference, with an optional filename. If recording is already in progress, an error will be returned. If RecordFile is not provided, the default record_file as specified in the conferences Bridge Profile will be used. If record_file is not specified, a file will automatically be generated in Asterisk's monitor directory.

Example

```
Action: ConfbridgeStartRecord
Conference: 1111

Response: Success
Message: Conference Recording Started.

Event: VarSet
Privilege: dialplan,all
Channel: ConfBridgeRecorder/conf-1111-uid-1653801660
Variable: MIXMONITOR_FILENAME
Value: /var/spool/asterisk/monitor/confbridge-1111-1303309869.wav
Uniqueid: 1303309869.6
```

ConfbridgeStopRecord

Stops recording a specified conference.

Example

```
Action: ConfbridgeStopRecord
Conference: 1111

Response: Success
Message: Conference Recording Stopped.

Event: Hangup
Privilege: call,all
Channel: ConfBridgeRecorder/conf-1111-uid-1653801660
Uniqueid: 1303309869.6
CallerIDNum: <unknown>
CallerIDName: <unknown>
Cause: 0
Cause-txt: Unknown
```

ConfbridgeSetSingleVideoSrc

This action sets a conference user as the single video source distributed to all other video-capable participants.

Example

```
Action: ConfbridgeSetSingleVideoSrc
Conference: 1111
Channel: SIP/mypeer-00000001

Response: Success
Message: Conference single video source set.
```

ConfBridge AMI Events

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ConfbridgeStart

This event is sent when the first user requests a conference and it is instantiated

Example

```
Event: ConfbridgeStart
Privilege: call,all
Conference: 1111
```

ConfbridgeJoin

This event is sent when a user joins a conference - either one already in progress or as the first user to join a newly instantiated bridge.

Example

```
Event: ConfbridgeJoin
Privilege: call,all
Channel: SIP/mypeer-00000001
Uniqueid: 1303309562.3
Conference: 1111
CallerIDnum: 1234
CallerIDname: mypeer
```

ConfbridgeLeave

This event is sent when a user leaves a conference.

Example

```
Event: ConfbridgeLeave
Privilege: call,all
Channel: SIP/mypeer-00000001
Uniqueid: 1303308745.0
Conference: 1111
CallerIDnum: 1234
CallerIDname: mypeer
```

ConfbridgeEnd

This event is sent when the last user leaves a conference and it is torn down.

Example

```
Event: ConfbridgeEnd
Privilege: call,all
Conference: 1111
```

ConfbridgeTalking

This event is sent when the conference detects that a user has either begin or stopped talking.

Start talking Example

```
Event: ConfbridgeTalking
Privilege: call, all
Channel: SIP/mypeer-00000001
Uniqueid: 1303308745.0
Conference: 1111
TalkingStatus: on
```

Stop talking Example

```
Event: ConfbridgeTalking
Privilege: call, all
Channel: SIP/mypeer-00000001
Uniqueid: 1303308745.0
Conference: 1111
TalkingStatus: off
```

ConfBridge CLI Commands

ConfBridge CLI Commands

ConfBridge offers several commands that may be invoked from the Asterisk CLI.

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confbridge kick <conference> <channel>

Removes the specified channel from the conference, e.g.:

```
*CLI> confbridge kick 1111 SIP/mypeer-00000000
Kicking SIP/mypeer-00000000 from confbridge 1111
```

confbridge list

Shows a summary listing of all bridges, e.g.:

```
*CLI> confbridge list
Conference Bridge Name           Users  Marked Locked?
================================ ====== ====== ========
1111                                  1      0 unlocked
```

confbridge list <conference>

Shows a detailed listing of participants in a specified conference, e.g.:

```
*CLI> confbridge list 1111
Channel                       User Profile     Bridge Profile   Menu
============================= ================ ================ ================
SIP/mypeer-00000001          default_user     1111             sample_user_menu
```

confbridge lock <conference>

Locks a specified conference so that only Admin users can join, e.g.:

```
*CLI> confbridge lock 1111
Conference 1111 is locked.
```

confbridge unlock <conference>

Unlocks a specified conference so that only Admin users can join, e.g.:

```
*CLI> confbridge unlock 1111
Conference 1111 is unlocked.
```

confbridge mute <conference> <channel>

Mutes a specified user in a specified conference, e.g.:

```
*CLI> confbridge mute 1111 SIP/mypeer-00000001
Muting SIP/mypeer-00000001 from confbridge 1111
```

confbridge unmute <conference> <channel>

Unmutes a specified user in a specified conference, e.g.:

```
*CLI> confbridge unmute 1111 SIP/mypeer-00000001
Unmuting SIP/mypeer-00000001 from confbridge 1111
```

confbridge record start <conference> <file>

Begins recording a conference. If "file" is specified, it will be used, otherwise, the Bridge Profile record_file will be used. If the Bridge Profile does not specify a record_file, one will be automatically generated in Asterisk's monitor directory. Usage:

```
*CLI> confbridge record start 1111
Recording started
*CLI>   == Begin MixMonitor Recording ConfBridgeRecorder/conf-1111-uid-618880445
```

confbridge record stop <confererence>

Stops recording the specified conference, e.g.:

```
*CLI> confbridge record stop 1111
Recording stopped.
*CLI>   == MixMonitor close filestream (mixed)
  == End MixMonitor Recording ConfBridgeRecorder/conf-1111-uid-618880445
```

confbridge show menus

Shows a listing of Conference Menus as defined in confbridge.conf, e.g.:

```
*CLI> confbridge show menus
--------- Menus -----------
sample_admin_menu
sample_user_menu
```

confbridge show menu <menu name>

Shows a detailed listing of a named Conference Menu, e.g.:

```
*CLI> confbridge show menu sample_admin_menu
Name: sample_admin_menu
*9=increase_talking_volume
*8=no_op
*7=decrease_talking_volume
*6=increase_listening_volume
*4=decrease_listening_volume
*3=admin_kick_last
*2=admin_toggle_conference_lock
*1=toggle_mute
*=playback_and_continue(conf-adminmenu)
```

confbridge show profile bridges

Shows a listing of Bridge Profiles as defined in confbridge.conf, e.g.:

```
*CLI> confbridge show profile bridges
--------- Bridge Profiles -----------
1111
default_bridge
```

confbridge show profile bridge <bridge>

Shows a detailed listing of a named Bridge Profile, e.g.:

```
*CLI> confbridge show profile bridge 1111
--------------------------------------------
Name:                 1111
Internal Sample Rate: 16000
Mixing Interval:      10
Record Conference:    no
Record File:          Auto Generated
Max Members:          No Limit
sound_only_person:    conf-onlyperson
sound_has_joined:     conf-hasjoin
sound_has_left:       conf-hasleft
sound_kicked:         conf-kicked
sound_muted:          conf-muted
sound_unmuted:        conf-unmuted
sound_there_are:      conf-thereare
sound_other_in_party: conf-otherinparty
sound_place_into_conference: conf-placeintoconf
sound_wait_for_leader:       conf-waitforleader
sound_get_pin:        conf-getpin
sound_invalid_pin:    conf-invalidpin
sound_locked:         conf-locked
sound_unlocked_now:   conf-unlockednow
sound_lockednow:      conf-lockednow
sound_error_menu:     conf-errormenu
```

confbridge show profile users

Shows a listing of User Profiles as defined in confbridge.conf, e.g.:

```
*CLI> confbridge show profile users
--------- User Profiles -----------
awesomeusers
default_user
```

confbirdge show profile user <user>

Shows a detailed listing of a named Bridge Profile, e.g.:

```
*CLI> confbridge show profile user default_user
--------------------------------------------
Name:                    default_user
Admin:                   false
Marked User:             false
Start Muted:             false
MOH When Empty:          enabled
MOH Class:               default
Quiet:                   disabled
Wait Marked:             disabled
END Marked:              disabled
Drop_silence:            enabled
Silence Threshold:       2500ms
Talking Threshold:       160ms
Denoise:                 disabled
Talk Detect Events:      disabled
DTMF Pass Through:       disabled
PIN:                     None
Announce User Count:     enabled
Announce join/leave:     enabled
Announce User Count all: enabled
```

ConfBridge Configuration

ConfBridge Configuration

On This Page

ConfBridge Profiles and Menus are configured in the confbridge.conf configuration file - normally located at /etc/asterisk/confbridge.conf. The file contains three reserved sections:

- [general]
- [default_bridge]
- [default_user]

The **[general]** section is currently unused, but is reserved for future use.
The **[default_bridge]** section contains all options invoked when ConfBridge is instantiated from the dialplan without a bridge profile argument.
The **[default_user]** section contains all options invoked when ConfBridge is instantiated from the dialplan without a user profile argument.

Each section contains a **type** definition. The type definition determines the function of the section. The three **types** are:

- bridge
- user
- menu

bridge is used to denote Bridge Profile section definitions.
user is used to denote User Profile section definitions.
menu is used to denote Conference Menu section definitions.

All other sections, defined by a section identifier encapsulated in square brackets, are user-definable.

Example

This is an example, using invalid options and functions, of a confbridge.conf configuration file, displaying the organizational layout. The various options and functions are described later in this page.

```
[general]
; comments are preceded by a comma
;
; the general section is blank
;
[default_bridge]
type=bridge
; Bridge Profile options go here
myoption=value
myoption2=othervalue
;
[default_user]
type=user
; User Profile options go here
myoption=value
myoption2=othervalue
;
[sample_menu]
type=menu
; Conferece Menu options go here
DTMF=function
otherDTMF=otherFunction
;
```

Bridge Profile Configuration Options

A Bridge Profile provides the following configuration options:

Option	Values	Description	Notes
type	bridge	Set this to bridge to configure a bridge profile	
max_members	integer; e.g. 50	Limits the number of participants for a single conference to a specific number. By default, conferences have no participant limit. After the limit is reached, the conference will be locked until someone leaves. Admin-level users are exempt from this limit and will still be able to join otherwise-locked, because of limit, conferences.	

record_conference	yes/no	Records the conference call starting when the first user enters the room, and ending when the last user exits the room. The default recorded filename is 'confbridge-<name of conference bridge>-<start time>.wav and the default format is 8kHz signed linear. By default, this option is disabled. This file will be located in the configured monitoring directory as set in asterisk.conf	
record_file	path, e.g. /tmp/myfiles	When record_conference is set to yes, the specific name of the recorded file can be set using this option. Note that since multiple conferences may use the same Bridge profile, this can cause issues, depending on the configuration. It is recommended to only use this option dynamically with the CONFBRIDGE() dialplan function. This allows the recorded name to be specified and a unique name to be chosen. By default, the recorded file is stored in Asterisk's spool/monitory directory, with a unique filename starting with the 'confbridge' prefix.	
internal_sample_rate	auto, 8000, 12000, 16000, 24000, 32000, 44100, 48000, 96000, 192000	Sets the internal native sample rate at which to mix the conference. The "auto" option allows Asterisk to adjust the sample rate to the best quality / performance based on the participant makeup. Numbered values lock the rate to the specified numerical rate. If a defined number does not match an internal sampling rate supported by Asterisk, the nearest sampling rate will be used instead.	
mixing_interval	10, 20, 40, 80	Sets, in milliseconds, the internal mixing interval. By default, the mixing interval of a bridge is 20ms. This setting reflects how "tight" or "loose" the mixing will be for the conference. Lower intervals provide a "tighter" sound with less delay in the bridge and consume more system resources. Higher intervals provide a "looser" sound with more delay in the bridge and consume less resources	
video_mode	none, follow_talker, last_marked, first_marked	Configured video (as opposed to audio) distribution method for conference participants. Participants must use the same video codec. Confbridge does not provide MCU functionality. It does not transcode, scale, transrate, or otherwise manipulate the video. Options are "none," where no video source is set by default and a video source may be later set via AMI or DTMF actions; "follow_talker," where video distrubtion follows whomever is talking and providing video; "last_marked," where the last marked user with video capabilities to join the conference will be the single video source distributed to all other participants - when the current video source leaves, the marked user previous to the last-joined will be used as the video source; and "first-marked," where the first marked user with video capabilities to join the conference will be the single video source distributed to all other participants - when the current video source leaves, the marked user that joined next will be used as the video source. Use of video in conjunction with the jitterbuffer results in the audio being slightly out of sync with the video - because the jitterbuffer only operates on the audio stream, not the video stream. Jitterbuffer should be disabled when video is used.	
sound_join	filename	The sound played to the bridge when a user joins, typically some kind of beep sound	
sound_leave	filename	The sound played to the bridge when a user leaves, also typically some kind of beep sound	
sound_has_joined	filename	The sound played as a user intro, e.g. "xxxx has joined the conference."	
sound_has_left	filename	The sound played as a user parts the conference, e.g. "xxxx has left the conference."	
sound_kicked	filename	The sound played to a user who has been kicked from the conference.	
sound_muted	filename	The sound played to a user when the mute option is toggled on.	
sound_unmuted	filename	The sound played to a user when the mute option is toggled off.	
sound_only_person	filename	The sound played when a user is the only person in the conference.	
sound_only_one	filename	The sound played to a user when there is only one other person in the conference.	
sound_there_are	filename	The sound played when announcing how many users there are in a conference.	
sound_other_in_party	filename	Used in conjunction with the sound_there_are option, used like "sound_there_are" <number of participants> "sound_other_in_party"	
sound_place_into_conference	filename	The sound played when someone is placed into a conference, after waiting for a marked user.	
sound_wait_for_leader	filename	The sound played when a user is placed into a conference that cannot start until a marked user enters.	
sound_leader_has_left	filename	The sound played when the last marked user leaves the conference.	

sound_get_pin	filename	The sound played when prompting for a conference PIN	
sound_invalid_pin	filename	The sound played when an invalid PIN is entered too many (3) times	
sound_locked	filename	The sound played to a user trying to join a locked conference.	
sound_locked_now	filename	The sound played to an Admin-level user after toggling the conference to locked mode.	
sound_unlocked_now	filename	The sound played to an Admin-level user after toggling the conference to unlocked mode.	
sound_error_menu	filename	The sound played when an invalid menu option is entered.	
sound_participants_muted	filename	The sound played when all non-admin participants are muted.	**New in Asterisk 11**
sound_participants_unmuted	filename	The sound played when all non-admin participants are unmuted	**New in Asterisk 11**

Example
In this example, a Bridge Profile called "fancybridge" will be created. It will be configured to allow up to 20 callers, and will be set to mix at 10ms (tight mixing) at an automatic sampling rate. Additionally, it will be recorded.

```
[fancybridge]
type=bridge
max_members=20
mixing_interval=10
internal_sample_rate=auto
record_conference=yes
```

User Profile Configuration Options

A User Profile provides the following configuration options:

Option	Values	Description	Notes
type	user	Set this to user to configure a user profile	
admin	yes/no	Sets if the user is an Admin or not. By default, no.	
marked	yes/no	Sets if the user is Marked or not. By default, no.	
startmuted	yes/no	sets if the user should start out muted. By default, no.	
music_on_hold_when_empty	yes/no	Sets whether music on hold should be played when only one person is in the conference or when the user is waiting on a marked user to enter the conference. By default, off.	
music_on_hold_class	music on hold class	Sets the music on hold class to use for music on hold.	
quiet	yes/no	When set to "yes," enter/leave prompts and user introductions are not played. By default, no.	
announce_user_count	yes/no	Sets if the number of users in the conference should be announced to the caller. By default, no.	
announce_user_count_all	yes/no; or an integer	Sets if the number of users should be announced to all other users in the conference when someone joins. When set to a number, the announcement will only occur once the user count is above the specified number	
announce_only_user	yes/no	Sets if the only user announcement should be played when someone enters an empty conference. By default, yes.	
announcement	filename	If set, the sound file specified by `filename` will be played to the user, and only the user, upon joining the conference bridge.	**New in Asterisk 11**
wait_marked	yes/no	Sets if the user must wait for another marked user to enter before joining the conference. By default, no.	
end_marked	yes/no	If enabled, every user with this option in their profile will be removed from the conference when the last marked user exists the conference.	

dsp_drop_silence	yes/no	Drops what Asterisk detects as silence from entering into the bridge. Enabling this option will drastically improve performance and help remove the buildup of background noise from the conference. This option is highly recommended for large conferences, due to its performance improvements.	
dsp_talking_threshold	integer in milliseconds	The time, in milliseconds, by default 160, of sound above what the DSP has established as base-line silence for a user, before that user is considered to be talking. This value affects several options: 1. Audio is only mixed out of a user's incoming audio stream if talking is detected. If this value is set too loose, the user will hear themselves briefly each time they begin talking until the DSP has time to establish that they are in fact talking. 2. When talker detection AMI events are enabled, this value determines when talking has begun, which causes AMI events to fire. If this value is set too tight, AMI events may be falsely triggered by variants in the background noise of the caller. 3. The drop_silence option depends on this value to determine when the user's audio should be mixed into the bridge after periods of silence. If this value is too loose, the beginning of a user's speech will get cut off as they transition from silence to talking.	
dsp_silence_threshold	integer in milliseconds	The time, in milliseconds, by default 2500, of sound falling within what the DSP has established as the baseline silence, before a user is considered to be silent. The best way to approach this option is to set it slightly above the maximum amount of milliseconds of silence a user may generate during natural speech. This value affects several operations: 1. When talker detection AMI events are enabled, this value determines when the user has stopped talking after a period of talking. If this value is set too low, AMI events indicating that the user has stopped talking may get faslely sent out when the user briefly pauses during mid sentence. 2. The drop_silence option depends on this value to determine when the user's audio should begin to be dropped from the bridge, after the user stops talking. If this value is set too low, the user's audio stream may sound choppy to other participants.	
talk_detection_events	yes/no	Sets whether or not notifications of when a user begins and ends talking should be sent out as events over AMI. By default, no.	
denoise	yes/no	Whether or not a noise reduction filter should be applied to the audio before mixing. By default, off. This requires codec_speex to be built and installed. Do not confuse this option with drop_silence. denoise is useful if there is a lot of background noise for a user, as it attempts to remove the noise while still preserving the speech. This option does not remove silence from being mixed into the conference and does come at the cost of a slight performance hit.	
jitterbuffer	yes/no	Whether or not to place a jitter buffer on the caller's audio stream before any audio mixing is performed. This option is highly recommended, but will add a slight delay to the audio and will incur a slight performance penalty. This option makes use of the JITTERBUFFER dialplan function's default adaptive jitter buffer. For a more fine-tuned jitter buffer, disable this option and use the JITTERBUFFER dialplan function on the calling channel, before it enters the ConfBridge application.	
pin	integer	Sets if the user must enter a PIN before joining the conference. The user will be prompted for the PIN.	
announce_join_leave	yes/no	When enabled, this option prompts the user for their name when entering the conference. After the name is recorded, it will be played as the user enters and exists the conference. By default, no.	
dtmf_passthrough	yes/no	Whether or not DTMF received from users should pass through the conference to other users. By default, no.	

Example
In this example, we will create a user profile called "fancyuser" that includes music on hold, user count announcements, join/leave announcements, silence detection, noise reduction and requires a PIN of 456.

```
[fancyuser]
type=user
music_on_hold_when_empty=yes
music_on_hold_class=default
announce_user_count_all=yes
announce_join_leave=yes
dsp_drop_silence=yes
denoise=yes
pin=456
```

Conference Menu Configuration Options

A Conference Menu provides the following configuration options:

Option	Values	Description	Notes
type	menu	Set this to menu to configure a conference menu	
playback	(<name of audio file1>&<name of audio file2>&...)	Plays back an audio file, or a string of audio files chained together using the & character, to the user and then immediately returns them to the conference.	
playback_and_continue	(<name of audio file 1>&<name of audio file 2>&...)	Plays back an audio file, or a series of audio files chained together using the & character, to the user while continuing the collect the DTMF sequence. This is useful when using a menu prompt that describes all of the menu options. Note that any DTMF during this action will terminate the prompt's playback.	
toggle_mute		Toggles mute on and off. When a user is muted, they will not be able to speak to other conference users, but they can still listen to other users. While muted, DTMF keys from the caller will continue to be collected.	
no_op		This action does nothing. Its only real purpose exists for being able to reserve a sequence in the configuration as a menu exit sequence.	
decrease_listening_volume		Decreases the caller's listening volume. Everything they hear will sound quieter.	
increase_listening_volume		Increases the caller's listening volume. Everything they hear will sound louder.	
reset_listening_volume		Resets the caller's listening volume to the default level.	
decrease_talking_volume		Decreases the caller's talking volume. Everything they say will sound quieter to other callers.	
increase_talking_volume		Increases the caller's talking volume. Everything they say will sound louder to other callers.	
reset_talking_volume		Resets the caller's talking volume to the default level.	
dialplan_exec	(context,exten,priority)	Allows one to escape from the conference and execute commands in the dialplan. Once the dialplan exits, the user will be put back into the conference.	
leave_conference		Allows a user to exit the conference and continue execution in the dialplan.	
admin_kick_last		Allows an Admin to remove the last participant from the conference. This action only works for users whose User Profiles set them as conference Admins.	
admin_toggle_conference_lock		Allows an Admin to toggle locking and unlocking the conference. When the conference is locked, only other Admin users can join. When the conference is unlocked, any user may join up to the limit defined by the max_members Bridge Profile option. This action only works for users whose User Profiles set them as conference Admins.	
admin_toggle_mute_participants		Allows an Admin to mute/unmute all non-admin participants in the conference.	**New in Asterisk 11**
set_as_single_video_src		Allows a user to set themselves as the single video distribution source for all other participants. This overrides the video_mode setting.	
release_as_single_video_src		Allows a user to release themselves as the single video source. Upon release of the video source, and/or if video_mode is set to "none," this action will result in the conference returning to whatever video mode the Bridge Profile is using. This action will have no effect if the user is not currently the video source. The user is also not guaranteed that the use of this action will prevent them from becoming the video source later.	
participant_count		Plays back the current number of participants into the conference.	**New in Asterisk 11**

Example
In this example, we'll create a menu called "fancymenu." This menu will utilize many of the options listed above. We will construct a features menu that plays when the user enters the * character. Since we will do this using the playback_and_continue option, we will define other menu items as being a

"subset" of the * command, e.g. *4, so that once the user presses *, they can listen to the menu options and then press the specific "after-star" option, e.g. 4, to affect the option. Additionally, we will duplicate those same sub-features as non-* features, so that the user does not need to have entered the * menu structure in order to affect the options, they can just press the key, e.g. "4" at any time, regardless of whether or not they're in the *-tree.

```
[fancymenu]
type=menu
*=playback_and_continue(conf-togglemute&press&digits/1&silence/1&conf-leave&press&digits/2&silence/1&add-a-caller&press&digits/3&silence/1&conf-decrease-talking&press&digits/4&silence/1&reset-talking&press&digits/5&silence/1&increase-talking&press&digits/6&silence/1&conf-decrease-listening&press&digits/7&silence/1&conf-reset-listening&press&digits/8&silence/1&conf-increase-listening&press&digits/9&silence/1&conf-exit-menu&press&digits/0)
*1=toggle_mute
1=toggle_mute
*2=leave_conference
2=leave_conference
*3=dialplan_exec(addcallers,1,1)
3=dialplan_exec(addcallers,1,1)
*4=decrease_listening_volume
4=decrease_listening_volume
*5=reset_listening_volume
5=reset_listening_volume
*6=increase_listening_volume
6=increase_listening_volume
*7=decrease_talking_volume
7=decrease_talking_volume
*8=reset_talking_volume
8=reset_talking_volume
*9=increase_talking_volume
9=increase_talking_volume
*0=no_op
0=no_op
```

Of particular note in this example, we're calling the dialplan_exec option. Here, we're specifying "addcaller,1,1." This means that when someone dials 3, Asterisk will escape them out of the bridge momentarily to go execute priority 1 of extension 1 in the addcaller context of the dialplan (extensions.conf). Our dialplan, including the addcaller context, in this case, might look like:

```
[addcaller]
exten => 1,1,Originate(SIP/otherpeer,exten,conferences,100,1)

[conferences]
exten => 100,1,ConfBridge(1234)
```

Thus, when someone dials "3" while in the bridge, they'll Originate a call from the dialplan that puts SIP/otherpeer into the conference. Once the dial has completed, the person that dialed "3" will find themselves back in the bridge, with the other participants.

ConfBridge Functions

Function CONFBRIDGE

The CONFBRIDGE dialplan function is used to set customized Bridge and/or User Profiles on a channel for the ConfBridge application. It uses the same options defined in confbridge.conf and allows the creation of dynamic, dialplan-driven conferences.

On This Page

Syntax

```
CONFBRIDGE(type,option)
```

- type - Refers to which type of profile the option belongs to. Type can be either "bridge" or "user."
- option - Refers to the confbridge.conf option that is to be set dynamically on the channel. This can also refer to an existing Bridge or User Profile by using the keyword "template." In this case, an existing Bridge or User Profile can be appended or modified on-the-fly.

Examples

Example 1
In this example, the custom set User Profile on this channel enables announce_join_leave (so users will be announced as they come and go), sets users to join muted (so that they're not able to speak), and pushes them into bridge "1."

```
exten => 1,1,Answer()
exten => 1,n,Set(CONFBRIDGE(user,announce_join_leave)=yes)
exten => 1,n,Set(CONFBRIDGE(user,startmuted)=yes)
exten => 1,n,ConfBridge(1)
```

Example 2
In this example, we will include an existing User Profile, the default_user User Profile as defined in confbridge.comf, and we will set additional parameters (admin and marked) that aren't already defined in the default_user User Profile.

```
exten => 1,1,Answer()
exten => 1,n,Set(CONFBRIDGE(user,template)=default_user)
exten => 1,n,Set(CONFBRIDGE(user,admin)=yes)
exten => 1,n,Set(CONFBRIDGE(user,marked)=yes)
exten => 1,n,ConfBridge(1)
```

Function CONFBRIDGE_INFO

The CONFBRIDGE_INFO dialplan function is used to retrieve information about a conference, such as locked/unlocked status and the number of parties including admins and marked users.

Syntax

```
CONFBRIDGE_INFO(type,conf)
```

- type - Refers to which information type to be retrieved. Type can be either "parties," "admins," "marked," or "locked."
- conf - Refers to the name of the conference being referenced.

The CONFBRIDGE_INFO function returns a non-negative integer for valid conference identifiers, 0 or 1 for locked, and "" for invalid conference identifiers.

ConfBridge Sound Prompts

The following Conference Menu and Bridge Profile options sound files are available as part of the latest Asterisk core sounds package - currently only available in the English language package.

- confbridge-begin-glorious-a - "The conference will begin when our glorious leader arrives."
- confbridge-begin-glorious-b - "The conference will begin when our **glorious leader** arrives."
- confbridge-begin-glorious-c - "The conference will begin when our **glorious leader arrives.**"
- confbridge-conf-begin - "The conference will now begin."
- confbridge-conf-end - "The conference has ended."
- confbridge-dec-list-vol-in - "To decrease the audio volume from other participants..."
- confbridge-dec-list-vol-out - "...to decrease the audio volume from other participants."
- confbridge-dec-talk-vol-in - "To decrease your speaking volume to other participants..."
- confbridge-dec-talk-vol-out - "...to decrease your speaking volume to other participants."
- confbridge-has-joined - "...has joined the conference."
- confbridge-has-left - "...has left the conference."
- confbridge-inc-list-vol-in - "To increase the audio volume from other participants..."
- confbridge-inc-list-vol-out - "...to increase the audio volume from other participants."
- confbridge-inc-talk-vol-in - "To increase your speaking volume to other participants..."
- confbridge-inc-talk-vol-out - "...to increase your speaking volume to other participants."
- confbridge-invalid - "You have entered an invalid option."
- confbridge-leave-in - "To leave the conference..."
- confbridge-leave-out - "...to leave the conference."
- confbridge-lock-extended - "...to lock or unlock the conference. When a conference is locked, only conference administrators can join."
- confbridge-lock-in - "To lock or unlock the conference."
- confbridge-lock-no-join - "The conference is currently locked and cannot be joined."
- confbridge-lock-out 0- "...to lock or unlock the conference."
- confbridge-locked - "The conference is now locked."
- confbridge-menu-exit-in - "To exit the menu..."
- confbridge-menu-exit-out - "...to exit the menu."
- confbridge-mute-extended - "...to mute or unmute yourself. When you are muted, you cannot send audio to other participants; however you will still hear audio from other unmuted participants.
- confbridge-mute-in - "To mute or unmute yourself..."
- confbridge-mute-out - "...to mute or unmute yourself."
- confbridge-muted - "You are now muted."
- confbridge-only-one - "There is currently one other participant in the conference."
- confbridge-only-participant - "You are currently the only participant in the conference."
- confbridge-participants - "...participants in the conference."
- confbridge-pin-bad - "You have entered too many invalid personal identification numbers."
- confbridge-pin - "Please enter your personal identification number followed by the pound or hash key."
- confbridge-remove-last-in - "To remove the participant who most recently joined the conference..."
- confbridge-remove-last-out - "...to remove the participant who most recently joined the conference."
- confbridge-removed - "You have been removed from the conference."
- confbridge-rest-list-vol-in - "To reset the audio volume of the conference to the default level..."
- confbridge-rest-list-vol-out - "...to reset the audio volume of the conference to the default level."
- confbridge-rest-talk-vol-in - "To reset your speaking volume to the default level..."
- confbridge-rest-talk-vol-out - "...to reset your speaking volume to the default level."
- confbridge-there-are - "There are currently..."
- confbridge-unlocked - "The conference is now unlocked."
- confbridge-unmuted - "You are no longer muted."

Dial Application

Overview of the Dial Application

The Dial application is probably the most well known and crucial Asterisk application. Asterisk is often used to interface between communication devices and technologies, and Dial is a simple way to establish a connection from the dialplan. When a channel executes Dial then Asterisk will attempt to contact or "dial" all devices passed to the application. If an answer is received then the two channels will be bridged. Dial provides many options to control behavior and will return results and status of the dial operation on a few channel variables.

Using the Dial application

Here is a few ways to learn the usage of the Dial application.

- Read the detailed documentation for your Asterisk version: e.g. for Asterisk 13 - Asterisk 13 Application_Dial
- See how to use Dial for a specific channel driver: Dialing PJSIP Channels
- Note how Dial is used in a simple Asterisk dialplan: Creating Dialplan Extensions

See Also

- Pre-Dial Handlers
- Hangup Handlers

Directory Application

The next application we'll cover is named **Directory()**, because it presents the callers with a dial-by-name directory. It asks the caller to enter the first few digits of the person's name, and then attempts to find matching names in the specified voice mail context in **voicemail.conf**. If the matching mailboxes have a recorded name greeting, Asterisk will play that greeting. Otherwise, Asterisk will spell out the person's name letter by letter.

```
Directory([voicemail_context,[dialplan_context,[options]]])
```

The **Directory()** application takes three parameters:

voicemail_context

This is the context within **voicemail.conf** in which to search for a matching directory entry. If not specified , the **[docs:default]** context will be searched.

dialplan_context

When the caller finds the directory entry they are looking for, Asterisk will dial the extension matching their mailbox in this context.

options

A set of options for controlling the dial-by-name directory. Common options include **f** for searching based on first name instead of last name and **e** to read the extension number as well as the name.

✓ **Directory() Options**
To see the complete list of options for the Directory() application, type **core show application Directory** at the Asterisk CLI.

Let's add a dial-by-name directory to our dialplan. Simply add this line to your **[docs:users]** context in **extensions.conf**:

```
exten => 6501,1,Directory(vm-demo,users,ef)
```

Now you should be able to dial extension **6501** to test your dial-by-name directory.

Early Media and the Progress Application

Many dialplan applications within Asterisk support a common VOIP feature known as early media. Early Media is most frequently associated with the SIP channel, but it is also a feature of other channel drivers such as H323. In simple situations, any call in Asterisk that is going to involve audio should invoke either Progress() or Answer().

By making use of the progress application, a phone call can be made to play audio before answering a call or even without ever even intending to answer the full call.

Simple Example involving playback:

```
exten => 500,1,Progress()
exten => 500,n,Wait(1)
exten => 500,n,Playback(WeAreClosedGoAway,noanswer)
exten => 500,n,Hangup()
```

In the example above, we start an early media call which waits for a second and then plays a rather rudely named message indicating that the requested service has closed for whatever reason before hanging up. It is worth observing that the Playback application is called with the 'noanswer' argument. Without that argument, Playback would automatically answer the call and then we would no longer be in early media mode.

Strictly speaking, Asterisk will send audio via RTP to any device that calls in regardless of whether Asterisk ever answers or progresses the call. It is possible to make early media calls to some devices without ever sending the progress message, however this is improper and can lead to a myriad of nasty issues that vary from device to device. For instance, in internal testing, there was a problem reported against the Queue application involving the following extension:

```
exten => 500,1,Queue(queuename)
```

This is certainly a brief example. The queue application does not perform any sort of automatic answering, so at this point Asterisk will be sending the phone audio packets, but it will not have formally answered the call or have sent a progress indication. At this point, different phones will behave differently. In the case of the internal test, our Polycom Soundpoint IP 330 phone played nothing while our SNOM360 phone played audio until approximately one minute into the call before it started ceaselessly generating a ring-back indication. There is nothing wrong with either of these phones... they are simply reacting to an oddly formed SIP dialog. Obviously though, neither of these is ideal for a queue and the problem wouldn't have existed had Queue been started after using the Progress application like below:

```
exten => 500,1,Progress()
exten => 500,n,Queue(queuename)
```

Getting the hang of when to use Progress and Answer can be a little tricky, and it varies greatly from application to application. If you want to be safe, you can always just answer the calls and keep things simple, but there are a number of use cases where it is more appropriate to use early media, and most people who actually need this feature will probably be aware of when it is necessary.

Applications which can use early media and do not automatically answer (incomplete list, please contribute):
SayAlpha/SayDigits/SayNumber/etc
Playback (conditionally)
MP3
MixMonitor
MorseCode
Echo
Queue
MusicOnHold

External IVR Interface

Asterisk External IVR Interface

If you load `app_externalivr.so` in your Asterisk instance, you will have an `ExternalIVR` application available in your dialplan. This application implements a simple protocol for bidirectional communication with an external process, while simultaneously playing audio files to the connected channel (without interruption or blocking).

There are two ways to use `ExternalIVR`; you can execute an application on the local system or you can establish a socket connection to a TCP/IP socket server.

To execute a local application use the form:

```
ExternalIVR(/full/path/to/applcation[(arguments)],options)
```

The arguments are optional, however if they exist they must be enclosed in parentheses. The external application will be executed in a child process, with its standard file handles connected to the Asterisk process as follows:

- `stdin` (0) - Events will be received on this handle
- `stdout` (1) - Commands can be sent on this handle
- `stderr` (2) - Messages can be sent on this handle

Use of `stderr` for message communication is discouraged because it is not supported by a socket connection.

To create a socket connection use the form:

```
ExternalIVR(ivr://host[:port][(arguments)],options)
```

The host can be a fully qualified domain name or an IP address (both IPv4 and IPv6 are supported). The port is optional and, if not specified, is `2949` by default. The `ExternalIVR` application will connect to the specified socket server and establish a bidirectional socket connection, where events will be sent to the TCP/IP server and commands received from it.

The specific `ExternalIVR` options, #events and #commands are detailed below.

Upon execution, if not specifically prevented by an option, the `ExternalIVR` application will answer the channel (if it's not already answered), create an audio generator, and start playing silence. When your application wants to send audio to the channel, it can send a command to add a file to the generator's playlist. The generator will then work its way through the list, playing each file in turn until it either runs out of files to play, the channel is hung up, or a command is received to clear the list and start with a new file. At any time, more files can be added to the list and the generator will play them in sequence.

While the generator is playing audio (or silence), any DTMF #events received on the channel will be sent to the child process. Note that this can happen at any time, since the generator, the child process and the channel thread are all executing independently. It is very important that your external application be ready to receive events from Asterisk at all times (without blocking), or you could cause the channel to become non-responsive.

If the child process dies, or the remote server disconnects, `ExternalIVR` will notice this and hang up the channel immediately (and also send a message to the log).

`ExternalIVR` Options

- `n` - 'n'oanswer, don't answer an otherwise unanswered channel.
- `i` - 'i'gnore_hangup, instead of sending an `H` event and exiting `ExternalIVR` upon channel hangup, it instead sends an `I` event and expects the external application to exit the process.
- `d` - 'd'ead, allows the operation of `ExternalIVR` on channels that have already been hung up.

Events

All events are be newline-terminated strings and are sent in the following format:

```
tag,timestamp[,data]
```

The tag can be one of the following characters:

- `0-9` - DTMF event for keys 0 through 9
- `A-D` - DTMF event for keys A through D
- `*` - DTMF event for key *

- `#` - DTMF event for key `#`
- `H` - The channel was hung up by the connected party
- `E` - The script requested an exit
- `Z` - The previous command was unable to be executed. There may be a data element if appropriate, see specific commands below for details
- `T` - The play list was interrupted (see `S` command)
- `D` - A file was dropped from the play list due to interruption (the data element will be the dropped file name) NOTE: this tag conflicts with the `D` DTMF event tag. The existence of the data element is used to differentiate between the two cases
- `F` - A file has finished playing (the data element will be the file name)
- `P` - A response to the `P` command
- `G` - A response to the `G` command
- `I` - A Inform message, meant to "inform" the client that something has occurred. (see Inform Messages below)

The timestamp will be a decimal representation of the standard Unix epoch-based timestamp, e.g., `284654100`.

Commands

All commands are newline-terminated (`\n`) strings.

The child process can send one of the following commands:

- `S,filename`
- `A,filename`
- `I,TIMESTAMP`
- `H,message`
- `E,message`
- `O,option`
- `V,name=value[,name=value[,name=value]]`
- `G,name[,name[,name]]`
- `L,log_message`
- `P,TIMESTAMP`
- `T,TIMESTAMP`

The `S` command checks to see if there is a playable audio file with the specified name, and if so, clears the generator's playlist and places the file onto the list. Note that the playability check does not take into account transcoding requirements, so it is possible for the file to not be played even though it was found. If the file does not exist it sends a `Z` response with the data element set to the file requested. If the generator is not currently playing silence, then `T` and `D` events will be sent to signal the playlist interruption and notify it of the files that will not be played.

The `A` command checks to see if there is a playable audio file with the specified name, and if so, appends it to the generator's playlist. The same playability and exception rules apply as for the `S` command.

The `I` command stops any audio currently playing and clears the generator's playlist. The `I` command was added in Asterisk 11.

The `E` command logs the supplied message to the Asterisk log, stops the generator and terminates the `ExternalIVR` application, but continues execution in the dialplan.

The `H` command logs the supplied message to the Asterisk log, stops the generator, hangs up the channel and terminates the ExternalIVR application.

The `O` command allows the child to set/clear options in the ExternalIVR() application. The supported options are:

- `(no)autoclear` - Automatically interrupt and clear the playlist upon reception of DTMF input.

The `T` command will answer an unanswered channel. If it fails either answering the channel or starting the generator it sends a `Z` response of `Z,TIMESTAMP,ANSWER_FAILED` or `Z,TIMESTAMP,GENERATOR_FAILED` respectively.

The `V` command sets the specified channel variable(s) to the specified value(s).

The `G` command gets the specified channel variable(s). Multiple variables are separated by commas. Response is in `name=value` format.

The `P` command gets the parameters passed into `ExternalIVR` minus the options to `ExternalIVR` itself:

If `ExternalIVR` is executed as:

```
ExternalIVR(/usr/bin/foo(arg1,arg2),n)
```

The response to the `P` command would be:

```
P,TIMESTAMP,/usr/bin/foo,arg1,arg2
```

This is the only way for a TCP/IP server to be able to get retrieve the arguments.

The L command puts a message into the Asterisk log.

This is preferred to using stderr and is the only way for a TCP/IP server to log a message.

Inform Messages

The only inform message that currently exists is a HANGUP message, in the form I,TIMESTAMP,HANGUP and is used to inform of a hangup when the i option is specified.

Errors

Any newline-terminated (\n) output generated by the child process on its stderr handle will be copied into the Asterisk log.

MacroExclusive

About the MacroExclusive application

By: Steve Davies <steve@connection-telecom.com

The MacroExclusive application was added to solve the problem of synchronization between calls running at the same time.

This is usually an issue when you have calls manipulating global variables or the Asterisk database, but may be useful elsewhere.

Consider this example macro, intended to return a "next" number - each caller is intended to get a different number:

```
[macro-next]
exten => s,1,Set(RESULT=${COUNT})
exten => s,n,SetGlobalVar(COUNT=$[${COUNT} + 1])
```

The problem is that in a box with high activity, you can be sure that two calls will come along together - both will get the same "RESULT", or the "COUNT" value will get mangled.

Calling this Macro via MacroExclusive will use a mutex to make sure that only one call executes in the Macro at a time. This ensures that the two lines execute as a unit.

Note that even the s,2 line above has its own race problem. Two calls running that line at once will step on each other and the count will end up as +1 rather than +2.

I've also been able to use MacroExclusive where I have two Macros that need to be mutually exclusive.

Here's the example:

```
[macro-push]
; push value ${ARG2} onto stack ${ARG1}
exten => s,1,Set(DB(STACK/${ARG1})=${ARG2}^${DB(STACK/${ARG1})})

[macro-pop]
; pop top value from stack ${ARG1}
exten => s,1,Set(RESULT=${DB(STACK/${ARG1})})
exten => s,n,Set(DB(STACK/${ARG1})=${CUT(RESULT,^,2)})
exten => s,n,Set(RESULT=${CUT(RESULT,^,1)})
```

All that futzing with the STACK/${ARG1} in the astdb needs protecting if this is to work. But neither push nor pop can run together.

So add this "pattern":

```
[macro-stack]
exten => Macro(${ARG1},${ARG2},${ARG3})
```

... and use it like so:

```
exten => s,1,MacroExclusive(stack,push,MYSTACK,bananas)
exten => s,n,MacroExclusive(stack,push,MYSTACK,apples)
exten => s,n,MacroExclusive(stack,push,MYSTACK,guavas)
exten => s,n,MacroExclusive(stack,push,MYSTACK,pawpaws)
exten => s,n,MacroExclusive(stack,pop,MYSTACK) ; RESULT gets pawpaws (yum)
exten => s,n,MacroExclusive(stack,pop,MYSTACK) ; RESULT gets guavas
exten => s,n,MacroExclusive(stack,pop,MYSTACK) ; RESULT gets apples
exten => s,n,MacroExclusive(stack,pop,MYSTACK) ; RESULT gets bananas
```

We get to the push and pop macros "via" the stack macro. But only one call can execute the stack macro at a time; ergo, only one of push OR pop can run at a time.

Hope people find this useful.

Lastly, its worth pointing out that only Macros that access shared data will require this MacroExclusive protection. And Macro's that you call with macroExclusive should run quickly or you will clog up your Asterisk system.

Asterisk Queues

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.

Building Queues

Building Queues

Written by: Leif Madsen
Initial version: 2010-01-14

> Note that this documentation is based on Asterisk 1.6.2, and this is just one approach to creating queues and the dialplan logic. You may create a better way, and in that case, I would encourage you to submit it to the Asterisk issue tracker at http://issues.asterisk.org for inclusion in Asterisk.

In this article, we'll look at setting up a pair of queues in Asterisk called 'sales' and 'support'. These queues can be logged into by queue members, and those members will also have the ability to pause and unpause themselves.

All configuration will be done in flat files on the system in order to maintain simplicity in configuration.

Adding SIP Devices to Your Server

The first thing we want to do is register a couple of SIP devices to our server. These devices will be our agents that can login and out of the queues we'll create later. Our naming convention will be to use MAC addresses as we want to abstract the concepts of user (agent), device, and extension from each other.

In sip.conf, we add the following to the bottom of our file:

On This Page

```
sip.conf
--------

[std-device](!)
type=peer
context=devices
host=dynamic
secret=s3CuR#p@s5
dtmfmode=rfc2833
disallow=all
allow=ulaw

[0004f2040001](std-device)

[0004f2040002](std-device)
```

What we're doing here is creating a [std-device] template and applying it to a pair of peers that we'll register as 0004f2040001 and 0004f2040002; our devices.

Then our devices can register to Asterisk. In my case I have a hard phone and a soft phone registered. I can verify their connectivity by running 'sip show peers'.

```
*CLI> sip show peers
Name/username              Host            Dyn Nat ACL Port     Status
0004f2040001/0004f2040001  192.168.128.145  D          5060     Unmonitored
0004f2040002/0004f2040002  192.168.128.126  D          5060     Unmonitored
2 sip peers [Monitored: 0 online, 0 offline Unmonitored: 2 online, 0 offline]
```

Configuring Device State

Next, we need to configure our system to track the state of the devices. We do this by defining a 'hint' in the dialplan which creates the ability for a device subscription to be retained in memory. By default we can see there are no hints registered in our system by running the 'core show hints' command.

```
*CLI> core show hints
There are no registered dialplan hint
```

We need to add the devices we're going to track to the extensions.conf file under the [default] context which is the default configuration in sip.conf, however we can change this to any context we want with the 'subscribecontext'
option.

Add the following lines to extensions.conf:

```
[default]
exten => 0004f2040001,hint,SIP/0004f2040001
exten => 0004f2040002,hint,SIP/0004f2040002
```

Then perform a 'dialplan reload' in order to reload the dialplan.

After reloading our dialplan, you can see the status of the devices with 'core show hints' again.

```
*CLI> core show hints

    -= Registered Asterisk Dial Plan Hints =-
           0004f2040002@default             : SIP/0004f2040002      State:Idle
Watchers  0
           0004f2040001@default             : SIP/0004f2040001      State:Idle
Watchers  0
----------------
- 2 hints registered
```

At this point, create an extension that you can dial that will play a prompt that is long enough for you to go back to the Asterisk console to check the state of your device while it is in use.

To do this, add the 555 extension to the [devices] context and make it playback the tt-monkeys file.

```
extensions.conf
---------------

[devices]
exten => 555,1,Playback(tt-monkeys)
```

Dial that extension and then check the state of your device on the console.

```
*CLI>   == Using SIP RTP CoS mark 5
    -- Executing [555@devices:1] Playback("SIP/0004f2040001-00000001", "tt-monkeys") in
new stack
    -- <SIP/0004f2040001-00000001> Playing 'tt-monkeys.slin' (language 'en')

*CLI> core show hints

    -= Registered Asterisk Dial Plan Hints =-
           0004f2040002@default             : SIP/0004f2040002      State:Idle
Watchers  0
           0004f2040001@default             : SIP/0004f2040001      State:Idle
Watchers  0
----------------
- 2 hints registered
```

Aha, we're not getting the device state correctly. There must be something else we need to configure.

In sip.conf, we need to enable 'callcounter' in order to activate the ability for Asterisk to monitor whether the device is in use or not. In versions prior to 1.6.0 we needed to use 'call-limit' for this functionality, but call-limit is now deprecated and is no longer necessary.

So, in sip.conf, in our [std-device] template, we need to add the callcounter option.

```
sip.conf
--------

[std-device](!)
type=peer
context=devices
host=dynamic
secret=s3CuR#p@s5
dtmfmode=rfc2833
disallow=all
allow=ulaw
callcounter=yes     ; <-- add this
```

Then reload chan_sip with 'sip reload' and perform our 555 test again. Dial 555 and then check the device state with 'core show hints'.

```
*CLI>   == Using SIP RTP CoS mark 5
    -- Executing [555@devices:1] Playback("SIP/0004f2040001-00000002", "tt-monkeys") in
new stack
    -- <SIP/0004f2040001-00000002> Playing 'tt-monkeys.slin' (language 'en')

*CLI> core show hints

    -= Registered Asterisk Dial Plan Hints =-
           0004f2040002@default             : SIP/0004f2040002      State:Idle
Watchers  0
           0004f2040001@default             : SIP/0004f2040001      State:InUse
Watchers  0
----------------
- 2 hints registered
```

Note that now we have the correct device state when extension 555 is dialed, showing that our device is InUse after dialing extension 555. This is important when creating queues, otherwise our queue members would get multiple calls from the queues.

Adding Queues to Asterisk

The next step is to add a couple of queues to Asterisk that we can assign queue members into. For now we'll work with two queues; sales and support. Lets create those queues now in queues.conf.

We'll leave the default settings that are shipped with queues.conf.sample in the [general] section of queues.conf. See the queues.conf.sample file for more information about each of the available options.

```
queues.conf
--------

[general]
persistentmembers=yes
autofill=yes
monitor-type=MixMonitor
shared_lastcall=no
```

We can then define a [queue_template] that we'll assign to each of the queues we create. These definitions can be overridden by each queue individually if you reassign them under the [sales] or [support] headers. So under the [general]
section of your queues.conf file, add the following.

```
queues.conf
----------

[queue_template](!)
musicclass=default      ; play [default] music
strategy=rrmemory       ; use the Round Robin Memory strategy
joinempty=yes           ; join the queue when no members available
leavewhenempty=no       ; don't leave the queue no members available
ringinuse=no            ; don't ring members when already InUse

[sales](queue_template)
; Sales queue

[support](queue_template)
; Support queue
```

After defining our queues, lets reload our app_queue.so module.

```
*CLI> module reload app_queue.so
    -- Reloading module 'app_queue.so' (True Call Queueing)

  == Parsing '/etc/asterisk/queues.conf':   == Found
```

Then verify our queues loaded with 'queue show'.

```
*CLI> queue show
support      has 0 calls (max unlimited) in 'rrmemory' strategy (0s holdtime, 0s
talktime), W:0, C:0, A:0, SL:0.0% within 0s
   No Members
   No Callers

sales        has 0 calls (max unlimited) in 'rrmemory' strategy (0s holdtime, 0s
talktime), W:0, C:0, A:0, SL:0.0% within 0s
   No Members
   No Callers
```

Adding Queue Members

You'll notice that we have no queue members available to take calls from the queues. We can add queue members from the Asterisk CLI with the 'queue add member' command.

This is the format of the 'queue add member' command:

```
Usage: queue add member <channel> to <queue> [[[penalty <penalty>] as <membername>]
state_interface <interface>]
       Add a channel to a queue with optionally:  a penalty, membername and a
state_interface
```

The penalty, membername, and state_interface are all optional values. Special attention should be brought to the 'state_interface' option for a member though. The reason for state_interface is that if you're using a channel that does not have device state itself (for example, if you were using the Local channel to deliver a call to an end point) then you could assign the device state of a SIP device to the pseudo channel. This allows the state of a SIP device to be applied to the Local channel for correct device state information.

Lets add our device located at SIP/0004f2040001

```
*CLI> queue add member SIP/0004f2040001 to sales
Added interface 'SIP/0004f2040001' to queue 'sales'
```

Then lets verify our member was indeed added.

```
*CLI> queue show sales
sales        has 0 calls (max unlimited) in 'rrmemory' strategy (0s holdtime, 0s
talktime), W:0, C:0, A:0, SL:0.0% within 0s
   Members:
      SIP/0004f2040001 (dynamic) (Not in use) has taken no calls yet
   No Callers
```

Now, if we dial our 555 extension, we should see that our member becomes InUse within the queue.

```
*CLI>   == Using SIP RTP CoS mark 5
    -- Executing [555@devices:1] Playback("SIP/0004f2040001-00000001", "tt-monkeys") in
new stack
    -- <SIP/0004f2040001-00000001> Playing 'tt-monkeys.slin' (language 'en')


*CLI> queue show sales
sales        has 0 calls (max unlimited) in 'rrmemory' strategy (0s holdtime, 0s
talktime), W:0, C:0, A:0, SL:0.0% within 0s
   Members:
      SIP/0004f2040001 (dynamic) (In use) has taken no calls yet
   No Callers
```

We can also remove our members from the queue using the 'queue remove' CLI command.

```
*CLI> queue remove member SIP/0004f2040001 from sales
Removed interface 'SIP/0004f2040001' from queue 'sales'
```

Because we don't want to have to add queue members manually from the CLI, we should create a method that allows queue members to login and out from their devices. We'll do that in the next section.

But first, lets add an extension to our dialplan in order to permit people to dial into our queues so calls can be delivered to our queue members.

```
extensions.conf
---------------

[devices]
exten => 555,1,Playback(tt-monkeys)

exten => 100,1,Queue(sales)

exten => 101,1,Queue(support)
```

Then reload the dialplan, and try calling extension 100 from SIP/0004f2040002, which is the device we have not logged into the queue.

```
*CLI> dialplan reload
```

And now we call the queue at extension 100 which will ring our device at SIP/0004f2040001.

```
*CLI>   == Using SIP RTP CoS mark 5
    -- Executing [100@devices:1] Queue("SIP/0004f2040002-00000005", "sales") in new stack
    -- Started music on hold, class 'default', on SIP/0004f2040002-00000005
  == Using SIP RTP CoS mark 5
    -- SIP/0004f2040001-00000006 is ringing
```

We can see the device state has changed to Ringing while the device is ringing.

```
*CLI> queue show sales
sales        has 1 calls (max unlimited) in 'rrmemory' strategy (2s holdtime, 3s
talktime), W:0, C:1, A:1, SL:0.0% within 0s
   Members:
      SIP/0004f2040001 (dynamic) (Ringing) has taken 1 calls (last was 14 secs ago)
   Callers:
      1. SIP/0004f2040002-00000005 (wait: 0:03, prio: 0)
```

Our queue member then answers the phone.

```
*CLI>     -- SIP/0004f2040001-00000006 answered SIP/0004f2040002-00000005
    -- Stopped music on hold on SIP/0004f2040002-00000005
    -- Native bridging SIP/0004f2040002-00000005 and SIP/0004f2040001-00000006
```

And we can see the queue member is now in use.

```
*CLI> queue show sales
sales        has 0 calls (max unlimited) in 'rrmemory' strategy (3s holdtime, 3s
talktime), W:0, C:1, A:1, SL:0.0% within 0s
   Members:
      SIP/0004f2040001 (dynamic) (In use) has taken 1 calls (last was 22 secs ago)
   No Callers
```

Then the call is hung up.

```
*CLI>   == Spawn extension (devices, 100, 1) exited non-zero on
'SIP/0004f2040002-00000005'
```

And we see that our queue member is available to take another call.

```
*CLI> queue show sales
sales        has 0 calls (max unlimited) in 'rrmemory' strategy (3s holdtime, 4s
talktime), W:0, C:2, A:1, SL:0.0% within 0s
   Members:
      SIP/0004f2040001 (dynamic) (Not in use) has taken 2 calls (last was 6 secs ago)
   No Callers
```

Logging In and Out of Queues

In this section we'll show how to use the AddQueueMember() and RemoveQueueMember() dialplan applications to login and out of queues. For more information about the available options to AddQueueMember() and RemoveQueueMember() use the 'core show application <app>' command from the CLI.

The following bit of dialplan is a bit long, but stick with it, and you'll see that it isn't really all that bad. The gist of the dialplan is that it will check to see if the active user (the device that is dialing the extension) is currently logged into the queue extension that has been requested, and if logged in, then will log them out; if not logged in, then they will be logged into the queue.

We've updated the two lines we added in the previous section that allowed us to dial the sales and support queues. We've abstracted this out a bit in order to make it easier to add new queues in the future. This is done by adding the queue names to a global variable, then utilizing the extension number dialed to look up the queue name.

So we replace extension 100 and 101 with the following dialplan.

```
; Call any of the queues we've defined in the [globals] section.
exten => _1XX,1,Verbose(2,Call queue as configured in the QUEUE_${EXTEN} global variable)
exten => _1XX,n,Set(thisQueue=${GLOBAL(QUEUE_${EXTEN})})
exten => _1XX,n,GotoIf($["${thisQueue}" = ""]?invalid_queue,1)
exten => _1XX,n,Verbose(2, --> Entering the ${thisQueue} queue)
exten => _1XX,n,Queue(${thisQueue})
exten => _1XX,n,Hangup()

exten => invalid_queue,1,Verbose(2,Attempted to enter invalid queue)
exten => invalid_queue,n,Playback(silence/1&invalid)
exten => invalid_queue,n,Hangup()
```

The globals section contains the following two global variables.

```
[globals]
QUEUE_100=sales
QUEUE_101=support
```

So when we dial extension 100, it matches our pattern _1XX. The number we dialed (100) is then retrievable via ${EXTEN} and we can get the name of queue 100 (sales) from the global variable QUEUE_100. We then assign it to the channel variable thisQueue so it is easier to work with in our dialplan.

```
exten => _1XX,n,Set(thisQueue=${GLOBAL(QUEUE_${EXTEN})})
```

We then check to see if we've gotten a value back from the global variable which would indicate whether the queue was valid or not.

```
exten => _1XX,n,GotoIf($["${thisQueue}" = ""]?invalid_queue,1)
```

If ${thisQueue} returns nothing, then we Goto the invalid_queue extension and playback the 'invalid' file.

We could alternatively limit our pattern match to only extension 100 and 101 with the _10[0-1] pattern instead.

Lets move into the nitty-gritty section and show how we can login and logout our devices to the pair of queues we've created.

First, we create a pattern match that takes star ⭐ plus the queue number that we want to login or logout of. So to login/out of the sales queue (100) we would dial *100. We use the same extension for logging in and out.

```
; Extension *100 or *101 will login/logout a queue member from sales or support queues
respectively.
exten => _*10[0-1],1,Set(xtn=${EXTEN:1})                        ; save ${EXTEN} with *
chopped off to ${xtn}
exten => _*10[0-1],n,Goto(queueLoginLogout,member_check,1)      ; check if already logged
into a queue
```

We save the value of ${EXTEN:1} to the 'xtn' channel variable so we don't need to keep typing the complicated pattern match.

Now we move into the meat of our login/out dialplan inside the [queueLoginLogout] context.

The first section is initializing some variables that we need throughout the member_check extension such as the name of the queue, the members currently logged into the queue, and the current device peer name (i.e. SIP/0004f2040001).

```
; ### Login or Logout a Queue Member
[queueLoginLogout]
exten => member_check,1,Verbose(2,Logging queue member in or out of the request queue)
exten => member_check,n,Set(thisQueue=${GLOBAL(QUEUE_${xtn})})                  ; assign
queue name to a variable
exten => member_check,n,Set(queueMembers=${QUEUE_MEMBER_LIST(${thisQueue})})    ; assign
list of logged in members of thisQueue to
                                                                                ; a
variable (comma separated)
exten => member_check,n,Set(thisActiveMember=SIP/${CHANNEL(peername)})          ;
initialize 'thisActiveMember' as current device

exten => member_check,n,GotoIf($["${queueMembers}" = ""]?q_login,1)             ; short
circuit to logging in if we don't have
                                                                                ; any
members logged into this queue
```

At this point if there are no members currently logged into our sales queue, we then short-circuit our dialplan to go to the 'q_login' extension since there is no point in wasting cycles searching to see if we're already logged in.

The next step is to finish initializing some values we need within the While() loop that we'll use to check if we're already logged into the queue. We set our ${field} variable to 1, which will be used as the field number offset in the CUT() function.

```
; Initialize some values we'll use in the While() loop
exten => member_check,n,Set(field=1)                                            ; start
our field counter at one
exten => member_check,n,Set(logged_in=0)                                        ;
initialize 'logged_in' to "not logged in"
exten => member_check,n,Set(thisQueueMember=${CUT(queueMembers,\,,${field})})   ;
initialize 'thisQueueMember' with the value in the
                                                                                ;   first
field of the comma-separated list
```

Now we get to enter our While() loop to determine if we're already logged in.

```
; Enter our loop to check if our member is already logged into this queue
exten => member_check,n,While($[${EXISTS(${thisQueueMember})}])
; while we have a queue member...
```

This is where we check to see if the member at this position of the list is the same as the device we're calling from. If it doesn't match, then we go to the 'check_next' priority label (where we increase our ${field} counter variable). If it does match, then we continue on in the dialplan.

```
exten => member_check,n,GotoIf($["${thisQueueMember}" !=
"${thisActiveMember}"]?check_next)     ; if 'thisQueueMember' is not the

;   same as our active peer, then

;   check the next in the list of

;   logged in queue members
```

If we continued on in the dialplan, then we set the ${logged_in} channel variable to '1' which represents we're already logged into this queue. We then exit the While() loop with the ExitWhile() dialplan application.

```
exten => member_check,n,Set(logged_in=1)
; if we got here, set as logged in
exten => member_check,n,ExitWhile()
;  then exit our loop
```

If we didn't match this peer name in the list, then we increase our ${field} counter variable by one, update the ${thisQueueMember} channel variable and then move back to the top of the loop for another round of checks.

```
exten => member_check,n(check_next),Set(field=$[${field} + 1])
; if we got here, increase counter
exten => member_check,n,Set(thisQueueMember=${CUT(queueMembers,\,,${field})})
; get next member in the list
exten => member_check,n,EndWhile()
; ...end of our loop
```

And once we exit our loop, we determine whether we need to log our device in or out of the queue.

```
; if not logged in, then login to this queue, otherwise, logout
exten => member_check,n,GotoIf($[${logged_in} = 0]?q_login,1:q_logout,1)        ; if not
logged in, then login, otherwise, logout
```

The following two extensions are used to either log the device in or out of the queue. We use the AddQueueMember() and RemovQueueMember() applications to login or logout the device from the queue.

The first two arguments for AddQueueMember() and RemoveQueueMember() are 'queue' and 'device'. There are additional arguments we can pass, and you can check those out with 'core show application AddQueueMember' and 'core show application RemoveQueueMember()'.

```
; ### Login queue member ###
exten => q_login,1,Verbose(2,Logging ${thisActiveMember} into the ${thisQueue} queue)
exten => q_login,n,AddQueueMember(${thisQueue},${thisActiveMember})                     ;
login our active device to the queue
                                                                                        ;
requested
exten => q_login,n,Playback(silence/1)  ; answer the channel by playing one second of
silence

; If the member was added to the queue successfully, then playback "Agent logged in",
otherwise, state an error occurred
exten => q_login,n,ExecIf($["${AQMSTATUS}" =
"ADDED"]?Playback(agent-loginok):Playback(an-error-has-occurred))
exten => q_login,n,Hangup()


; ### Logout queue member ###
exten => q_logout,1,Verbose(2,Logging ${thisActiveMember} out of ${thisQueue} queue)
exten => q_logout,n,RemoveQueueMember(${thisQueue},${thisActiveMember})
exten => q_logout,n,Playback(silence/1)
exten => q_logout,n,ExecIf($["${RQMSTATUS}" =
"REMOVED"]?Playback(agent-loggedoff):Playback(an-error-has-occurred))
exten => q_logout,n,Hangup()
```

And that's it! Give it a shot and you should see console output similar to the following which will login and logout your queue members to the queues you've configured.

You can see there are already a couple of queue members logged into the sales queue.

```
*CLI> queue show sales
sales        has 0 calls (max unlimited) in 'rrmemory' strategy (3s holdtime, 4s
talktime), W:0, C:2, A:1, SL:0.0% within 0s
   Members:
      SIP/0004f2040001 (dynamic) (Not in use) has taken no calls yet
      SIP/0004f2040002 (dynamic) (Not in use) has taken no calls yet
   No Callers
```

Then we dial *100 to logout the active device from the sales queue.

```
*CLI>   == Using SIP RTP CoS mark 5
    -- Executing [*100@devices:1] Set("SIP/0004f2040001-00000012", "xtn=100") in new
stack
    -- Executing [*100@devices:2] Goto("SIP/0004f2040001-00000012",
"queueLoginLogout,member_check,1") in new stack
    -- Goto (queueLoginLogout,member_check,1)
    -- Executing [member_check@queueLoginLogout:1] Verbose("SIP/0004f2040001-00000012",
"2,Logging queue member in or out of the request queue") in new stack
  == Logging queue member in or out of the request queue
    -- Executing [member_check@queueLoginLogout:2] Set("SIP/0004f2040001-00000012",
"thisQueue=sales") in new stack
    -- Executing [member_check@queueLoginLogout:3] Set("SIP/0004f2040001-00000012",
"queueMembers=SIP/0004f2040001,SIP/0004f2040002") in new stack
    -- Executing [member_check@queueLoginLogout:4] Set("SIP/0004f2040001-00000012",
"thisActiveMember=SIP/0004f2040001") in new stack
    -- Executing [member_check@queueLoginLogout:5] GotoIf("SIP/0004f2040001-00000012",
"0?q_login,1") in new stack
    -- Executing [member_check@queueLoginLogout:6] Set("SIP/0004f2040001-00000012",
"field=1") in new stack
    -- Executing [member_check@queueLoginLogout:7] Set("SIP/0004f2040001-00000012",
"logged_in=0") in new stack
    -- Executing [member_check@queueLoginLogout:8] Set("SIP/0004f2040001-00000012",
"thisQueueMember=SIP/0004f2040001") in new stack
    -- Executing [member_check@queueLoginLogout:9] While("SIP/0004f2040001-00000012",
"1") in new stack
    -- Executing [member_check@queueLoginLogout:10] GotoIf("SIP/0004f2040001-00000012",
"0?check_next") in new stack
    -- Executing [member_check@queueLoginLogout:11] Set("SIP/0004f2040001-00000012",
"logged_in=1") in new stack
    -- Executing [member_check@queueLoginLogout:12]
ExitWhile("SIP/0004f2040001-00000012", "") in new stack
    -- Jumping to priority 15
    -- Executing [member_check@queueLoginLogout:16] GotoIf("SIP/0004f2040001-00000012",
"0?q_login,1:q_logout,1") in new stack
    -- Goto (queueLoginLogout,q_logout,1)
    -- Executing [q_logout@queueLoginLogout:1] Verbose("SIP/0004f2040001-00000012",
"2,Logging SIP/0004f2040001 out of sales queue") in new stack
  == Logging SIP/0004f2040001 out of sales queue
    -- Executing [q_logout@queueLoginLogout:2]
RemoveQueueMember("SIP/0004f2040001-00000012", "sales,SIP/0004f2040001") in new stack
[Nov 12 12:08:51] NOTICE[11582]: app_queue.c:4842 rqm_exec: Removed interface
'SIP/0004f2040001' from queue 'sales'
    -- Executing [q_logout@queueLoginLogout:3] Playback("SIP/0004f2040001-00000012",
"silence/1") in new stack
    -- <SIP/0004f2040001-00000012> Playing 'silence/1.slin' (language 'en')
    -- Executing [q_logout@queueLoginLogout:4] ExecIf("SIP/0004f2040001-00000012",
"1?Playback(agent-loggedoff):Playback(an-error-has-occurred)") in new stack
    -- <SIP/0004f2040001-00000012> Playing 'agent-loggedoff.slin' (language 'en')
    -- Executing [q_logout@queueLoginLogout:5] Hangup("SIP/0004f2040001-00000012", "") in
new stack
  == Spawn extension (queueLoginLogout, q_logout, 5) exited non-zero on
'SIP/0004f2040001-00000012'
```

And we can see that the device we loggd out by running 'queue show sales'.

```
*CLI> queue show sales
sales        has 0 calls (max unlimited) in 'rrmemory' strategy (3s holdtime, 4s
talktime), W:0, C:2, A:1, SL:0.0% within 0s
   Members:
      SIP/0004f2040002 (dynamic) (Not in use) has taken no calls yet
   No Callers
```

Pausing and Unpausing Members of Queues

Once we have our queue members logged in, it is inevitable that they will want to pause themselves during breaks, and other short periods of inactivity. To do this we can utilize the 'queue pause' and 'queue unpause' CLI commands.

We have two devices logged into the sales queue as we can see with the 'queue show sales' CLI command.

```
*CLI> queue show sales
sales        has 0 calls (max unlimited) in 'rrmemory' strategy (0s holdtime, 0s
talktime), W:0, C:0, A:0, SL:0.0% within 0s
   Members:
      SIP/0004f2040002 (dynamic) (Not in use) has taken no calls yet
      SIP/0004f2040001 (dynamic) (Not in use) has taken no calls yet
   No Callers
```

We can then pause our devices with 'queue pause' which has the following format.

```
Usage: queue {pause|unpause} member <member> [queue <queue> [reason <reason>]]
 Pause or unpause a queue member. Not specifying a particular queue
 will pause or unpause a member across all queues to which the member
 belongs.
```

Lets pause device 0004f2040001 in the sales queue by executing the following.

```
*CLI> queue pause member SIP/0004f2040001 queue sales
paused interface 'SIP/0004f2040001' in queue 'sales' for reason 'lunch'
```

And we can see they are paused with 'queue show sales'.

```
*CLI> queue show sales
sales        has 0 calls (max unlimited) in 'rrmemory' strategy (0s holdtime, 0s
talktime), W:0, C:0, A:0, SL:0.0% within 0s
   Members:
      SIP/0004f2040002 (dynamic) (Not in use) has taken no calls yet
      SIP/0004f2040001 (dynamic) (paused) (Not in use) has taken no calls yet
   No Callers
```

At this point the queue member will no longer receive calls from the system. We can unpause them with the CLI command 'queue unpause member'.

```
*CLI> queue unpause member SIP/0004f2040001 queue sales
unpaused interface 'SIP/0004f2040001' in queue 'sales'
```

And if you don't specify a queue, it will pause or unpause from all queues.

```
*CLI> queue pause member SIP/0004f2040001
paused interface 'SIP/0004f2040001'
```

Of course we want to allow the agents to pause and unpause themselves from their devices, so we need to create an extension and some dialplan logic for that to happen.

Below we've created the pattern patch **_0[01]! which will match on *00 and *01, and will *also** match with zero or more digits following it, such as the queue extension number.

So if we want to pause ourselves in all queues, we can dial *00; unpausing can be done with *01. But if our agents just need to pause or unpause themselves from a single queue, then we will also accept *00100 to pause in queue 100 (sales), or we can unpause ourselves from sales with *01100.

```
extensions.conf
---------------

; Allow queue members to pause and unpause themselves from all queues, or an individual
queue.
;
; _*0[01]! pattern match will match on *00 and *01 plus 0 or more digits.
exten => _*0[01]!,1,Verbose(2,Pausing or unpausing queue member from one or more queues)
exten => _*0[01]!,n,Set(xtn=${EXTEN:3})
; save the queue extension to 'xtn'
exten => _*0[01]!,n,Set(thisQueue=${GLOBAL(QUEUE_${xtn})})
; get the queue name if available
exten => _*0[01]!,n,GotoIf($[${ISNULL(${thisQueue})} &
${EXISTS(${xtn})}]?invalid_queue,1)  ; if 'thisQueue' is blank and the

;  the agent dialed a queue exten,

;  we will tell them it's invalid
```

The following line will determine if we're trying to pause or unpause. This is done by taking the value dialed (e.g. *00100) and chopping off the first 2 digits which leaves us with 0100, and then the :1 will return the next digit, which in this case is '0' that we're using to signify that the queue member wants to be paused (in queue 100).

So we're doing the following with our EXTEN variable.

```
${EXTEN:2:1}
offset        ^ ^  length
```

Which causes the following.

```
*00100
 ^^      offset these characters

 *00100
   ^     then return a digit length of one, which is digit 0
```

```
exten => _*0[01]!,n,GotoIf($[${EXTEN:2:1} = 0]?pause,1:unpause,1)                       ;
determine if they wanted to pause
                                                                                         ;
or to unpause.
```

The following two extensions, pause & unpause, are used for pausing and unpausing our extension from the queue(s). We use the PauseQueueMember() and UnpauseQueueMember() dialplan applications which accept the queue name (optional) and the queue member name. If the queue name is not provided, then it is assumed we want to pause or unpause from all logged in queues.

```
; Unpause ourselves from one or more queues
exten => unpause,1,NoOp()
exten => unpause,n,UnpauseQueueMember(${thisQueue},SIP/${CHANNEL(peername)})            ;
if 'thisQueue' is populated we'll pause in
                                                                                         ;
that queue, otherwise, we'll unpause in
                                                                                         ;
in all queues
```

Once we've unpaused ourselves, we use GoSub() to perform some common dialplan logic that is used for pausing and unpausing. We pass three arguments to the subroutine:

- variable name that contains the result of our operation
- the value we're expecting to get back if successful
- the filename to play

```
exten => unpause,n,GoSub(changePauseStatus,start,1(UPQMSTATUS,UNPAUSED,available))      ;
use the changePauseStatus subroutine and
                                                                                        ;
pass the values for: variable to check,
                                                                                        ;
value to check for, and file to play
exten => unpause,n,Hangup()
```

And the same method is done for pausing.

```
; Pause ourselves in one or more queues
exten => pause,1,NoOp()
exten => pause,n,PauseQueueMember(${thisQueue},SIP/${CHANNEL(peername)})
exten => pause,n,GoSub(changePauseStatus,start,1(PQMSTATUS,PAUSED,unavailable))
exten => pause,n,Hangup()
```

Lets explore what happens in the subroutine we're using for pausing and unpausing.

```
; ### Subroutine we use to check pausing and unpausing status ###
[changePauseStatus]
; ARG1:  variable name to check, such as PQMSTATUS and UPQMSTATUS (PauseQueueMemberStatus
/ UnpauseQueueMemberStatus)
; ARG2:  value to check for, such as PAUSED or UNPAUSED
; ARG3:  file to play back if our variable value matched the value to check for
;
exten => start,1,NoOp()
exten => start,n,Playback(silence/1)
; answer line with silence
```

The following line is probably the most complex. We're using the IF() function inside the Playback() application which determines which file to playback to the user.

Those three values we passed in from the pause and unpause extensions could have been something like:

- ARG1 – PQMSTATUS
- ARG2 – PAUSED
- ARG3 – unavailable

So when expanded, we'd end up with the following inside the IF() function.

```
$["${PQMSTATUS}" = "PAUSED"]?unavailable:not-yet-connected
```

${PQMSTATUS} would then be expanded further to contain the status of our PauseQueueMember() dialplan application, which could either be PAUSED or NOTFOUND. So if ${PQMSTATUS} returned PAUSED, then it would match what we're looking to match on, and we'd then return 'unavailable' to Playback() that would tell the user they are now unavailable.

Otherwise, we'd get back a message saying "not yet connected" to indicate they are likely not logged into the queue they are attempting to change status in.

```
; Please note that ${ARG1} is wrapped in ${  } in order to expand the value of ${ARG1}
into
;   the variable we want to retrieve the value from, i.e. ${${ARG1}} turns into
${PQMSTATUS}
exten => start,n,Playback(${IF($["${${ARG1}}" = "${ARG2}"]?${ARG3}:not-yet-connected)})
; check if value of variable

;  matches the value we're looking

;  for and playback the file we want

;  to play if it does
```

If ${xtn} is null, then we just go to the end of the subroutine, but if it isn't then we will play back "in the queue" followed by the queue extension number indicating which queue they were (un)paused from.

```
exten => start,n,GotoIf($[${ISNULL(${xtn})}]?end)       ; if ${xtn} is null, then just Return()
exten => start,n,Playback(in-the-queue)                 ;   if not null, then playback "in the queue"
exten => start,n,SayNumber(${xtn})                      ;   and the queue number that we (un)paused from
exten => start,n(end),Return()                          ; return from were we came
```

Queue Variables

Sometimes you may want to retrieve information about a particular queue's state. You can do this by using Queue Variables, and its associated functions. For instance here is a contrived example of how to print a couple variables for the "thisQueue":

```
exten => show_variables,1,NoOp()
  same => n,Noop(${QUEUE_VARIABLES(thisQueue)})
  same => n,Verbose(0,strategy = ${QUEUESTRATEGY})
  same => n,Verbose(0,calls = ${QUEUECALLS})
  same => n,Hangup()
```

Note, QUEUE_VARIABLES needs to be called with a valid queue name, and prior to calling the other queue variable functions in order to ensure retrieval of the correctly associated values for a given queue.

Conclusion

You should now have a simple system that permits you to login and out of queues you create in queues.conf, and to allow queue members to pause themselves within one or more queues. There are a lot of dialplan concepts utilized in this article, so you are encouraged to seek out additional documentation if any of these concepts are a bit fuzzy for you.

A good start is the doc/ subdirectory of the Asterisk sources, or the various configuration samples files located in the configs/ subdirectory of your Asterisk source code.

Configuring Call Queues with AEL

Top-level for configuring call queues

Using 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".

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/PleaseHoldAndYourCallWillBeAnsweredByOurNextAvailableSalesRepresentativ
e);
        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 => {
        Set(CALLERID(name)=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.

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.

Controlling the way Queues Call 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_ONCE=${exten}@agents);
        Dial(${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.

Queue Pre-Acknowledgement Messages

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)
```

Queue Caveats

In the above examples, some of the possible error checking has been omitted, to reduce clutter and make the examples clearer.

The Read Application

The **Read()** application allows you to play a sound prompt to the caller and retrieve DTMF input from the caller, and save that input in a variable. The first parameter to the **Read()** application is the name of the variable to create, and the second is the sound prompt or prompts to play. (If you want multiple prompts, simply concatenate them together with ampersands, just like you would with the **Background()** application.) There are some additional parameters that you can pass to the **Read()** application to control the number of digits, timeouts, and so forth. You can get a complete list by running the core show application read command at the Asterisk CLI. If no timeout is specified, **Read()** will finish when the caller presses the hash key (**#**) on their keypad.

```
exten=>6123,1,Read(Digits,enter-ext-of-person)
exten=>6123,n,Playback(you-entered)
exten=>6123,n,SayNumber(${Digits})
```

In this example, the **Read()** application plays a sound prompt which says "Please enter the extension of the person you are looking for", and saves the captured digits in a variable called **Digits**. It then plays a sound prompt which says "You entered" and then reads back the value of the **Digits** variable.

SMS

The SMS application

SMS() is an application to handles calls to/from text message capable phones and message centres using ETSI ES 201 912 protocol 1 FSK messaging over analog calls.

Basically it allows sending and receiving of text messages over the PSTN. It is compatible with BT Text service in the UK and works on ISDN and PSTN lines. It is designed to connect to an ISDN or DAHDI interface directly and uses FSK so would
probably not work over any sort of compressed link (like a VoIP call using GSM codec).

Typical applications include:-

1. Connection to a message centre to send text messages - probably initiated via the manager interface or "outgoing" directory
2. Connection to an POTS line with an SMS capable phone to send messages - probably initiated via the manager interface or "outgoing" directory
3. Acceptance of calls from the message centre (based on CLI) and storage of received messages
4. Acceptance of calls from a POTS line with an SMS capable phone and storage of received messages

Arguments to sms():

- First argument is queue name
- Second is options:
  - a: SMS() is to act as the answering side, and so send the initial FSK frame
  - s: SMS() is to act as a service centre side rather than as terminal equipment

- If a third argument is specified, then SMS does not handle the call at all, but takes the third argument as a destination number to send an SMS to

- The forth argument onward is a message to be queued to the number in the third argument. All this does is create the file in the me-sc directory.

- If 's' is set then the number is the source address and the message placed in the sc-me directory.

All text messages are stored in /var/spool/asterisk/sms

A log is recorded in /var/log/asterisk/sms

There are two subdirectories called sc-me.<queuename> holding all messages from service centre to phone, and me-sc.<queuename> holding all messages from phone to service centre.

In each directory are messages in files, one per file, using any filename not starting with a dot.

When connected as a service centre, SMS(s) will send all messages waiting in the sc-me-<queuename> directory, deleting the files as it goes. Any received in this mode are placed in the me-sc-<queuename> directory.

When connected as a client, SMS() will send all messages waiting in the me-sc-<queuename> directory, deleting the files as it goes. Any received in this mode are placed in the sc-me-<queuename> directory.

Message files created by SMS() use a time stamp/reference based filename.

The format of the sms file is lines that have the form of key=value

Keys are :

- oa - Originating Address. Telephone number, national number if just digits. Telephone number starting with + then digits for international. Ignored on sending messages to service centre (CLI used)
- da - Destination Address. Telephone number, national number if just digits. Telephone number starting with + then digits for international.
- scts - Service Centre Time Stamp in the format YYYY-MM-DD HH:MM:SS
- pid - Protocol Identifier (decimal octet value)
- dcs - Data coding scheme (decimal octet value)
- mr - Message reference (decimal octet value)
- ud - The message (see escaping below)
- srr - 0/1 Status Report Request
- rp - 0/1 Return Path
- vp - mins validity period

Omitted fields have default values.

Note that there is special format for ud, ud# instead of ud= which is followed by raw hex (2 characters per octet). This is used in output where characters other than 10,13,32-126,128-255 are included in the data. In this case a comment (line starting ;) is added showing the printable characters

When generating files to send to a service centre, only da and ud need be specified. oa is ignored.

When generating files to send to a phone, only oa and ud need be specified. da is ignored.

When receiving a message as a service centre, only the destination address is sent, so the originating address is set to the callerid.

EXAMPLES

The following are examples of use within the UK using BT Text SMS/landline service.

This is a context to use with a manager script.

```
[smsdial]
; create and send a text message, expects number+message and
; connect to 17094009
exten => _X.,1,SMS(${CALLERIDNUM},,${EXTEN},${CALLERIDNAME})
exten => _X.,n,SMS(${CALLERIDNUM})
exten => _X.,n,Hangup
```

The script sends

```
action: originate
 callerid: message <from>
 exten: to
 channel: Local/17094009
 context: smsdial
 priority: 1
```

You put the message as the name of the caller ID (messy, I know), the originating number and hence queue name as the number of the caller ID and the exten as the number to which the sms is to be sent. The context uses SMS to create the message in the queue and then SMS to communicate with 17094009 to actually send the message.

Note that the 9 on the end of 17094009 is the sub address 9 meaning no sub address (BT specific). If a different digit is used then that is the sub address for the sending message source address (appended to the outgoing CLI by BT).

For incoming calls you can use a context like this :-

```
[incoming]
exten => _XXXXXX/_8005875290,1,SMS(${EXTEN:3},a)
exten => _XXXXXX/_8005875290,n,System(/usr/lib/asterisk/smsin ${EXTEN:3})
exten => _XXXXXX/_80058752[0-8]0,1,SMS(${EXTEN:3}${CALLERIDNUM:8:1},a)
exten => _XXXXXX/_80058752[0-8]0,n,System(/usr/lib/asterisk/smsin
${EXTEN>:3}${CALLERIDNUM:8:1})
exten => _XXXXXX/_80058752[0-8]0,n,Hangup
```

In this case the called number we get from BT is 6 digits (XXXXXX) and we are using the last 3 digits as the queue name.

Priority 1 causes the SMS to be received and processed for the incoming call. It is from 080058752X0. The two versions handle the queue name as 3 digits (no sub address) or 4 digits (with sub address). In both cases, after the call a script (smsin) is run - this is optional, but is useful to actually processed the received queued SMS. In our case we email them based on the target number. Priority 3 hangs up.

If using the CAPI drivers they send the right CLI and so the _800... would be _0800...

The Verbose and NoOp Applications

Asterisk has a convenient dialplan applications for printing information to the command-line interface, called **Verbose()**. The **Verbose()** application takes two parameters: the first parameter is the minimum verbosity level at which to print the message, and the second parameter is the message to print. This extension would print the current channel identifier and unique identifier for the current call, if the verbosity level is two or higher.

```
exten=>6123,1,Verbose(2,The channel name is ${CHANNEL})
exten=>6123,n,Verbose(2,The unique id is ${UNIQUEID})
```

The **NoOp()** application stands for "No Operation". In other words, it does nothing. Because of the way Asterisk prints everything to the console if your verbosity level is three or higher, however, the **NoOp()** application is often used to print debugging information to the console like the **Verbose()** does. While you'll probably come across examples of the **NoOp()** application in other examples, we recommend you use the **Verbose()** application instead.

Voicemail

The Asterisk voicemail module provides two key applications for dealing with voice mail.

The **VoiceMail()** application takes two parameters:

1. **Mailbox**
   - This parameter specifies the mailbox in which the voice mail message should be left. It should be a mailbox number and a voice mail context concatenated with an at-sign (@), like **6001@default**. (Voice mail boxes are divided out into various voice mail context, similar to the way that extensions are broken up into dialplan contexts.) If the voice mail context is omitted, it will default to the **default** voice mail context.
2. **Options**
   - One or more options for controlling the mailbox greetings. The most popular options include the u option to play the unavailable message, the **b** option to play the busy message, and the **s** option to skip the system-generated instructions.

The **VoiceMailMain()** application allows the owner of a voice mail box to retrieve their messages, as well as set mailbox options such as greetings and their PIN number. The **VoiceMailMain()** application takes two parameters:

1. **Mailbox** - This parameter specifies the mailbox to log into. It should be a mailbox number and a voice mail context, concatenated with an at-sign (@), like 6001@default. If the voice mail context is omitted, it will default to the default voice mail context. If the mailbox number is omitted, the system will prompt the caller for the mailbox number.
2. **Options** - One or more options for controlling the voicemail system. The most popular option is the s option, which skips asking for the PIN number

> **Direct Access to Voicemail**
> Please exercise extreme caution when using the s option! With this option set, anyone which has access to this extension can retrieve voicemail messages without entering the mailbox passcode.

ODBC Voicemail 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	
msg_id	varchar(40)	Yes		NULL	(See Note)

Upgrade Notice
The msg_id column is new in Asterisk 11. Existing installations should add this column to their schema when upgrading to Asterisk 11. Existing voicemail messages will have this value populated when the messages are initially manipulated by app_voicemail in Asterisk 11.

The database name (from /etc/asterisk/res_odbc.conf) is in the odbcstorage variable in the general section of voicemail.conf.

You may modify the voicemessages table name by using odbctable=table_name in voicemail.conf.

IMAP Voicemail 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.

IMAP VM Storage 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:
If you will not be using SSL to connect to the IMAP server:

```
$ make slx SSLTYPE=none
```

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.

IMAP VM Storage Voicemail.conf Modifications

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.

Voicemail and 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.

Separate vs. Shared E-mail 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.

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 '/'.

IMAP Voicemail 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.

IMAP Voicemail 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
- X-Asterisk-VM-Message-ID

⚠ **Upgrade Notice**
The X-Asterisk-VM-Message-ID header is new in Asterisk 11. Existing voicemail messages from older versions of Asterisk will have this header added to the message when the messages are manipulated by app_voicemail in Asterisk 11.

Message Waiting Indication

What is MWI?

This page explains the resources available for Message Waiting Indicator(or Indication) functionality in Asterisk and how to configure.

Documentation Under Construction

Configuring MWI

Here we talk about configuring Asterisk to provide MWI to endpoints or other systems.

Providing MWI to chan_pjsip endpoints

Providing MWI to a chan_pjsip endpoint requires configuring the "**mailboxes**" option in either the **endpoint** type config section, or the **aor** section.

See the descriptions linked below which explain when to use the option in each section.

Description of "mailboxes" option for the Endpoint section

Description of "mailboxes" option for the AOR section.

Configuring External MWI

Let's look at configuring Asterisk to receive MWI from other systems.

Depending on your Asterisk version and configuration, there are a few different ways to configure receiving MWI from external sources.

1. **chan_sip**: outbound MWI subscriptions and receiving unsolicited MWI NOTIFY messages
2. **res_external_mwi**: A module providing an API for other systems to communicate MWI state to Asterisk
3. **chan_pjsip**: Setting `incoming_mwi_mailbox` on an endpoint

> ⚠ **res_pjsip**: The functionality for outbound SIP subscription is not available in res_pjsip yet. Internal infrastructure is built that would allow it, so if this is something you want to work on, please contact the Asterisk development community.

Outbound MWI subscription with chan_sip

Asterisk can subscribe to receive MWI from another SIP server and store it locally for retrieval by other phones. At this time, you can only subscribe using UDP as the transport. Format for the MWI register statement is:

```
;[general]
;mwi => user[:secret[:authuser]]@host[:port]/mailbox
;
; Examples:
;mwi => 1234:password@mysipprovider.example.com/1234
;mwi => 1234:password@myportprovider.example.com:6969/1234
;mwi => 1234:password:authuser@myauthprovider.example.com/1234
;mwi => 1234:password:authuser@myauthportprovider.example.com:6969/1234
```

MWI received will be stored in the 1234 mailbox of the SIP_Remote context. It can be used by other phones by setting their SIP peers "mailbox" option to the <mailbox_number>@SIP_Remote. e.g. mailbox=1234@SIP_Remote

Reception of unsolicited MWI NOTIFY with chan_sip

A chan_sip peer can be configured to receive unsolicited MWI NOTIFY messages and associate them with a particular mailbox.

```
;[somesippeer]
;unsolicited_mailbox=123456789
```

If the remote SIP server sends an unsolicited MWI NOTIFY message the new/old message count will be stored in the configured virtual mailbox. It can be used by any device supporting MWI by specifying mailbox=<configured value>@SIP_Remote as the mailbox for the desired SIP peer.

res_external_mwi

External sources can use the API provided by res_external_mwi to communicate MWI and mailbox state.

Documentation Under Construction

Asterisk 12 Configuration_res_mwi_external

res_external_mwi.so is mutually exclusive with app_voicemail.so. You'll have to load only the one you want to use.

chan_pjsip

The endpoint parameter `incoming_mwi_mailbox` (introduced in 13.18.0 and 14.7.0) takes a `<mailbox>@<context>` value. When an unsolicited NOTIFY message is received ***from*** this endpoint with an event type of `message-summary` and the `incoming_mwi_mailbox` parameter is set, Asterisk will automatically publish the new/old message counts for the specified mailbox on the internal stasis bus for any other module to use. For instance, if you have an analog phone and you specify `mailbox=userx@default` in chan_dahdi.conf, when a NOTIFY comes in on a pjsip endpoint with `incoming_mwi_mailbox=userx@default`, chan_dahdi will automatically pick that up and turn the MWI light on on the analog phone.

Short Message Service (SMS)

Information about Asterisk and SMS

Introduction to SMS

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 http://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.

SMS and extensions.conf

The following contexts are recommended.

```
; Mobile Terminated, RX. This is used when an incoming call from the SMS arrives
; 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)
```

SMS 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.

SMS 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.

SMS 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.

SMS 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.

SMS 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

SMS 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).

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

> ⚠ 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

Shared Line Appearances (SLA)

What are shared line appearances and how do they work in Asterisk? Read on...

Introduction to Shared Line Appearances (SLA)

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.

SLA Configuration

How-to configure SLA in Asterisk

SLA Configuration 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.

SLA Dialplan Configuration

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.

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.

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.

SLA Trunk Configuration

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 => _1NXXNXXXXXX,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.

SLA Station Configuration

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:

- Add the SIP channel as a station in sla.conf.

- 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.
- 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".
    - Two line buttons must be configured to subscribe to the state of the following extensions: - station1_line1 - station1_line2
    - The line appearance buttons should be configured to dial the extensions that they are subscribed to when they are pressed.
    - 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.

SLA Configuration Examples

Example configurations for SLA

Basic SLA Configuration Example

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)
```

SLA and Voicemail Example

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)
```

SLA and Call Handling

Read about call handling related to SLA

SLA Call Handling 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.

SLA 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.

SLA 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.

SLA 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.

Functions

Under Construction

In This Section

Asterisk Dialplan Functions

Asterisk functions are very similar to functions in many programming languages.

Functions are:

- Sophisticated subroutines that help you manipulate data in a variety of ways.
- Callable from within dialplan and Asterisk's various interfaces.

Compared to Dialplan Applications:

- Dialplan Functions tend to be geared towards manipulating channel data and attributes as well as providing general tools for manipulating data in variables and expressions, whether they are channel related or not.
- Dialplan Applications tend to take over the channel and provide more complex features to the channel.

Both will be typically be used with a channel but in different ways. For example you may send a channel to an application such as Playback and then let it fly, but with functions you might use one or many functions in a single expression to manipulate or move around data related to the channels operation.

Applications and functions are regularly used together and as you use them you'll see that the distinction between applications and the more powerful functions can sometimes be murky.

Available functions

Many functions come with Asterisk by default. For a complete list of the dialplan functions available to your installation of Asterisk, type **core show functions** at the Asterisk CLI. Not all functions are compiled with Asterisk by default, so if you have the source available then you may want to browse the functions listed in menuselect under "Dialplan Functions". Dedicated function modules start with "func_", but the core or other large modules may provide functions as well.

Since anyone can write an Asterisk module they can also be obtained from sources outside the Asterisk distribution. Pre-packaged community or commercial Asterisk distributions that have a special purpose may include a custom function or two.

General function syntax

The general syntax for calling a function follows:

```
FUNCTION(argument1,argument2, ...)
```

A function's value can be set using the Set application. The example below will show how to set the CHANNEL function argument "tonezone" to the value "de" (for Germany).

```
same => n,Set(CHANNEL(tonezone)=de)
```

A function's value can be referenced almost anywhere in dialplan where you can use an expression or reference a variable. The value can be referenced by encapsulating the call with curly braces and a leading dollar sign.

```
${FUNCTION(argument)}
```

For example, if we wanted to log the destination address for the audio stream of the channel:

```
same => n,Log(NOTICE, The destination for the audio stream is: ${CHANNEL(rtp,dest)})
```

Help for specific functions

The wiki section CLI Syntax and Help Commands details how to use the CLI-accessible documentation. This will allow you to access the syntax and usage info for each function including detail on all the arguments for each function.

Wikibot also publishes the same documentation on the wiki. You can find function docs in the version specific top level sections; such as Asterisk 13 Dialplan Functions.

Asterisk Dialplan Function Examples

UNDER CONSTRUCTION

Function Examples

Asterisk includes a wide variety of functions. Here we'll show you a few commonly used functions and a selection of others to give you an idea of what you can do.

CHANNEL and CHANNELS

CHANNEL Gets or sets various pieces of information about the channel. Additional arguments may be available from the channel driver; see its documentation for details. Any item requested that is not available on the current channel will return an empty string. CHANNELS on the other hand, gets the list of channels while optionally filtering by a regular expression (provided via argument). If no argument is provided, all known channels are returned. The regular_expression must correspond to the POSIX.2 specification, as shown in regex(7). The list returned will be space-delimited.

See the CHANNEL function reference documentation for an extensive list of arguments.

Examples:

Push a hangup handler subroutine onto the channel. The hangup handler must exist at the location specified (default,s,1).

```
same = n,Set(CHANNEL(hangup_handler_push)=default,s,1)
```

Using the CHANNEL function along with the Log application, we can log the current state of the channel.

```
same = n,Log(NOTICE, This channel is: ${CHANNEL(state)})
```

Set the channel variable myvar to a space-delimited list of all channels.

```
same = n,Set(myvar=${CHANNELS})
```

DB and other DB functions

The DB function will read from or write a value to the Asterisk Internal Database. On a read, this function returns the corresponding value from the database, or blank if it does not exist. Reading a database value will also set the variable DB_RESULT. There are a few related functions. DB_EXISTS, DB_DELETE and DB_KEYS.

If you wish to find out if an entry exists, use the DB_EXISTS function. The DB_DELETE function will retrieve a value from the Asterisk database and then remove that key from the database. DB_RESULT will be set to the key's value if it exists. Finally, the DB_KEYS will return a comma-separated list of keys existing at the prefix specified within the Asterisk database. If no argument is provided, then a list of key families will be returned.

Examples:

Set the key "testkey" in family "testfamily" to the value "Alice".

```
same = n,Set(DB(testfamily/testkey)=Alice)
```

Dialing a PJSIP endpoint using the value of the previously set key as the endpoint name.

```
same = n,Dial(PJSIP/${DB(testfamily/testkey)})
```

Go to a specific dialplan location (via label) depending on if the key exists or does not.

```
same = n,Gotoif($[${DB_EXISTS(testfamily/testkey)}]?keyexists:keydoesnotexist)
```

Delete the entry while logging the value of the key!

```
same = n,Log(NOTICE, Deleting the key testfamily/testkey which had the value:
${DB_DELETE(testfamily/testkey)})
```

Database Transactions

As of 1.6.2, Asterisk now supports doing database transactions from the Dialplan. A number of new applications and functions have been introduced for this purpose and this document should hopefully familiarize you with all of them.

First, the `ODBC()` function has been added which is used to set up all new database transactions. Simply write the name of the transaction to this function, along with the arguments of "transaction" and the database name, e.g. `Set(ODBC(transaction,postgres-asterisk)=foo)`. In this example, the name of the transaction is "foo". The name doesn't really matter, unless you're manipulating multiple transactions within the same dialplan, at the same time. Then, you use the transaction name to change which transaction is active for the next dialplan function.

The `ODBC()` function is also used to turn on a mode known as `forcecommit`. For most cases, you won't need to use this, but it's there. It simply causes a transaction to be committed, when the channel hangs up. The other property which may be set is the `isolation` property. Please consult with your database vendor as to which values are supported by their ODBC driver. Asterisk supports setting all standard ODBC values, but many databases do not support the entire complement.

Finally, when you have run multiple statements on your transaction and you wish to complete the transaction, use the `ODBC_Commit` and `ODBC_Rollback` applications, along with the transaction ID (in the example above, "foo") to commit or rollback the transaction. Please note that if you do not explicitly commit the transaction or if `forcecommit` is not turned on, the transaction will be automatically rolled back at channel destruction (after hangup) and all related database resources released back to the pool.

Manipulating Party ID Information

Introduction

This chapter aims to explain how to use some of the features available to manipulate party ID information. It will not delve into specific channel configuration options described in the respective sample configuration files. The party ID information can consist of Caller ID, Connected Line ID, redirecting to party ID information, and redirecting from party ID information. Meticulous control is needed particularly when interoperating between different channel technologies.

- Caller ID: The Caller ID information describes who is originating a call.
- Connected Line ID: The Connected Line ID information describes who is connected to the other end of a call while a call is established. Unlike Caller ID, the connected line information can change over the life of a call when call transfers are performed. The connected line information can also change in either direction because either end could transfer the call. For ISDN it is known as Connected Line Identification Presentation (COLP), Connected Line Identification Restriction (COLR), and Explicit Call Transfer (ECT). For SIP it is known either as P-Asserted-Identity or Remote-Party-Id.
- Redirecting information: When a call is forwarded, the call originator is informed that the call is redirecting-to a new destination. The new destination is also informed that the incoming call is redirecting-from the forwarding party. A call can be forwarded repeatedly until a new destination answers it or a forwarding limit is reached.

Tools available

Asterisk contains several tools for manipulating the party ID information for a call. Additional information can be found by using the 'core show function' or 'core show application' console commands at the Asterisk CLI. The following list identifies some of the more common tools for manipulating the party ID information:

- `CALLERID(datatype,caller-id)`
- `CONNECTEDLINE(datatype,i)`
- `REDIRECTING(datatype,i)`
- `Dial()` and `Queue()` dialplan application 'I' option
- Interception macros
- Channel driver specific configuration options.

CALLERID dialplan function

The CALLERID function has been around for quite a while and its use is straightforward. It is used to examine and alter the caller information that came into the dialplan with the call. Then the call with it's caller information passes on to the destination using the Dial() or Queue() application.

The CALLERID information is passed during the initial call setup. However, depending on the channel technology, the caller name may be delayed. Q.SIG is an example where the caller name may be delayed so your dialplan may need to wait for it.

CONNECTEDLINE dialplan function

The CONNECTEDLINE function does the opposite of the CALLERID function. CONNECTEDLINE can be used to setup connected line information to be sent when the call is answered. You can use it to send new connected line information to the remote party on the channel when a call is transferred. The CONNECTEDLINE information is passed when the call is answered and when the call is transferred.

> ⚠ It is up to the channel technology to determine when to act upon connected line updates before the call is answered. ISDN will just store the updated information until the call is answered. SIP could immediately update the caller with a provisional response or wait for some other event to notify the caller.

Since the connected line information can be sent while a call is connected, you may need to prevent the channel driver from acting on a **partial** update.

The 'i' option is used to inhibit the channel driver from sending the changed information immediately.

REDIRECTING dialplan function

The REDIRECTING function allows you to report information about forwarded/deflected calls to the caller and to the new destination. The use of the REDIRECTING function is the most complicated of the party information functions.

The REDIRECTING information is passed during the initial call setup and while the call is being routed through the network. Since the redirecting information is sent before a call is answered, you need to prevent the channel driver from acting on a partial update. The 'i' option is used to inhibit the channel driver from sending the changed information immediately.

The incoming call may have already been redirected. An incoming call has already been redirected if the REDIRECTING(count) is not zero. (Alternate indications are if the REDIRECTING(from-num-valid) is non-zero or if the REDIRECTING(from-num) is not empty.)

There are several things to do when a call is forwarded by the dialplan:

- Setup the REDIRECTING(to-xxx) values to be sent to the caller.
- Setup the REDIRECTING(from-xxx) values to be sent to the new destination.
- Increment the REDIRECTING(count).
- Set the REDIRECTING(reason).
- Dial() the new destination.

Special REDIRECTING considerations for ISDN

Special considerations for Q.SIG and ISDN point-to-point links are needed to make the DivertingLegInformation1, DivertingLegInformation2, and DivertingLegInformation3 messages operate properly.

You should manually send the COLR of the redirected-to party for an incoming redirected call if the incoming call could experience further redirects. For chan_misdn, just set the REDIRECTING(to-num,i) = ${EXTEN} and set the REDIRECTING(to-num-pres) to the COLR. For chan_dahdi, just set the REDIRECTING(to-num,i) = CALLERID(dnid) and set the REDIRECTING(to-num-pres) to the COLR. (Setting the REDIRECTING(to-num,i) value may not be necessary since the channel driver has already attempted to preset that value for automatic generation of the needed DivertingLegInformation3 message.)

For redirected calls out a trunk line, you need to use the 'i' option on all of the REDIRECTING statements before dialing the redirected-to party. The call will update the redirecting-to presentation (COLR) when it becomes available.

Dial() and Queue() dialplan application 'I' option

In the dialplan applications Dial() and Queue(), the 'I' option is a brute force option to block connected line and redirecting information updates while the application is running. Blocking the updates prevents the update from overwriting any CONNECTEDLINE or REDIRECTING values you may have setup before running the application.

The option blocks all redirecting updates since they should only happen before a call is answered. The option only blocks the connected line update from the initial answer. Connected line updates resulting from call transfers happen after the application has completed. Better control of connected line and redirecting information is obtained using the interception macros.

Party ID Interception Macros and Routines

You can find detailed information in the Party ID Interception Macros and Routines section.

Manipulation examples

The following examples show several common scenarios in which you may need to manipulate party ID information from the dialplan.

Simple recording playback

```
exten => 1000,1,NoOp
; The CONNECTEDLINE information is sent when the call is answered.
exten => 1000,n,Set(CONNECTEDLINE(name,i)=Company Name)
exten => 1000,n,Set(CONNECTEDLINE(name-pres,i)=allowed)
exten => 1000,n,Set(CONNECTEDLINE(num,i)=5551212)
exten => 1000,n,Set(CONNECTEDLINE(num-pres)=allowed)
exten => 1000,n,Answer
exten => 1000,n,Playback(tt-weasels)
exten => 1000,n,Hangup
```

Straightforward dial through

```
exten => 1000,1,NoOp
; The CONNECTEDLINE information is sent when the call is answered.
exten => 1000,n,Set(CONNECTEDLINE(name,i)=Company Name)
exten => 1000,n,Set(CONNECTEDLINE(name-pres,i)=allowed)
exten => 1000,n,Set(CONNECTEDLINE(num,i)=5551212)
exten => 1000,n,Set(CONNECTEDLINE(num-pres)=allowed)
; The I option prevents overwriting the CONNECTEDLINE information
; set above when the call is answered.
exten => 1000,n,Dial(SIP/1000,20,I)
exten => 1000,n,Hangup
```

Use of interception macro

```
[macro-add_pfx]
; ARG1 is the prefix to add.
; ARG2 is the number of digits at the end to add the prefix to.
; When the macro ends the CONNECTEDLINE data is passed to the
; channel driver.
exten => s,1,NoOp(Add prefix to connected line)
exten => s,n,Set(NOPREFIX=${CONNECTEDLINE(number):-${ARG2}})
exten => s,n,Set(CONNECTEDLINE(num,i)=${ARG1}${NOPREFIX})
exten => s,n,MacroExit

exten => 1000,1,NoOp
exten => 1000,n,Set(__CONNECTED_LINE_CALLER_SEND_MACRO=add_pfx)
exten => 1000,n,Set(__CONNECTED_LINE_CALLER_SEND_MACRO_ARGS=45,4)
exten => 1000,n,Dial(SIP/1000,20)
exten => 1000,n,Hangup
```

Simple redirection

```
exten => 1000,1,NoOp
; For Q.SIG or ISDN point-to-point we should determine the COLR for this
; extension and send it if the call was redirected here.
exten => 1000,n,GotoIf($[${REDIRECTING(count)}>0]?redirected:notredirected)
exten => 1000,n(redirected),Set(REDIRECTING(to-num,i)=${CALLERID(dnid)})
exten => 1000,n,Set(REDIRECTING(to-num-pres)=allowed)
exten => 1000,n(notredirected),NoOp
; Determine that the destination has forwarded the call.
; ...
exten => 1000,n,Set(REDIRECTING(from-num,i)=1000)
exten => 1000,n,Set(REDIRECTING(from-num-pres,i)=allowed)
exten => 1000,n,Set(REDIRECTING(to-num,i)=2000)
; The DivertingLegInformation3 message is needed because at this point
; we do not know the presentation (COLR) setting of the redirecting-to
; party.
exten => 1000,n,Set(REDIRECTING(count,i)=$[${REDIRECTING(count)} + 1])
exten => 1000,n,Set(REDIRECTING(reason,i)=cfu)
; The call will update the redirecting-to presentation (COLR) when it
; becomes available with a redirecting update.
exten => 1000,n,Dial(DAHDI/g1/2000,20)
exten => 1000,n,Hangup
```

Party ID propagation

For normal operations where Party A calls Party B this is what the relationship between CALLERID/CONNECTEDLINE information looks like:

```
Channel A                       Channel B
            Incoming channel --- bridge --- Outgoing channel
Party A ___ CALLERID() -------------------> CONNECTEDLINE() ___ Party B
            CONNECTEDLINE() <-------------- CALLERID()
```

The CALLERID() information is the party identification of the remote party. For Channel A that is Party A. For Channel B that is Party B.

The CONNECTEDLINE() information is the party identification of the party connected across the bridge. For Channel A that is Party B. For Channel B that is Party A.

Local channels behave in a similar way because there is an implicit two party bridge between the channels. For normal call setups, Local;1 is an outgoing channel and Local;2 is an incoming channel.

```
Local;1              Local;2
Outgoing channel --- Incoming channel
CONNECTEDLINE() ---> CALLERID()
CALLERID() <-------- CONNECTEDLINE()
```

A normal call where Party A calls Party B with a local channel in the chain.

```
Channel A                       Local;1              Local;2                         Channel B
            Incoming channel --- bridge --- Outgoing channel --- Incoming channel --- bridge --- Outgoing channel
Party A ___ CALLERID() -------------------> CONNECTEDLINE() ---> CALLERID() -------------------> CONNECTEDLINE() ___ Party B
            CONNECTEDLINE() <-------------- CALLERID() <-------- CONNECTEDLINE() <-------------- CALLERID()
```

Originated calls make the incoming and outgoing labels a bit confusing because both channels start off as outgoing. Once the originated channel answers it becomes an "incoming" channel to run dialplan. A better way is to just distinguish which channel is running dialplan. For consistency, I'll continue using the incoming and outgoing labels.

An example of originating a normal channel (Channel A) to a dialplan exten.
1) Channel A dials Party A
2) Party A answers
3) The CONNECTEDLINE update from Channel A triggered by the answer is discarded because it has nowhere to go.
4) Channel A becomes an incoming channel to run dialplan to dial Party B.

```
Channel A                       Channel B
            Incoming channel --- bridge --- Outgoing channel
Party A ___ CALLERID() -------------------> CONNECTEDLINE() ___ Party B
            CONNECTEDLINE() <-------------- CALLERID()
```

An example of originating a local channel (which will always be a Local;1) to a dialplan exten.
1) Local;1 makes Local;2 run dialplan to call Party A
2) Party A answers
3) The CONNECTEDLINE update from Channel A triggered by the answer propagates to Local;1 if not blocked by the Dial 'I' option on Local;2.
4) Local;1 becomes an incoming channel to run dialplan to dial Party B.

```
Channel A                       Local;2              Local;1                         Channel B
            Outgoing channel --- bridge --- Incoming channel --- Incoming channel --- bridge --- Outgoing channel
Party A ___ CALLERID() -------------------> CONNECTEDLINE() ---> CALLERID() -------------------> CONNECTEDLINE() ___ Party B
            CONNECTEDLINE() <-------------- CALLERID() <-------- CONNECTEDLINE() <-------------- CALLERID()
```

Ideas for usage

The following is a list of ideas in which the manipulation of party ID information would be beneficial.

- IVR that updates connected name on each selection made.
- Disguise the true number of an individual with a generic company number.
- Use interception macros to make outbound connected number E.164 formatted.
- You can do a lot more in an interception macro than just manipulate party information...

Troubleshooting tips

- For CONNECTEDLINE and REDIRECTING, check the usage of the 'i' option.
- Check channel configuration settings. The default settings may not be what you want or expect.
- Check packet captures. Your equipment may not support what Asterisk sends.

For further reading...

- Relevant ETSI ISDN redirecting specification: EN 300 207-1
- Relevant ETSI ISDN COLP specification: EN 300 097-1
- Relevant ETSI ISDN ECT specification: EN 300 369-1
- Relevant Q.SIG ISDN redirecting specification: ECMA-174
- Relevant Q.SIG ISDN COLP specification: ECMA-148
- Relevant Q.SIG ISDN ECT specification: ECMA-178
- Relevant SIP RFC for P-Asserted-Id: RFC3325
- The expired draft (draft-ietf-sip-privacy-04.txt) defines Remote-Party-Id. Since Remote-Party-Id has not made it into an RFC at this time, its use is non-standard by definition.

Simple Message Desk Interface (SMDI) Integration

Accessing SMDI information in the dialplan.

There are two dialplan functions that can be used to access the details of incoming SMDI messages.

```
*CLI> core show function SMDI_MSG_RETRIEVE

  -= Info about function 'SMDI_MSG_RETRIEVE' =-
```

Syntax

```
SMDI_MSG_RETRIEVE(<smdi port>,<search key>[,timeout[,options]])
```

Synopsis

Retrieve an SMDI message.

Description

This function is used to retrieve an incoming SMDI message. It returns an ID which can be used with the SMDI_MSG() function to access details of the message. Note that this is a destructive function in the sense that once an SMDI message is retrieved using this function, it is no longer in the global SMDI message queue, and can not be accessed by any other Asterisk channels. The timeout for this function is optional, and the default is 3 seconds. When providing a timeout, it should be in milliseconds. The default search is done on the forwarding station ID. However, if you set one of the search key options in the options field, you can change this behavior.

Options

- t - Instead of searching on the forwarding station, search on the message desk terminal.
- n - Instead of searching on the forwarding station, search on the message desk number.

```
*CLI> core show function SMDI_MSG

  -= Info about function 'SMDI_MSG' =-
```

Syntax

```
SMDI_MSG(<message_id>,<component>)
```

Synopsis

Retrieve details about an SMDI message.

Description

This function is used to access details of an SMDI message that was pulled from the incoming SMDI message queue using the SMDI_MSG_RETRIEVE() function. Valid message components are:

- station - The forwarding station
- callerid - The callerID of the calling party that was forwarded
- type - The call type. The value here is the exact character that came in on the SMDI link. Typically, example values are: D - Direct Calls, A - Forward All Calls, B - Forward Busy Calls, N - Forward No Answer Calls

Here is an example of how to use these functions:

```
; Retrieve the SMDI message that is associated with the number that
; was called in Asterisk.
exten => _0XXX,1,Set(SMDI_MSG_ID=${SMDI_MSG_RETRIEVE(/dev/tty0,${EXTEN})})

; Ensure that the message was retrieved.
exten => _0XXX,n,GotoIf($["x${SMDI_MSG_ID}" != "x"]?processcall:hangup)
exten => _0XXX,n(hangup),NoOp(No SMDI message retrieved for ${EXTEN})

; Grab the details out of the SMDI message.
exten => _0XXX,n(processcall),NoOp(Message found for ${EXTEN})
exten => _0XXX,n,Set(SMDI_EXTEN=${SMDI_MSG(${SMDI_MSG_ID},station)})
exten => _0XXX,n,Set(SMDI_CID=${SMDI_MSG(${SMDI_MSG_ID},callerid)})

; Map SMDI message types to the right voicemail option.  If it is "B", use the
; busy option.  Otherwise, use the unavailable option.
exten => _0XXX,n,GotoIf($["${SMDI_MSG(${SMDI_MSG_ID},type)}" == "B"]?usebusy:useunavail)

exten => _0XXX,n(usebusy),Set(SMDI_VM_TYPE=b)
exten => _0XXX,n,Goto(continue)

exten => _0XXX,n,(useunavil),Set(SMDI_VM_TYPE=u)

exten => _0XXX,n(continue),NoOp( Process the rest of the call ... )
```

Ensuring complete MWI information over SMDI

Another change has been made to ensure that MWI state is properly propagated over the SMDI link. This replaces the use of externnotify=smdi for voicemail.conf. The issue is that we have to poll mailboxes occasionally for changes that were made using an IMAP client. So, this ability was added to res_smdi. To configure this, there is a new section in smdi.conf. It looks like this:

```
[mailboxes]
; This section configures parameters related to MWI handling for the SMDI link.
 ;
; This option configures the polling interval used to check to see if the
; mailboxes have any new messages.  This option is specified in seconds.
; The default value is 10 seconds.
;
;pollinginterval=10
;
; Before specifying mailboxes, you must specify an SMDI interface.  All mailbox
; definitions that follow will correspond to that SMDI interface.  If you
; specify another interface, then all definitions following that will correspond
; to the new interface.
;
; Every other entry in this section of the configuration file is interpreted as
; a mapping between the mailbox ID on the SMDI link, and the local Asterisk
; mailbox name.  In many cases, they are the same thing, but they still must be
; listed here so that this module knows which mailboxes it needs to pay
; attention to.
;
; Syntax:
;   <SMDI mailbox ID>=<Asterisk Mailbox Name>[@Asterisk Voicemail Context]
;
; If no Asterisk voicemail context is specified, "default" will be assumed.
;
;
;smdiport=/dev/ttyS0
;2565551234=1234@vmcontext1
;2565555678=5678@vmcontext2
;smdiport=/dev/ttyS1
;2565559999=9999
```

Reporting

Asterisk has two reporting systems. **Call Detail Records (CDR)** and **Channel Event Logging (CEL)**. Both of these systems log specific events that occur on calls and individual channels. The events and their details are provided in a machine readable format separate from Asterisk's standard logging and debug facilities. Both systems provide at least CSV output and utilize other modules to output through a variety of back-end interfaces.

In This Section

Call Detail Records is the older system that provides one or more records for each call depending on what version of Asterisk you are using and what is happening in the call. It is useful for administrators who need a simple way to track what calls have taken place on the Asterisk system. It isn't recommended for generating billing data.

Channel Event Logging is the newer system that provides much more detail than CDR. CEL is designed to excel where CDR fails and this is noticed first in the amount of detail that CEL provides. For any given calling scenario CDR may produce one or two simple records compared to dozens of records for CEL. If you want very precise data on every call happening in Asterisk then you should use CEL. Hence, CEL is the recommended reporting system to use for generating billing data.

If you are looking for logging, debugging or Application Programming Interfaces then you should check out the following resources:

- Asterisk Logging and Asterisk Logging Configuration
- Asterisk Interfaces

Call Detail Records (CDR)

Top-level page for all things CDR

CDR Variables

If the channel has a CDR, that CDR 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.

CDR Storage Backends

Top-level page for information about storage backends for Asterisk's CDR engine.

MSSQL CDR Backend

Asterisk can currently store CDRs into a Microsoft SQL Server 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.

Only choose one db connector.

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
```

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.

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.

MySQL CDR Backend

ODBC

Using MySQL for CDR records is supported by using ODBC and the cdr_adaptive_odbc module (depends on res_odbc).

> The below cdr_mysql module has been deprecated in 1.8.

Native

Alternatively, there is a native MySQL CDR module.

To use it, configure the module in cdr_mysql.conf. Create a table called cdr under the database name you will be using the following schema.

```
CREATE TABLE cdr (
        calldate datetime NOT NULL default '0000-00-00 00:00:00',
        clid varchar(80) NOT NULL default '',
        src varchar(80) NOT NULL default '',
        dst varchar(80) NOT NULL default '',
        dcontext varchar(80) NOT NULL default '',
        channel varchar(80) NOT NULL default '',
        dstchannel varchar(80) NOT NULL default '',
        lastapp varchar(80) NOT NULL default '',
        lastdata varchar(80) NOT NULL default '',
        duration int(11) NOT NULL default '0',
        billsec int(11) NOT NULL default '0',
        disposition varchar(45) NOT NULL default '',
        amaflags int(11) NOT NULL default '0',
        accountcode varchar(20) NOT NULL default '',
        uniqueid varchar(32) NOT NULL default '',
        userfield varchar(255) NOT NULL default ''
);
```

In 1.8 and later

The following columns can also be defined:

```
peeraccount varchar(20) NOT NULL default ''
        linkedid varchar(32) NOT NULL default ''
        sequence int(11) NOT NULL default '0'
```

PostgreSQL CDR Backend

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 timestamp 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
);
```

In 1.8 and later

The following columns can also be defined:

```
peeraccount varchar(20) NOT NULL
        linkedid varchar(150) NOT NULL
        sequence int NOT NULL
```

SQLite 2 CDR Backend

SQLite version 2 is supported in cdr_sqlite.

SQLite 3 CDR Backend

SQLite version 3 is supported in cdr_sqlite3_custom.

RADIUS CDR Backend

What is needed

- FreeRADIUS server
- Radiusclient-ng library
- Asterisk PBX

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"

> ⚠ 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.
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

Channel Event Logging (CEL)

Overview

Asterisk's Channel Event Logging provides a mechanism for tracking many channel related events. CEL is granular and fine-grained, having been designed with billing information in mind. It supports many storage back-ends and is a great alternative to Call Detail Records for administrators that need extremely detailed event logs. The extensive detail will allow building of accurate billing or call-flow data.

Features:

- Control over which Asterisk applications are tracked.
- Control over which events should be raised.
- Configurable date format.
- Integration with the Asterisk Manager Interface.
- Integration with RADIUS
- Modules for various logging back-ends including customized CEL output, integration with ODBC, PGSQL, SQLite and TDS.

The child pages in this section discuss the configuration of Channel Event Logging.

Events

While CEL, CDR and AMI are all basically event tracking mechanisms, the events tracked by CEL are focused on a use case for generating billing data. The specific events and fields are covered in the **Asterisk 12 CEL Specification.**

Historical Information and Changes

CEL underwent significant rework and improvement in Asterisk 12. Please see the specific changes as mentioned under the CEL section of the New in 12 wiki page.

CEL Design Goals

Overview

This page is simply an effort at explaining the reason and purpose behind the design of CEL. You can skip this section if you don't care why things are the way they are and only want to get it up and running.

Improving on CDR

CEL, or Channel Event Logging, was written with the hopes that it will help solve some of the problems that were difficult to address in CDR records. Some difficulties in CDR generation are the fact that the CDR record stores three events: the "Start" time, the "Answer" time, and the "End" time. Billing time is usually the difference between "Answer" and "End", and total call duration was the difference in time from "Start" to "End". The trouble with this direct and simple approach is the fact that calls can be transferred, put on hold, involved in conferencing, forwarded, etc. In general, those who create billing applications with Asterisk find they have to do all sorts of very creative things to overcome the shortcomings of CDR records, often supplementing the CDR records with AGI scripts and manager event filters.

Channel-relevant events

The fundamental assumption is that the Channel is the fundamental communication object in asterisk, which basically provides a communication channel between two communication ports. It makes sense to have an event system aimed at recording important events on channels. Each event is attached to a channel, like ANSWER or HANGUP. Some events are meant to connect two or more channels, like the BRIDGE_START event. Some events, like BLINDTRANSFER, are initiated by one channel, but affect two others. These events use the Peer field, like BRIDGE would, to point to the target channel.

Billing requires detail

The design philosophy of CEL is to generate event data that can be grouped together to form a billing record. There are definite parallels between Asterisk Manager Interface events and CEL events, but there are some differences. Some events that are generated by CEL are not generated by the Manager interface (yet). CEL is optimized for databases, and Manager events are not. The focus of CEL is billing. The Manager interface is targeted to real-time monitoring and control of asterisk.

Looking at an example of all the interactions and events involved in a moderately complex call scenario can give the reader a feel for the complexities involved in billing. Please take note of the following sequence of events.

Remember that 150, 151, and 152 are all Zap extension numbers, and their respective devices are Zap/50, Zap/51, and Zap/52.

- 152 dials 151; 151 answers.
- 152 parks 151; 152 hangs up.
- 150 picks up the park (dials 701).
- 150 and 151 converse.
- 151 flashes hook; dials 152, talks to 152, then 151 flashes hook again for 3-way conference.
- 151 converses with the other two for a while, then hangs up.
- 150 and 152 keep conversing, then hang up. 150 hangs up first.(not that it matters).

Click the expansion link to see the resulting 42 CEL events with annotations.

Click here to expand...

Note that the actual CEL events below are in CSV format and do not include the ;;; and text after that which gives a description of what the event represents.

```
"EV_CHAN_START","2007-05-09
12:46:16","fxs.52","152","","","","s","extension","Zap/52-1","","","DOCUMENTATION","","1178736376.3","","" ;;; 152 takes the
phone off-hook
"EV_APP_START","2007-05-09
12:46:18","fxs.52","152","152","","","151","extension","Zap/52-1","Dial","Zap/51|30|TtWw","DOCUMENTATION","","1178736376.3"
;;; 152 finishes dialing 151
"EV_CHAN_START","2007-05-09
12:46:18","fxs.51","151","","","","s","extension","Zap/51-1","","","DOCUMENTATION","","1178736378.4","","" ;;; 151 channel
created, starts ringing
(151 is ringing)
"EV_ANSWER","2007-05-09 12:46:19","","151","152","","","151","extension","Zap/51-1","AppDial","(Outgoing
Line)","DOCUMENTATION","","1178736378.4","","" ;;; 151 answers
"EV_ANSWER","2007-05-09
12:46:19","fxs.52","152","152","","","151","extension","Zap/52-1","Dial","Zap/51|30|TtWw","DOCUMENTATION","","1178736376.3","
","" ;;; so does 152 (???)
"EV_BRIDGE_START","2007-05-09
12:46:20","fxs.52","152","152","","","151","extension","Zap/52-1","Dial","Zap/51|30|TtWw","DOCUMENTATION","","1178736376.3","
","Zap/51-1" ;;; 152 and 151 are bridged
(151 and 152 are conversing)
"EV_BRIDGE_END","2007-05-09
12:46:25","fxs.52","152","152","","","151","extension","Zap/52-1","Dial","Zap/51|30|TtWw","DOCUMENTATION","","1178736376.3","
","" ;;; after 5 seconds, the bridge ends (152 dials #700?)
"EV_BRIDGE_START","2007-05-09
12:46:25","fxs.52","152","152","","","151","extension","Zap/52-1","Dial","Zap/51|30|TtWw","DOCUMENTATION","","1178736376.3","
","Zap/51-1" ;;; extraneous 0-second bridge?
"EV_BRIDGE_END","2007-05-09
12:46:25","fxs.52","152","152","","","151","extension","Zap/52-1","Dial","Zap/51|30|TtWw","DOCUMENTATION","","1178736376.3","
","" ;;;
```

```
"EV_PARK_START","2007-05-09 12:46:27","","151","152","","","","extension","Zap/51-1","Parked
Call","","DOCUMENTATION","","1178736378.4","","" ;;; 151 is parked
"EV_HANGUP","2007-05-09
12:46:29","fxs.52","152","152","","","h","extension","Zap/52-1","","","DOCUMENTATION","","1178736376.3" ,"","" ;;; 152 hangs
up 2 sec later
"EV_CHAN_END","2007-05-09
12:46:29","fxs.52","152","152","","","h","extension","Zap/52-1","","","DOCUMENTATION","","1178736376.3","","" ;;; 152's
channel goes away
(151 is parked and listening to MOH! now, 150 picks up, and dials 701)
"EV_CHAN_START","2007-05-09
12:47:08","fxs.50","150","","","","s","extension","Zap/50-1","","","DOCUMENTATION","","1178736428.5","","" ;;; 150 picks up
the phone, dials 701
"EV_PARK_END","2007-05-09 12:47:11","","151","152","","","","extension","Zap/51-1","Parked
Call","","DOCUMENTATION","","1178736378.4","","" ;;; 151's park comes to end
"EV_ANSWER","2007-05-09
12:47:11","fxs.50","150","150","","","701","extension","Zap/50-1","ParkedCall","701","DOCUMENTATION","","1178736428.5","",""
;;; 150 gets answer (twice)
"EV_ANSWER","2007-05-09
12:47:12","fxs.50","150","150","","","701","extension","Zap/50-1","ParkedCall","701","DOCUMENTATION","","1178736428.5","",""
;;;
"EV_BRIDGE_START","2007-05-09
12:47:12","fxs.50","150","150","","","701","extension","Zap/50-1","ParkedCall","701","DOCUMENTATION","","1178736428.5","","Za
p/51-1" ;;; bridge begins between 150 and recently parked 151 (150 and 151 are conversing, then 151 hits flash)
"EV_CHAN_START","2007-05-09
12:47:51","fxs.51","151","","","","s","extension","Zap/51-2","","","DOCUMENTATION","","1178736471.6","","" ;;; 39 seconds
later, 51-2 channel is created. (151 flashes hook)
"EV_HOOKFLASH","2007-05-09 12:47:51","","151","152","","","","extension","Zap/51-1","Bridged
Call","Zap/50-1","DOCUMENTATION","","1178736378.4","","Zap/51-2" ;;; a marker to record that 151 flashed the hook
"EV_BRIDGE_END","2007-05-09
12:47:51","fxs.50","150","150","","","701","extension","Zap/50-1","ParkedCall","701","DOCUMENTATION","","1178736428.5","","Za
p/51-1" ;;; bridge ends between 150 and 151
"EV_BRIDGE_START","2007-05-09
12:47:51","fxs.50","150","150","","","701","extension","Zap/50-1","ParkedCall","701","DOCUMENTATION","","1178736428.5","","Za
p/51-1" ;;; 0-second bridge from 150 to ? 150 gets no sound at all
"EV_BRIDGE_END","2007-05-09
12:47:51","fxs.50","150","150","","","701","extension","Zap/50-1","ParkedCall","701","DOCUMENTATION","","1178736428.5","","Za
p/51-1" ;;;
"EV_BRIDGE_START","2007-05-09
12:47:51","fxs.50","150","150","","","701","extension","Zap/50-1","ParkedCall","701","DOCUMENTATION","","1178736428.5","","Za
p/51-1" ;;; bridge start on 150
(151 has dialtone after hitting flash; dials 152)
"EV_APP_START","2007-05-09
12:47:55","fxs.51","151","151","","","152","extension","Zap/51-2","Dial","Zap/52|30|TtWw","DOCUMENTATION","","1178736471.6","
","" ;;; 151-2 dials 152 after 4 seconds
"EV_CHAN_START","2007-05-09
12:47:55","fxs.52","152","","","","s","extension","Zap/52-1","","","DOCUMENTATION","","1178736475.7" ,"","" ;;; 152 channel
created to ring 152.
(152 ringing)
"EV_ANSWER","2007-05-09 12:47:58","","152","151","","","152","extension","Zap/52-1","AppDial","(Outgoing
Line)","DOCUMENTATION","","1178736475.7","","" ;;; 3 seconds later, 152 answers
"EV_ANSWER","2007-05-09
12:47:58","fxs.51","151","151","","","152","extension","Zap/51-2","Dial","Zap/52|30|TtWw","DOCUMENTATION","","1178736471.6","
","" ;;; ... and 151-2 also answers
"EV_BRIDGE_START","2007-05-09
12:47:59","fxs.51","151","151","","","152","extension","Zap/51-2","Dial","Zap/52|30|TtWw","DOCUMENTATION","","1178736471.6","
","Zap/51-1" ;;; 1 second later, bridge formed betw. 151-2 and 151 (152 answers, 151 and 152 convering; 150 is listening to
silence; 151 hits flash again... to start a 3way)
"EV_3WAY_START","2007-05-09 12:48:58","","151","152","","","","extension","Zap/51-1","Bridged
Call","Zap/50-1","DOCUMENTATION","","1178736378.4","","Zap/51-2" ;;; another hook-flash to begin a 3-way conference
"EV_BRIDGE_END","2007-05-09
12:48:58","fxs.50","150","150","","","701","extension","Zap/50-1","ParkedCall","701","DOCUMENTATION","","1178736428.5","","Za
p/51-1" ;;; - almost 1 minute later, the bridge ends (151 flashes hook again)
"EV_BRIDGE_START","2007-05-09
12:48:58","fxs.50","150","150","","","701","extension","Zap/50-1","ParkedCall","701","DOCUMENTATION","","1178736428.5","","Za
p/51-1" ;;; 0-second bridge at 150. (3 way conf formed)
"EV_BRIDGE_END","2007-05-09
12:48:58","fxs.50","150","150","","","701","extension","Zap/50-1","ParkedCall","701","DOCUMENTATION","","1178736428.5","","Za
p/51-1" ;;;
"EV_BRIDGE_START","2007-05-09
12:48:58","fxs.50","150","150","","","701","extension","Zap/50-1","ParkedCall","701","DOCUMENTATION","","1178736428.5","","Za
p/51-1" ;;; bridge starts for 150
(3way now, then 151 hangs up.)
"EV_BRIDGE_END","2007-05-09
12:49:26","fxs.50","150","150","","","701","extension","Zap/50-1","ParkedCall","701","DOCUMENTATION","","1178736428.5","","Za
p/51-1" ;;; 28 seconds later, bridge ends
"EV_HANGUP","2007-05-09 12:49:26","","151","152","","","","extension","Zap/51-1","Bridged
Call","Zap/50-1","DOCUMENTATION","","1178736378.4","","" ;;; 151 hangs up, leaves 150 and 152 connected
"EV_CHAN_END","2007-05-09 12:49:26","","151","152","","","","extension","Zap/51-1","Bridged
Call","Zap/50-1","DOCUMENTATION","","1178736378.4","","" ;;; 151 channel ends
"EV_CHAN_END","2007-05-09
12:49:26","fxs.51","151","151","","","h","extension","Zap/51-2ZOMBIE","","","DOCUMENTATION","","1178736428.5","","" ;;; 152-2
channel ends (zombie)
(just 150 and 152 now)
"EV_BRIDGE_END","2007-05-09
12:50:13","fxs.50","150","150","","","152","extension","Zap/50-1","Dial","Zap/52|30|TtWw","DOCUMENTATION","","1178736471.6","
","" ;;; 47 sec later, the bridge from 150 to 152 ends
"EV_HANGUP","2007-05-09 12:50:13","","152","151","","","","extension","Zap/52-1","Bridged
Call","Zap/50-1","DOCUMENTATION","","1178736475.7","","" ;;; 152 hangs up
"EV_CHAN_END","2007-05-09 12:50:13","","152","151","","","","extension","Zap/52-1","Bridged
```

```
Call","Zap/50-1","DOCUMENTATION","","1178736475.7","","" ;;; 152 channel ends
"EV_HANGUP","2007-05-09
12:50:13","fxs.50","150","150","","","h","extension","Zap/50-1","","","DOCUMENTATION","","1178736471.6","","" ;;; 150 hangs
up
```

```
"EV_CHAN_END","2007-05-09
12:50:13","fxs.50","150","150","","","h","extension","Zap/50-1","","","DOCUMENTATION","","1178736471.6","","" ;;; 150 ends
```

In terms of Manager events, the above Events correspond to the following 80 Manager events shown in the expansion link.
Click here to expand...

```
Event: Newchannel
Privilege: call,all
Channel: Zap/52-1
State: Rsrvd
CallerIDNum: 152
CallerIDName: fxs.52
Uniqueid: 1178801102.5

Event: Newcallerid
Privilege: call,all
Channel: Zap/52-1
CallerIDNum: 152
CallerIDName: fxs.52
Uniqueid: 1178801102.5
CID-CallingPres: 0 (Presentation Allowed, Not Screened)
Event: Newcallerid
Privilege: call,all
Channel: Zap/52-1
CallerIDNum: 152
CallerIDName: fxs.52
Uniqueid: 1178801102.5
CID-CallingPres: 0 (Presentation Allowed, Not Screened)

Event: Newstate
Privilege: call,all
Channel: Zap/52-1
State: Ring
CallerIDNum: 152
CallerIDName: fxs.52
Uniqueid: 1178801102.5
Event: Newexten
Privilege: call,all
Channel: Zap/52-1
Context: extension
Extension: 151
Priority: 1
Application: Set
AppData: CDR(myvar)=zingo
Uniqueid: 1178801102.5
Event: Newexten
Privilege: call,all
Channel: Zap/52-1
Context: extension
Extension: 151
Priority: 2
Application: Dial
AppData: Zap/51|30|TtWw
Uniqueid: 1178801102.5

Event: Newchannel
Privilege: call,all
Channel: Zap/51-1
State: Rsrvd
CallerIDNum: 151
CallerIDName: fxs.51
Uniqueid: 1178801108.6
Event: Newstate
Privilege: call,all
Channel: Zap/51-1
State: Ringing
CallerIDNum: 152
CallerIDName: fxs.52
Uniqueid: 1178801108.6

Event: Dial
Privilege: call,all
SubEvent: Begin
Source: Zap/52-1
Destination: Zap/51-1
CallerIDNum: 152
CallerIDName: fxs.52
SrcUniqueID: 1178801102.5
DestUniqueID: 1178801108.6
Event: Newcallerid
Privilege: call,all
Channel: Zap/51-1
CallerIDNum: 151
CallerIDName: <Unknown>
```

```
Uniqueid: 1178801108.6
CID-CallingPres: 0 (Presentation Allowed, Not Screened)

Event: Newstate
Privilege: call,all
Channel: Zap/52-1
State: Ringing
CallerIDNum: 152
CallerIDName: fxs.52
Uniqueid: 1178801102.5
Event: Newstate
Privilege: call,all
Channel: Zap/51-1
State: Up
CallerIDNum: 151
CallerIDName: <unknown>
Uniqueid: 1178801108.6
Event: Newstate
Privilege: call,all
Channel: Zap/52-1
State: Up
CallerIDNum: 152
CallerIDName: fxs.52
Uniqueid: 1178801102.5

Event: Link
Privilege: call,all
Channel1: Zap/52-1
Channel2: Zap/51-1
Uniqueid1: 1178801102.5
Uniqueid2: 1178801108.6
CallerID1: 152
CallerID2: 151
Event: Unlink
Privilege: call,all
Channel1: Zap/52-1
Channel2: Zap/51-1
Uniqueid1: 1178801102.5
Uniqueid2: 1178801108.6
CallerID1: 152
CallerID2: 151

Event: Link
Privilege: call,all
Channel1: Zap/52-1
Channel2: Zap/51-1
Uniqueid1: 1178801102.5
Uniqueid2: 1178801108.6
CallerID1: 152
CallerID2: 151
Event: Unlink
Privilege: call,all
Channel1: Zap/52-1
Channel2: Zap/51-1
Uniqueid1: 1178801102.5
Uniqueid2: 1178801108.6
CallerID1: 152
CallerID2: 151

Event: ParkedCall
Privilege: call,all
Exten: 701
Channel: Zap/51-1
From: Zap/52-1
Timeout: 45
CallerIDNum: 151
CallerIDName: <unknown>
Event: Dial
Privilege: call,all
SubEvent: End
Channel: Zap/52-1
DialStatus: ANSWER

Event: Newexten
Privilege: call,all
Channel: Zap/52-1
Context: extension
Extension: h
Priority: 1
Application: Goto
AppData: label1
Uniqueid: 1178801102.5
Event: Newexten
Privilege: call,all
Channel: Zap/52-1
Context: extension
Extension: h
Priority: 4
Application: Goto
```

```
AppData: label2
Uniqueid: 1178801102.5

Event: Newexten
Privilege: call,all
Channel: Zap/52-1
Context: extension
Extension: h
Priority: 2
Application: NoOp
AppData: In Hangup! myvar is zingo and accountcode is billsec is 26 and duration is 40 and end is 2007-05-10 06:45:42.
Uniqueid: 1178801102.5
Event: Newexten
Privilege: call,all
Channel: Zap/52-1
Context: extension
Extension: h
Priority: 3
Application: Goto
AppData: label3
Uniqueid: 1178801102.5

Event: Newexten
Privilege: call,all
Channel: Zap/52-1
Context: extension
Extension: h
Priority: 5
Application: NoOp
AppData: More Hangup message after hopping around"
Uniqueid: 1178801102.5
Event: Hangup
Privilege: call,all
Channel: Zap/52-1
Uniqueid: 1178801102.5
Cause: 16
Cause-txt: Normal Clearing

Event: Newchannel
Privilege: call,all
Channel: Zap/50-1
State: Rsrvd
CallerIDNum: 150
CallerIDName: fxs.50
Uniqueid: 1178801162.7
Event: Newcallerid
Privilege: call,all
Channel: Zap/50-1
CallerIDNum: 150
CallerIDName: fxs.50
Uniqueid: 1178801162.7
CID-CallingPres: 0 (Presentation Allowed, Not Screened)

Event: Newcallerid
Privilege: call,all
Channel: Zap/50-1
CallerIDNum: 150
CallerIDName: fxs.50
Uniqueid: 1178801162.7
CID-CallingPres: 0 (Presentation Allowed, Not Screened)
Event: Newstate
Privilege: call,all
Channel: Zap/50-1
State: Ring
CallerIDNum: 150
CallerIDName: fxs.50
Uniqueid: 1178801162.7

Event: Newexten
Privilege: call,all
Channel: Zap/50-1
Context: extension
Extension: 701
Priority: 1
Application: ParkedCall
AppData: 701
Uniqueid: 1178801162.7
Event: UnParkedCall
Privilege: call,all
Exten: 701
Channel: Zap/51-1
From: Zap/50-1
CallerIDNum: 151
CallerIDName: <unknown>
Event: Newstate
Privilege: call,all
Channel: Zap/50-1
State: Up
CallerIDNum: 150
```

```
CallerIDName: fxs.50
Uniqueid: 1178801162.7

Event: Link
Privilege: call,all
Channel1: Zap/50-1
Channel2: Zap/51-1
Uniqueid1: 1178801162.7
Uniqueid2: 1178801108.6
CallerID1: 150
CallerID2: 151
Event: Newchannel
Privilege: call,all
Channel: Zap/51-2
State: Rsrvd
CallerIDNum: 151
CallerIDName: fxs.51
Uniqueid: 1178801218.8

Event: Unlink
Privilege: call,all
Channel1: Zap/50-1
Channel2: Zap/51-1
Uniqueid1: 1178801162.7
Uniqueid2: 1178801108.6
CallerID1: 150
CallerID2: 151
Event: Link
Privilege: call,all
Channel1: Zap/50-1
Channel2: Zap/51-1
Uniqueid1: 1178801162.7
Uniqueid2: 1178801108.6
CallerID1: 150
CallerID2: 151

Event: Unlink
Privilege: call,all
Channel1: Zap/50-1
Channel2: Zap/51-1
Uniqueid1: 1178801162.7
Uniqueid2: 1178801108.6
CallerID1: 150
CallerID2: 151
Event: Link
Privilege: call,all
Channel1: Zap/50-1
Channel2: Zap/51-1
Uniqueid1: 1178801162.7
Uniqueid2: 1178801108.6
CallerID1: 150
CallerID2: 151
Event: Newcallerid
Privilege: call,all
Channel: Zap/51-2
CallerIDNum: 151
CallerIDName: fxs.51
Uniqueid: 1178801218.8
CID-CallingPres: 0 (Presentation Allowed, Not Screened)

Event: Newcallerid
Privilege: call,all
Channel: Zap/51-2
CallerIDNum: 151
CallerIDName: fxs.51
Uniqueid: 1178801218.8
CID-CallingPres: 0 (Presentation Allowed, Not Screened)
Event: Newstate
Privilege: call,all
Channel: Zap/51-2
State: Ring
CallerIDNum: 151
CallerIDName: fxs.51
Uniqueid: 1178801218.8

Event: Newexten
Privilege: call,all
Channel: Zap/51-2
Context: extension
Extension: 152
Priority: 1
Application: Set
AppData: CDR(myvar)=zingo
Uniqueid: 1178801218.8
Event: Newexten
Privilege: call,all
Channel: Zap/51-2
Context: extension
Extension: 152
```

```
Priority: 2
Application: Dial
AppData: Zap/52|30|TtWw
Uniqueid: 1178801218.8

Event: Newchannel
Privilege: call,all
Channel: Zap/52-1
State: Rsrvd
CallerIDNum: 152
CallerIDName: fxs.52
Uniqueid: 1178801223.9
Event: Newstate
Privilege: call,all
Channel: Zap/52-1
State: Ringing
CallerIDNum: 151
CallerIDName: fxs.51
Uniqueid: 1178801223.9
Event: Dial
Privilege: call,all
SubEvent: Begin
Source: Zap/51-2
Destination: Zap/52-1
CallerIDNum: 151
CallerIDName: fxs.51
SrcUniqueID: 1178801218.8
DestUniqueID: 1178801223.9

Event: Newcallerid
Privilege: call,all
Channel: Zap/52-1
CallerIDNum: 152
CallerIDName: <Unknown>
Uniqueid: 1178801223.9
CID-CallingPres: 0 (Presentation Allowed, Not Screened)
Event: Newstate
Privilege: call,all
Channel: Zap/51-2
State: Ringing
CallerIDNum: 151
CallerIDName: fxs.51
Uniqueid: 1178801218.8

Event: Newstate
Privilege: call,all
Channel: Zap/52-1
State: Up
CallerIDNum: 152
CallerIDName: <unknown>
Uniqueid: 1178801223.9
Event: Newstate
Privilege: call,all
Channel: Zap/51-2
State: Up
CallerIDNum: 151
CallerIDName: fxs.51
Uniqueid: 1178801218.8

Event: Link
Privilege: call,all
Channel1: Zap/51-2
Channel2: Zap/52-1
Uniqueid1: 1178801218.8
Uniqueid2: 1178801223.9
CallerID1: 151
CallerID2: 152
Event: Unlink
Privilege: call,all
Channel1: Zap/50-1
Channel2: Zap/51-1
Uniqueid1: 1178801162.7
Uniqueid2: 1178801108.6
CallerID1: 150
CallerID2: 151

Event: Link
Privilege: call,all
Channel1: Zap/50-1
Channel2: Zap/51-1
Uniqueid1: 1178801162.7
Uniqueid2: 1178801108.6
CallerID1: 150
CallerID2: 151
Event: Unlink
Privilege: call,all
Channel1: Zap/50-1
Channel2: Zap/51-1
Uniqueid1: 1178801162.7
```

```
Uniqueid2: 1178801108.6
CallerID1: 150
CallerID2: 151

Event: Link
Privilege: call,all
Channel1: Zap/50-1
Channel2: Zap/51-1
Uniqueid1: 1178801162.7
Uniqueid2: 1178801108.6
CallerID1: 150
CallerID2: 151
Event: Unlink
Privilege: call,all
Channel1: Zap/50-1
Channel2: Zap/51-1
Uniqueid1: 1178801162.7
Uniqueid2: 1178801108.6
CallerID1: 150
CallerID2: 151

Event: Hangup
Privilege: call,all
Channel: Zap/51-1
Uniqueid: 1178801108.6
Cause: 16
Cause-txt: Normal
Clearing
Event: Newexten
Privilege: call,all
Channel: Zap/50-1
Context: extension
Extension: h
Priority: 1
Application: Goto
AppData: label1
Uniqueid: 1178801162.7
Event: Newexten
Privilege: call,all
Channel: Zap/50-1
Context: extension
Extension: h
Priority: 4
Application: Goto
AppData: label2
Uniqueid: 1178801162.7

Event: Newexten
Privilege: call,all
Channel: Zap/50-1
Context: extension
Extension: h
Priority: 2
Application: NoOp
AppData: In Hangup! myvar is and accountcode is billsec is 0 and duration is 0 and end is 2007-05-10 06:48:37.
Uniqueid: 1178801162.7
Event: Newexten
Privilege: call,all
Channel: Zap/50-1
Context: extension
Extension: h
Priority: 3
Application: Goto
AppData: label3
Uniqueid: 1178801162.7

Event: Newexten
 Privilege: call,all
Channel: Zap/50-1
Context: extension
Extension: h
Priority: 5
Application: NoOp
AppData: More
Hangup message after hopping around"
Uniqueid: 1178801162.7

Event: Masquerade
Privilege: call,all
Clone: Zap/50-1
CloneState: Up
Original: Zap/51-2
OriginalState: Up
Event: Rename
Privilege: call,all
Oldname: Zap/50-1
Newname: Zap/50-1<MASQ>
Uniqueid: 1178801162.7
```

```
Event: Rename
Privilege: call,all
 Oldname: Zap/51-2
Newname: Zap/50-1
Uniqueid: 1178801218.8
Event: Rename
Privilege: call,all
Oldname: Zap/50-1<MASQ>
Newname: Zap/51-2<ZOMBIE>
Uniqueid: 1178801162.7
Event: Hangup
Privilege: call,all
Channel: Zap/51-2<ZOMBIE>
 Uniqueid: 1178801162.7
Cause: 0
Cause-txt: Unknown

Event: Unlink
Privilege: call,all
Channel1: Zap/50-1
Channel2: Zap/52-1
Uniqueid1: 1178801218.8
Uniqueid2: 1178801223.9
CallerID1: 150
CallerID2: 152
Event: Hangup
Privilege: call,all
Channel: Zap/52-1
Uniqueid: 1178801223.9
Cause: 16
Cause-txt: Normal Clearing

Event: Dial
Privilege: call,all
SubEvent: End
Channel: Zap/50-1
DialStatus: ANSWER
Event: Newexten
Privilege: call,all
Channel: Zap/50-1
Context: extension
Extension: h
Priority: 1
Application: Goto
AppData: label1
Uniqueid: 1178801218.8

Event: Newexten
Privilege: call,all
Channel: Zap/50-1
Context: extension
Extension: h
Priority: 4
Application: Goto
AppData: label2
Uniqueid: 1178801218.8
Event: Newexten
Privilege: call,all
Channel: Zap/50-1
Context: extension
Extension: h
Priority: 2
Application: NoOp
AppData: In Hangup! myvar is and accountcode is billsec is 90 and duration is 94 and end is 2007-05-10 06:48:37.
Uniqueid: 1178801218.8

Event: Newexten
Privilege: call,all
Channel: Zap/50-1
Context: extension
Extension: h
Priority: 3
Application: Goto
AppData: label3
Uniqueid: 1178801218.8
Event: Newexten
Privilege: call,all
Channel: Zap/50-1
Context: extension
Extension: h
Priority: 5
Application: NoOp
AppData: More Hangup message after hopping around"
Uniqueid: 1178801218.8
Event: Hangup
Privilege: call,all
Channel: Zap/50-1
Uniqueid: 1178801218.8
```

```
Cause: 16
Cause-txt: Normal Clearing
```

And, humorously enough, the above 80 manager events, or 42 CEL events, correspond to the following two CDR records (at the moment!):

```
""fxs.52" 152","152","h","extension","Zap/52-1","Zap/51-1","NoOp","More Hangup message after hopping around"","2007-05-09
17:35:56","2007-05-09 17:36:20","2007-05-09 17:36:36","40","16","ANSWERED","DOCUMENTATION","","1178753756.0",""
""fxs.50" 150","150","152","extension","Zap/50-1","Zap/51-1","NoOp","More Hangup message after hopping around"","2007-05-09
17:37:59","2007-05-09 17:38:06","2007-05-09 17:39:11","72","65","ANSWERED","DOCUMENTATION","","1178753871.3",""
```

CEL Applications and Functions

Applications

The CELGenUserEvent application allows you to instruct Asterisk to generate a user defined event with custom type name.

The event triggered is the **`USER_DEFINED`** event as listed in the Asterisk 12 CEL Specification. The **`eventtype`** and **`userdeftype`** fields will be populated with data passed through the respective arguments provided to the CELGenUserEvent application.

> ! Please note that there is no restrictions on the name supplied. If it happens to match a standard CEL event name, it will look like that event was generated. This could be a blessing or a curse!

Functions

Most CEL fields are populated by common channel data, so a unique function is not required to read or write that data on the channel. That channel data is already available via the CHANNEL function in currently supported versions of Asterisk.

Older versions of Asterisk had a unique CEL function. You can run "core show function CEL" to see if you have this function and display the help text.

CEL Configuration Files

CEL General Configuration

Primary CEL configuration settings are located in `cel.conf`. Note that CEL only publishes record types to back-ends that are enabled in the general CEL configuration.

Back-end Configuration

Configuration for specific logging or storage back-ends is located in separate configuration files. The exception is the AMI and RADIUS back-ends. Sample configurations are provided with the Asterisk 12 source for all of these back-ends.

Name	Config File	Description
Manager (AMI)	cel.conf	The manager CEL output module publishes records over AMI as CEL events with the record type published under the "EventName" key. This module is configured in cel.conf in the [manager] section.
RADIUS	cel.conf	The RADIUS CEL output module allows the CEL engine to publish records to a RADIUS server. This module is configured in cel.conf in the [radius] section.
CEL Custom	cel_custom.conf	The Custom CEL output module provides logging capability to a CSV file in a format described in the configuration file.
ODBC	cel_odbc.conf	The ODBC CEL output module provides logging capability to any ODBC-compatible database.
PGSQL	cel_pgsql.conf	The PGSQL CEL output module provides logging capability to PostgreSQL databases when it is desirable to avoid the ODBC abstraction layer.
SQLite	cel_sqlite3_custom.conf	The SQLite CEL output module provides logging capability to a SQLite3 database in a format described in its configuration file.
TDS	cel_tds.conf	The TDS CEL output module provides logging capability to Sybase or Microsoft SQL Server databases when it is desirable to avoid the ODBC abstraction layer.

CEL Configuration Examples

The child pages in this section provide examples of configuration with the specific logging or storage back-ends mentioned in CEL Configuration Files.

MSSQL CEL Backend

Asterisk can currently store Channel Events into an MSSQL database in two different ways: cel_odbc or cel_tds

Channel Event Records can be stored using unixODBC (which requires the FreeTDS package) cel_odbc or directly by using just the FreeTDS package cel_tds

The following provide some examples known to get asterisk working with mssql.

> Only choose one db connector.

ODBC using cel_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 cel_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 (cel_odbc) and FreeTDS (cel_tds). This command will erase the contents of cel_tds.conf
>
> ```
> [ -f /etc/asterisk/cel_tds.conf ] > /etc/asterisk/cel_tds.conf
> ```

> unixODBC requires the freeTDS package, but asterisk does not call freeTDS directly.

Now set up cel_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/cel_odbc.conf

```
[global]
dsn=MSSQL-asterisk
username=voipdbuser
password=voipdbpass
loguniqueid=yes
```

And finally, create the 'cel' table in your mssql database.

```
CREATE TABLE cel (
        [eventtype] [varchar] (30) NOT NULL ,
        [eventtime] [datetime] NOT NULL ,
        [cidname] [varchar] (80) NOT NULL ,
        [cidnum] [varchar] (80) NOT NULL ,
        [cidani] [varchar] (80) NOT NULL ,
        [cidrdnis] [varchar] (80) NOT NULL ,
        [ciddnid] [varchar] (80) NOT NULL ,
        [exten] [varchar] (80) NOT NULL ,
        [context] [varchar] (80) NOT NULL ,
        [channame] [varchar] (80) NOT NULL ,
        [appname] [varchar] (80) NOT NULL ,
        [appdata] [varchar] (80) NOT NULL ,
        [amaflags] [int] NOT NULL ,
        [accountcode] [varchar] (20) NOT NULL ,
        [uniqueid] [varchar] (32) NOT NULL ,
        [peer] [varchar] (80) 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 desired channel event at the moment it occurs.

FreeTDS, using cel_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 cel_tds.

```
make clean && ./configure --with-tds && make update && make && make install
```

Only install one database connector. Do not confuse asterisk by using both ODBC (cel_odbc) and FreeTDS (cel_tds). This command will erase the contents of cel_odbc.conf

```
[ -f /etc/asterisk/cel_odbc.conf ] > /etc/asterisk/cel_odbc.conf
```

Setup cel_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/cel_tds.conf

```
[global]
hostname=192.168.1.25
port=1433
dbname=voipdb
user=voipdbuser
password=voipdpass
charset=BIG5
```

And finally, create the 'cel' table in your mssql database.

```
CREATE TABLE cel (
        [eventtype] [varchar] (30) NULL ,
        [eventtime] [datetime] NULL ,
        [cidname] [varchar] (80) NULL ,
        [cidnum] [varchar] (80) NULL ,
        [cidani] [varchar] (80) NULL ,
        [cidrdnis] [varchar] (80) NULL ,
        [ciddnid] [varchar] (80) NULL ,
        [exten] [varchar] (80) NULL ,
        [context] [varchar] (80) NULL ,
        [channame] [varchar] (80) NULL ,
        [appname] [varchar] (80) NULL ,
        [appdata] [varchar] (80) NULL ,
        [amaflags] [varchar] (16) NULL ,
        [accountcode] [varchar] (20) NULL ,
        [uniqueid] [varchar] (32) NULL ,
        [userfield] [varchar] (255) NULL ,
        [peer] [varchar] (80) 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.

PostgreSQL CEL Backend

If you want to go directly to postgresql database, and have the cel_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 cel_pgsql.conf file or create your own.

Here is a sample:

/etc/asterisk/cel_pgsql.conf

```
; Sample Asterisk config file for CEL logging to PostgresSQL
[global]
hostname=localhost
port=5432
dbname=asterisk
password=password
user=postgres
table=cel
```

Now create a table in postgresql for your cels

```
CREATE TABLE cel (
        id serial ,
        eventtype varchar (30) NOT NULL ,
        eventtime timestamp NOT NULL ,
        userdeftype varchar(255) NOT NULL ,
        cid_name varchar (80) NOT NULL ,
        cid_num varchar (80) NOT NULL ,
        cid_ani varchar (80) NOT NULL ,
        cid_rdnis varchar (80) NOT NULL ,
        cid_dnid varchar (80) NOT NULL ,
        exten varchar (80) NOT NULL ,
        context varchar (80) NOT NULL ,
        channame varchar (80) NOT NULL ,
        appname varchar (80) NOT NULL ,
        appdata varchar (80) NOT NULL ,
        amaflags int NOT NULL ,
        accountcode varchar (20) NOT NULL ,
        peeraccount varchar (20) NOT NULL ,
        uniqueid varchar (150) NOT NULL ,
        linkedid varchar (150) NOT NULL ,
        userfield varchar (255) NOT NULL ,
        peer varchar (80) NOT NULL
);
```

RADIUS CEL Backend

What is needed

- FreeRADIUS server
- Radiusclient-ng library
- Asterisk PBX

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"

⚠ 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/cel.h for all the fields which are recorded. By default, records in comma separated values will be created in /var/log/asterisk/cel-csv.

The configuration file for cel_radius.so module is :
/etc/asterisk/cel.conf This is where you can set CEL related parameters as well as the path to the radiusclient-ng library configuration file.

This is where you can set CDR related parameters as well as the path to the radiusclient-ng library configuration file.
Logged Values

- "Asterisk-Acc-Code", The account name of detail records
- "Asterisk-CidName",
- "Asterisk-CidNum",
- "Asterisk-Cidani",
- "Asterisk-Cidrdnis",
- "Asterisk-Ciddnid",
- "Asterisk-Exten",
- "Asterisk-Context", The destination context
- "Asterisk-Channame", The channel name
- "Asterisk-Appname", Last application run on the channel
- "Asterisk-App-Data", Argument to the last channel
- "Asterisk-Event-Time",
- "Asterisk-Event-Type",
- "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 SetCELUserField
- "Asterisk-Peer" Name of the Peer for 2-channel events (like bridge)

Interfaces

Overview

There are many ways to interface Asterisk with scripts, other applications or storage systems. From the very trivial, such as using Asterisk Call Files, to sophisticated APIs such as the Asterisk Rest Interface. The pages in this section cover many of Asterisk's built-in interfaces, all of which provide some aspect of control, monitoring or storage.

In this Section

Interface	Description
Asterisk Call Files	Asterisk can initiate calls based on information provided via flat text files in a spool directory. Asterisk can operate on these as soon as the file is inside the directory, or in the future depending on the timestamp of the file.
Asterisk Gateway Interface	AGI provides an interface between the Asterisk dialplan and an external program (via pipes, stdin and stdout) that wants to manipulate a channel in the dialplan.
Asterisk Management Interface*****	AMI is intended for management type functions. The manager is a client/server model over TCP. With the AMI you'll be able to control the PBX, originate calls, check mailbox status, monitor channels and queues as well as execute Asterisk commands.
Asterisk REST Interface*****	ARI is an asynchronous API that allows developers to build communications applications by exposing the raw primitive objects in Asterisk - channels, bridges, endpoints, media, etc. - through an intuitive REST interface. The state of the objects being controlled by the user are conveyed via JSON events over a WebSocket.
Calendaring	Asterisk's calendaring module allows read and write communication with various standardized calendar technologies. Asterisk dialplan can make use of calendar event information.
Database Connectivity	Asterisk has core support for ODBC connectivity, and many Asterisk modules provide support for a variety of back-end database connectivity, such as for MySQL or PostgreSQL.
Distributed Device State	Asterisk provides a few ways of distributing the state of devices between multiple Asterisk instances, whether on the same system or multiple systems.
SNMP	Basic SNMP support is included with Asterisk. This allows monitoring of a variety of Asterisk activity.
Speech Recognition API	The dialplan speech recognition API is based around a single speech utilities application file, which exports many applications to be used for speech recognition. These include an application to prepare for speech recognition, activate a grammar, and play back a sound file while waiting for the person to speak. Using a combination of these applications you can easily make a dialplan use speech recognition without worrying about what speech recognition engine is being used.
StatsD	This StatsD application is a dialplan application that is used to send statistics automatically whenever a call is made to an extension that employs the application. The user must provide the arguments to the application in the dialplan, but after that, the application will send statistics to StatsD without requiring the user to perform anymore actions whenever a call comes through that extension.

***** Event ordering is not guaranteed. Applications monitoring events from these interfaces should be aware that the order between received events is not assured unless otherwise, and elsewhere specified. For example, an event monitoring application may receive the following events: A->B->C. However, later it might receive those same events, but in a different order: B->A->C

Asterisk Calendaring

The Asterisk Calendaring API is provided by the res_calendar module. It aims to be a generic interface for integrating Asterisk with various calendaring technologies. The goal is to be able to support reading and writing of calendar events as well as allowing notification of pending events through the Asterisk dialplan.

There are four calendaring modules that ship with Asterisk that provide support for the following calendaring servers.

Calendar Server Support	Module Name
iCalendar	res_calendar_icalendar.so
CalDAV	res_calender_caldav.so
Microsoft Exchange Server	res_calendar_exchange.so
Microsoft Exchange Web Services	res_calendar_ews.so

All four modules support event notification. Both CalDAV and Exchange support reading and writing calendars, while iCalendar is a read-only format.

You can see list all registered calendar types at the CLI with **"calendar show types"**.

```
*CLI> calendar show types
Type       Description
caldav     CalDAV calendars
exchange   MS Exchange calendars
ews        MS Exchange Web Service calend
ical       iCalendar .ics calendars
```

Configuring Asterisk Calendaring

The calendar.conf file

All asterisk calendaring modules are configured through calendar.conf. Each calendar module can define its own set of required parameters in addition to the parameters available to all calendar types. An effort has been made to keep all options the same in all calendaring modules, but some options will diverge over time as features are added to each module.

An example calendar.conf might look like:

```
[calendar_joe]
type = ical
url = https://example.com/home/jdoe/Calendar
user = jdoe
secret = mysecret
refresh = 15
timeframe = 600
autoreminder = 10
channel = SIP/joe
context = calendar_event_notify
extension = s
waittime = 30
```

Module-independent settings

The settings related to calendar event notification are handled by the core calendaring API. These settings are:

- autoreminder - This allows the overriding of any alarms that may or may not be set for a calendar event. It is specified in minutes.
- refresh - How often to refresh the calendar data; specified in minutes.
- timeframe - How far into the future each calendar refresh should look. This is the amount of data that will be visible to queries from the dialplan. This setting should always be greater than or equal to the refresh setting or events may be missed. It is specified in minutes.
- channel - The channel that should be used for making the notification attempt.
- waittime - How long to wait, in seconds, for the channel to answer a notification attempt. There are two ways to specify how to handle a notification. One option is providing a context and extension, while the other is providing an application and the arguments to that application. One (and only one) of these options should be provided.
- context - The context of the extension to connect to the notification channel
- extension - The extension to connect to the notification. Note that the priority will always be 1.
- app - The dialplan application to execute upon the answer of a notification
- appdata - The data to pass to the notification dialplan application

Module-dependent settings

Connection-related options are specific to each module. Currently, all modules take a url, user, and secret for configuration and no other module-specific settings have been implemented. At this time, no support for HTTP redirects has been implemented, so it is important to specify the correct URL-paying attention to any trailing slashes that may be necessary.

Specific Examples

Google Calendar

Requirements:

- The res_calendar_icalendar.so module must be loaded and running.
- You must have a Google calendar!
- Google requires that you share the calendar publicly. However you can choose to only share busy/free information to limit exposure to

details.
- Configure Asterisk to connect to the specific public calendar.

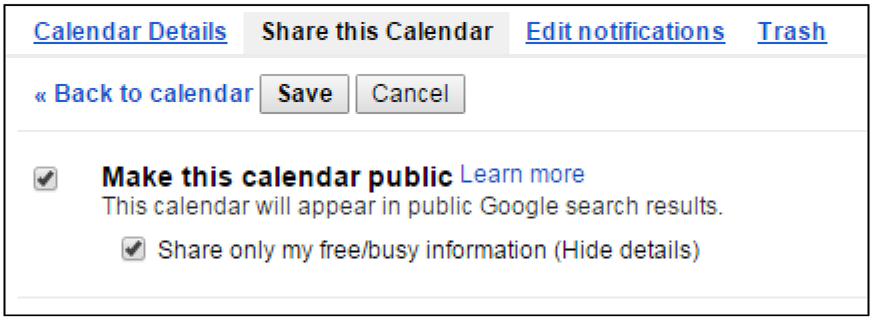
In this example we'll configure Asterisk to connect to your Google Calendar via the ical type calendar.

Sharing your Google calendar

For detailed instructions see Google's answer page.

We'll assume you are familiar with Google's interface and provide some brief instructions here.

Go to your Google calendar settings, navigate to a specific calendar and open the **Share this Calendar** tab.

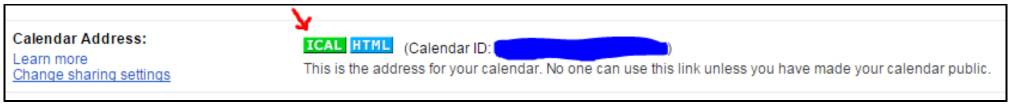


☑ **Make this calendar public** Learn more
This calendar will appear in public Google search results.
☑ Share only my free/busy information (Hide details)

Make the calendar public and choose save.

Get the Google calendar address

Go to the **Calendar Details** tab and find the button for getting the public **ical** address.



Calendar Address: (Calendar ID: [redacted])
This is the address for your calendar. No one can use this link unless you have made your calendar public.

Configure Asterisk with Google Calendar details

This specific configuration isn't too different than the generic example. Your calendar address goes in the **url** field.

```
[gcal1]
type = ical
url = https://calendar.google.com/calendar/ical/example%40gmail.com/public/basic.ics
user = example@gmail.com
secret = a_very_secure_password
refresh = 15
timeframe = 60
```

Once you have a configuration you can startup Asterisk or else reload the modules. After this you can check to see if the calendar is being read. Use the commands **"calendar show calendars"** and **"calendar show calendar <calendar name>"**

```
CentOSLab*CLI> calendar show calendars
Calendar             Type       Status
--------             ----       ------
gcal1                ical       busy
```

```
CentOSLab*CLI> calendar show calendar gcal1
Name              : gcal1
Notify channel    :
Notify context    :
Notify extension  :
Notify applicatio :
Notify appdata    :
Refresh time      : 15
Timeframe         : 60
Autoreminder      : 0
Events
------
Summary     : Busy
Description :
Organizer   :
Location    :
Categories  :
Priority    : 0
UID         : 6b6ikcvq165i470lq5sdm7r1v4@google.com
Start       : 2015-11-25 05:40:00 PM -0600
End         : 2015-11-25 06:10:00 PM -0600
Alarm       :
```

The output should reflect your calendar settings and if it is reading from the calendar server you should see events that are present (on your calendar) within the configured Timeframe. If you don't see any events then go to your Google calendar and create an event within the timeframe. Save that event, then wait at least the Refresh time before checking the commands again to see if the event shows up.

Calendaring Dialplan Functions

Read functions

The simplest dialplan query is the CALENDAR_BUSY query. It takes a single option, the name of the calendar defined, and returns "1" for busy (including tentatively busy) and "0" for not busy.

For more information about a calendar event, a combination of CALENDAR_QUERY and CALENDAR_QUERY_RESULT is used. CALENDAR_QUERY takes the calendar name and optionally a start and end time in "unix time" (seconds from unix epoch). It returns an id that can be passed to CALENDAR_QUERY_RESULT along with a field name to return the data in that field. If multiple events are returned in the query, the number of the event in the list can be specified as well. The available fields to return are:

- **summary** - A short summary of the event
- **description** - The full description of the event
- **organizer** - Who organized the event
- **location** - Where the event is located
- **calendar** - The name of the calendar from calendar.conf
- **uid** - The unique identifier associated with the event
- **start** - The start of the event in seconds since Unix epoch
- **end** - The end of the event in seconds since Unix epoch
- **busystate** - The busy state 0=Free, 1=Tentative, 2=Busy
- **attendees** - A comma separated list of attendees as stored in the event and may include prefixes such as "mailto:".

When an event notification is sent to the dial plan, the CALENDAR_EVENT function may be used to return the information about the event that is causing the notification. The fields that can be returned are the same as those from CALENDAR_QUERY_RESULT.

Write functions

To write an event to a calendar, the CALENDAR_WRITE function is used. This function takes a calendar name and also uses the same fields as CALENDAR_QUERY_RESULT. As a write function, it takes a set of comma-separated values that are in the same order as the specified fields. For example:

```
CALENDAR_WRITE(mycalendar,summary,organizer,start,end,busystate)= "My event","mailto:jdoe@example.com",228383580,228383640,1)
```

Calendaring Dialplan Examples

Office hours

A common business PBX scenario is would be executing dialplan logic based on when the business is open and the phones staffed. If the business is closed for holidays, it is sometimes desirable to play a message to the caller stating why the business is closed.

The standard way to do this in asterisk has been doing a series of GotoIfTime statements or time-based include statements. Either way can be tedious and requires someone with access to edit asterisk config files.

With calendaring, the administrator only needs to set up a calendar that contains the various holidays or even recurring events specifying the office hours. A custom greeting filename could even be contained in the description field for playback. For example:

```
[incoming]
exten => 5555551212,1,Answer
 same => n,GotoIf(${CALENDAR_BUSY(officehours)}?closed:attendant,s,1)
 same => n(closed),Set(id=${CALENDAR_QUERY(office,${EPOCH},${EPOCH})})
 same => n,Set(soundfile=${CALENDAR_QUERY_RESULT(${id},description)})
 same => n,Playback($[${ISNULL(soundfile)} ? generic-closed :: ${soundfile}])
 same => n,Hangup
```

Meeting reminders

One useful application of Asterisk Calendaring is the ability to execute dialplan logic based on an event notification. Most calendaring technologies allow a user to set an alarm for an event. If these alarms are set on a calendar that Asterisk is monitoring and the calendar is set up for event notification via calendar.conf, then Asterisk will execute notify the specified channel at the time of the alarm. If an overrode notification time is set with the autoreminder setting, then the notification would happen at that time instead.

The following example demonstrates the set up for a simple event notification that plays back a generic message followed by the time of the upcoming meeting. calendar.conf.

```
[calendar_joe]
type = ical
url = https://example.com/home/jdoe/Calendar
user = jdoe
secret = mysecret
refresh = 15
timeframe = 600
autoreminder = 10
channel = SIP/joe
context = calendar_event_notify
extension = s
waittime = 30
```

extensions.conf :

```
[calendar_event_notify]
exten => s,1,Answer
 same => n,Playback(you-have-a-meeting-at)
 same => n,SayUnixTime(${CALENDAR_EVENT(start)})
 same => n,Hangup
```

Writing an event

Both CalDAV and Exchange calendar servers support creating new events. The following example demonstrates writing a log of a call to a calendar.

```
[incoming]
exten => 6000,1,Set(start=${EPOCH})
exten => 6000,n,Dial(SIP/joe)
exten => h,1,Set(end=${EPOCH})
exten => h,n,Set(CALENDAR_WRITE(calendar_joe,summary,start,end)=Call from
${CALLERID(all)},${start},${end})
```

Asterisk Call Files

Overview

Asterisk has the ability to initiate a call from outside of the normal methods such as the dialplan, manager interface, or spooling interface.

Using the call file method, you must give Asterisk the following information:

- How to perform the call, similar to the Dial() application
- What to do when the call is answered

With call files you submit this information simply by creating a file with the required syntax and placing it in the `outgoing` spooling directory, located by default in `/var/spool/asterisk/outgoing/` (this is configurable in `asterisk.conf`).

The `pbx_spool.so` module watches the spooling directly, either using an event notification system supplied by the operating system such as `inotify` or `kqueue`, or by polling the directory each second when one of those notification systems is unavailable. When a new file appears, Asterisk initiates a new call based on the file's contents.

Creating Files in the Spool Directory
Do **not** write or create the call file directly in the `outgoing` directory, but always create the file in another directory of the same filesystem and then move the file to the `outgoing` directory, or Asterisk may read a partial file.

NFS Considerations
By default, Asterisk will prefer to use `inotify` or `kqueue` where available. When the spooling directory is on a remote server and is mounted via NFS, the `inotify` method will fail to work. You can force Asterisk to use the older polling method by passing the `--without-inotify` flag to `configure` during compilation (e.g. `./configure --without-inotify`).

Call File Syntax

The call file consists of <Key>: <value> pairs; one per line.

Comments are indicated by a '#' character that begins a line, or follows a space or tab character. To be consistent with the configuration files in Asterisk, comments can also be indicated by a semicolon. However, the multiline comments (;----;) used in Asterisk configuration files are not supported. Semicolons can be escaped by a backslash.

The following keys-value pairs are used to specify how setup a call:

- `Channel: <channel>` - The channel to use for the new call, in the form **technology/resource** as in the Dial application. This value is required.
- `Callerid: <callerid>` - The caller id to use.
- `WaitTime: <number>` - How many seconds to wait for an answer before the call fails (ring cycle). Defaults to 45 seconds.
- `MaxRetries: <number>` - Number of retries before failing, not including the initial attempt. Default = 0 e.g. don't retry if fails.
- `RetryTime: <number>` - How many seconds to wait before retry. The default is 300 (5 minutes).
- `Account: <account>` - The account code for the call. This value will be assigned to CDR(accountcode)

When the call answers there are two choices:

1. Execute a single application, or
2. Execute the dialplan at the specified context/extension/priority.

To execute an application:

- `Application: <appname>` - The application to execute
- `Data: <args>` - The application arguments

To start executing applications in the dialplan:

- `Context: <context>` - The context in the dialplan
- `Extension: <exten>` - The extension in the specified context
- `Priority: <priority>` - The priority of the specified extension; (numeric or label)
- `Setvar: <var=value>` - You may also assign values to variables that will be available to the channel, as if you had performed a Set(var=value) in the dialplan. More than one Setvar: may be specified.

The processing of the call file ends when the call is answered and terminated; when the call was not answered in the initial attempt and subsequent retries; or if the call file can't be successfully read and parsed.

To specify what to do with the call file at the end of processing:

- `Archive: <yes|no>` - If "no" the call file is deleted. If set to "yes" the call file is moved to the "outgoing_done" subdirectory of the Asterisk spool directory. The default is to delete the call file.

If the call file is archived, Asterisk will append to the call file:

- `Status: <exitstatus>` - Can be "Expired", "Completed" or "Failed"

Other lines generated by Asterisk:

Asterisk keep track of how many retries the call has already attempted, appending to the call file the following key-pairs in the form:

```
StartRetry: <pid> <retrycount> (<time>)
EndRetry: <pid> <retrycount> (<time>)
```

With the main process ID (pid) of the Asterisk process, the retry number, and the attempts start and end times in time_t format.

Directory locations

- `<astspooldir>/outgoing` - The outgoing dir, where call files are put for processing
- `<astspooldir>/outgoing_done` - The archive dir
- `<astspooldir>` - Is specified in asterisk.conf, usually /var/spool/asterisk

How to schedule a call

Call files that have the time of the last modification in the future are ignored by Asterisk. This makes it possible to modify the time of a call file to the wanted time, move to the outgoing directory, and Asterisk will attempt to create the call at that time.

Asterisk Gateway Interface (AGI)

This page is under construction!

On This Page

Overview

AGI is analogous to CGI in Apache. AGI provides an interface between the Asterisk dialplan and an external program that wants to manipulate a channel in the dialplan. In general, the interface is synchronous - actions taken on a channel from an AGI block and do not return until the action is completed.

AGI Libraries and Frameworks

Name	Language	Website	Protocols
Adhearsion	Ruby	http://www.adhearsion.com/	AMI/FastAGI
Asterisk-Java	Java	https://asterisk-java.org/	AMI/FastAGI
PAGI	PHP	https://github.com/marcelog/PAGI	AGI
PHPAGI	PHP	http://phpagi.sourceforge.net/	AGI
Panoramisk	Python+AsyncIO	https://github.com/gawel/panoramisk	AMI/FastAGI
Pyst2	Python	https://github.com/rdegges/pyst2	AMI/AGI
StarPy	Python+Twisted	https://github.com/asterisk/starpy	AMI/FastAGI
Nanoagi	C++	http://sourceforge.net/projects/nanoagi/	AGI
AsterNET	.NET (C#/VB.net)	https://github.com/skrusty/AsterNET	AMI/FastAGI
Ding-dong	node.js	https://www.npmjs.com/package/ding-dong	AGI

Asterisk Manager Interface (AMI)

What is the Asterisk Manager Interface, or AMI? Read on...

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.

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)

AMI 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

AMI 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 <commandname>" CLI command in Asterisk.

AMI 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.

Ensuring all modules are loaded with AMI

It is possible to connect to the manager interface before all Asterisk modules are loaded. To ensure that an application does not send AMI actions that might require a module that has not yet loaded, the application can listen for the FullyBooted manager event. It will be sent upon connection if all modules have been loaded, or as soon as loading is complete. The event:

```
Event: FullyBooted
Privilege: system,all
Status: Fully Booted
```

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)
- Direction: <type> – Audio to mute (read | write | both)
- 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" packet)
- Location: – Interface (whatever that is -maybe tech/name in app_queue)
- Loginchan: – Login channel for agent
- Logintime: <number> – Login time for agent

- Mailbox: – VM Mailbox (id@vmcontext) (mailboxstatus, mailboxcount)
- 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)
- MOHSuggest: – Suggested music on hold class for peer (mohsuggest)
- 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, command, agent, user)
- 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
- State: <1 | 0> – Mute flag
- 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 | in Originate)
- 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 Coding Guidelines if you develop new manager commands or events.

Asynchronous Javascript Asterisk Manager (AJAM)

AJAM is a new technology which allows web browsers or other HTTP enabled applications and web pages to directly access the Asterisk Manager Interface (AMI) via HTTP. Setting up your server to process AJAM involves a few steps:

Setting up the Asterisk HTTP server

Next you'll need to enable Asterisk's Builtin mini-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 blank, that is no prefix and the rest of these instructions assume that value.

Allow Manager Access via HTTP

Configuring manager.conf

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

Usage of AMI over HTTP

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:

> ⓘ Be sure the syntax for the URLs below is followed precisely

- http://localhost:8088/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/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/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/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.

> ⚠ For the demo, there is no need for **any external web server.**

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

Asterisk Manager Interface (AMI) Changes

AMI version numbers are formatted as MAJOR.BREAKING.NON-BREAKING:

- MAJOR – changes when a new major version of Asterisk is released
- BREAKING – changes when an incompatible API modification is made
- NON-BREAKING – changes when backwards compatible updates are made (new additions or bug fixes)

Any time a new major version of Asterisk is released the AMI version number is modified to reflect that. This is done by taking the major AMI number of the last major release of Asterisk, increasing it by one, and then resetting the other numbers to zero. From there the breaking and non-breaking versions are increased by one when a change (incompatible and compatible respectively) has been made to AMI.

This has the effect of making the AMI version number relative in relation to an associated Asterisk release, and contextual applicable only to that major release of Asterisk.

AMI 1.1 Changes

Changes to manager version 1.1:

SYNTAX CLEANUPS

```
- Response: headers are now either
 "Success" - Action OK, this message contains response
 "Error"  - Action failed, reason in Message: header
 "Follows" - Action OK, response follows in following Events.

- Manager version changed to 1.1
```

CHANGED EVENTS AND ACTIONS

```
- The Hold/Unhold events
 - Both are now "Hold" events
  For hold, there's a "Status: On" header, for unhold, status is off
 - Modules chan_sip/chan_iax2

- The Ping Action
 - Now use Response: success
 - New header "Ping: pong" :-)

- The Events action
 - Now use Response: Success
 - The new status is reported as "Events: On" or "Events: Off"

- The JabberSend action
 - The Response: header is now the first header in the response
 - now sends "Response: Error" instead of "Failure"

- Newstate and Newchannel events
 - these have changed headers
 "State"  -> ChannelStateDesc Text based channel state
   -> ChannelState  Numeric channel state
 - The events does not send "<unknown>" for unknown caller IDs just an empty field

- Newchannel event
 - Now includes "AccountCode"

- Newstate event
 - Now has "CalleridNum" for numeric caller id, like Newchannel
 - The event does not send "<unknown>" for unknown caller IDs just an empty field

- Newexten and VarSet events
 - Now are part of the new Dialplan privilege class, instead of the Call class

- Dial event
 - Event Dial has new headers, to comply with other events
 - Source -> Channel  Channel name (caller)
 - SrcUniqueID -> UniqueID  Uniqueid
 (new)  -> Dialstring  Dialstring in app data

- Link and Unlink events
 - The "Link" and "Unlink" bridge events in channel.c are now renamed to "Bridge"
 - The link state is in the bridgestate: header as "Link" or "Unlink"
 - For channel.c bridges, "Bridgetype: core" is added. This opens up for
   bridge events in rtp.c
 - The RTP channel also reports Bridge: events with bridgetypes
  - rtp-native RTP native bridge
  - rtp-direct RTP peer-2-peer bridge (NAT support only)
  - rtp-remote Remote (re-invite) bridge. (Not reported yet)

- The "Rename" manager event has a renamed header, to use the same
 terminology for the current channel as other events
 - Oldname -> Channel

- The "NewCallerID" manager event has a renamed header
 - CallerID -> CallerIDnum
 - The event does not send "<unknown>" for unknown caller IDs just an empty field

- Reload event
 - The "Reload" event sent at manager reload now has a new header and is now implemented
   in more modules than manager to alert a reload. For channels, there's a CHANNELRELOAD
   event to use.
 (new)  -> Module: manager | CDR | DNSmgr | RTP | ENUM
 (new)  -> Status: enabled | disabled
 - To support reload events from other modules too
  - cdr module added

- Status action replies (Event: Status)
 Header changes
 - link  -> BridgedChannel
 - Account -> AccountCode
 - (new)  -> BridgedUniqueid
- StatusComplete Event
 New header
 - (new)  -> Items  Number of channels reported


- The ExtensionStatus manager command now has a "StatusDesc" field with text description of the state

- The Registry and Peerstatus events in chan_sip and chan_iax now use "ChannelType" instead of "ChannelDriver"

- The Response to Action: IAXpeers now have a Response: Success header

- The MeetmeJoin now has caller ID name and Caller ID number fields (like MeetMeLeave)

- Action DAHDIShowChannels
 Header changes
 - Channel: -> DAHDIChannel
 For active channels, the Channel: and Uniqueid: headers are added
 You can now add a "DAHDIChannel: " argument to DAHDIshowchannels actions
 to only get information about one channel.

- Event DAHDIShowChannelsComplete
 New header
 - (new)  -> Items:  Reports number of channels reported

- Action VoicemailUsersList
 Added new headers for SayEnvelope, SayCID, AttachMessage, CanReview
        and CallOperator voicemail configuration settings.

- Action Originate
 Now requires the new Originate privilege.
 If you call out to a subshell in Originate with the Application parameter,
  you now also need the System privilege.

- Event QueueEntry now also returns the Uniqueid field like other events from app_queue.

- Action IAXpeerlist
 Now includes if the IAX link is a trunk or not

- Action IAXpeers
 Now includes if the IAX link is a trunk or not

- Action Ping
 Response now includes a timestamp

- Action SIPshowpeer
 Response now includes the configured parkinglot
```

```
- Action SKINNYshowline
 Response now includes the configured parkinglot
```

NEW ACTIONS

```
- Action: DataGet
 Modules: data.c
 Purpose:
  To be able to retrieve the asterisk data tree.
 Variables:
   ActionID: <id>          Action ID for this transaction. Will be returned.
   Path: <data path>       The path to the callback node to retrieve.
   Filter: <filter>        Which nodes to retrieve.
   Search: <search>        Search condition.

- Action: IAXregistry
 Modules: chan_iax2
 Purpose:
  To list all IAX2 peers in the IAX registry with their registration status.
 Variables:
   ActionID: <id>  Action ID for this transaction. Will be returned.

- Action: ModuleLoad
 Modules: loader.c
 Purpose:
  To be able to unload, reload and unload modules from AMI.
 Variables:
   ActionID: <id>          Action ID for this transaction. Will be returned.
     Module: <name>          Asterisk module name (including .so extension)
      or subsystem identifier:
    cdr, enum, dnsmgr, extconfig, manager, rtp, http
          LoadType: load | unload | reload
                          The operation to be done on module
 If no module is specified for a reload loadtype, all modules are reloaded

- Action: ModuleCheck
 Modules: loader.c
 Purpose:
  To check version of a module - if it's loaded
 Variables:
   ActionID: <id>          Action ID for this transaction. Will be returned.
     Module: <name>          Asterisk module name (not including extension)
 Returns:
  If module is loaded, returns version number of the module

  Note: This will have to change. I don't like sending Response: failure
  on both command not found (trying this command in earlier versions of
  Asterisk) and module not found.
  Also, check if other manager actions behave that way.

- Action: QueueSummary
 Modules: app_queue
 Purpose:
  To request that the manager send a QueueSummary event (see the NEW EVENTS
     section for more details).
 Variables:
   ActionID: <id>  Action ID for this transaction. Will be returned.
   Queue: <name>   Queue for which the summary is desired

- Action: QueuePenalty
 Modules: app_queue
 Purpose:
  To change the penalty of a queue member from AMI
 Variables:
   Interface: <tech/name> The interface of the member whose penalty you wish to change
   Penalty:  <number>  The new penalty for the member. Must be nonnegative.
   Queue:  <name>  If specified, only set the penalty for the member for this queue;
     Otherwise, set the penalty for the member in all queues to which
     he belongs.

- Action: QueueRule
 Modules: app_queue
 Purpose:
  To list queue rules defined in queuerules.conf
 Variables:
          ActionID: <id>                Action ID for this transaction. Will be returned.
   Rule: <name>   The name of the rule whose contents you wish to list. If this variable
       is not present, all rules in queuerules.conf will be listed.

- Action: Atxfer
 Modules: none
 Purpose:
  Initiate an attended transfer
 Variables:
  Channel: The transferer channel's name
  Exten: The extension to transfer to
  Priority: The priority to transfer to
  Context: The context to transfer to

- Action: SipShowRegistry
 Modules: chan_sip
 Purpose:
  To request that the manager send a list of RegistryEntry events.
 Variables:
   ActionId: <id>  Action ID for this transaction. Will be returned.

- Action: QueueReload
 Modules: app_queue
 Purpose:
  To reload queue rules, a queue's members, a queue's parameters, or all of the aforementioned
 Variable:
                ActionID: <id>
  Queue: <name> The name of the queue to take action on.
                              If no queue name is specified, then all queues are affected
  Rules: <yes or no> Whether to reload queuerules.conf
  Members: <yes or no> Whether to reload the queue's members
  Parameters: <yes or no> Whether to reload the other queue options

- Action: QueueReset
 Modules: app_queue
 Purpose:
  Reset the statistics for a queue
 Variables:
                ActionID: <id>
  Queue: <name> The name of the queue on which to reset statistics

- Action: SKINNYdevices
 Modules: chan_skinny
 Purpose:
  To list all SKINNY devices configured.
 Variables:
  ActionId: <id> Action ID for this transaction. Will be returned.

- Action: SKINNYlines
 Modules: chan_skinny
 Purpose:
  To list all SKINNY lines configured.
 Variables:
  ActionId: <id> Action ID for this transaction. Will be returned.

- Action SKINNYshowdevice
 Modules: chan_skinny
 Purpose:
  To list the information about a specific SKINNY device.
 Variables:
  Device: <device> Device to show information about.

- Action SKINNYshowline
 Modules: chan_skinny
 Purpose:
  To list the information about a specific SKINNY line.
 Variables:
  Line: <line> Line to show information about.

- Action: CoreSettings
 Modules: manager.c
 Purpose: To report core settings, like AMI and Asterisk version,
  maxcalls and maxload settings.
  * Integrated in SVN trunk as of May 4th, 2007
 Example:
  Response: Success
  ActionID: 1681692777
  AMIversion: 1.1
  AsteriskVersion: SVN-oej-moremanager-r61756M
  SystemName: EDVINA-node-a
  CoreMaxCalls: 120
  CoreMaxLoadAvg: 0.000000
  CoreRunUser: edvina
  CoreRunGroup: edvina

- Action: CoreStatus
 Modules: manager.c
 Purpose: To report current PBX core status flags, like
  number of concurrent calls, startup and reload time.
  * Integrated in SVN trunk as of May 4th, 2007
 Example:
  Response: Success
  ActionID: 1649760492
  CoreStartupTime: 22:35:17
  CoreReloadTime: 22:35:17
  CoreCurrentCalls: 20

- Action: MixMonitorMute
 Modules: app_mixmonitor.c
 Purpose:
  Mute / unMute a Mixmonitor recording.
```

```
Variables:
 ActionId: <id> Action ID for this transaction. Will be returned.
 Channel: the channel MixMonitor is running on
 Direction: Which part of the recording to mute:  read, write or both (from
  channel, to channel or both channels).
 State: Turn mute on or off : 1 to turn on, 0 to turn off.
```

NEW EVENTS

```
- Event: FullyBooted
 Modules: loader.c
 Purpose:
  It is handy to have a single event notification for when all Asterisk
  modules have been loaded--especially for situations like running
  automated tests. This event will fire 1) immediately upon all modules
  loading or 2) upon connection to the AMI interface if the modules have
  already finished loading before the connection was made. This ensures
  that a user will never miss getting a FullyBooted event. In vary rare
  circumstances, it might be possible to get two copies of the message
  if the AMI connection is made right as the modules finish loading.
 Example:
  Event: FullyBooted
  Privilege: system,all
  Status: Fully Booted

- Event: Transfer
 Modules: res_features, chan_sip
 Purpose:
  Inform about call transfer, linking transferer with transfer target
  You should be able to trace the call flow with this missing piece
  of information. If it works out well, the "Transfer" event should
  be followed by a "Bridge" event
  The transfermethod: header informs if this is a pbx core transfer
  or something done on channel driver level. For SIP, check the example:
 Example:

  Event: Transfer
  Privilege: call,all
  TransferMethod: SIP
  TransferType: Blind
  Channel: SIP/device1-01849800
  SIP-Callid: 091386f505842c87016c4d93195ec67d@127.0.0.1
  TargetChannel: SIP/device2-01841200
  TransferExten: 100
  TransferContext: default

- Event: ChannelUpdate
 Modules: chan_sip.c, chan_iax2.c
 Purpose:
  Updates channel information with ID of PVT in channel driver, to
  be able to link events on channel driver level.
  * Integrated in SVN trunk as of May 4th, 2007

 Example:

  Event: ChannelUpdate
  Privilege: system,all
  Uniqueid: 1177271625.27
  Channel: SIP/olle-01843c00
  Channeltype: SIP
  SIPcallid: NTQzYWFiOWM4NmE0MWRkZjExMzU2YzQ3OWQwNzg3ZmI.
  SIPfullcontact: sip:olle@127.0.0.1:49054

- Event: NewAccountCode
 Modules: cdr.c
 Purpose: To report a change in account code for a live channel
 Example:
  Event: NewAccountCode
  Privilege: call,all
  Channel: SIP/olle-01844600
  Uniqueid: 1177530895.2
  AccountCode: Stinas account 1234848484
  OldAccountCode: OllesAccount 12345

- Event: ModuleLoadReport
 Modules: loader.c
 Purpose: To report that module loading is complete. Some aggressive
  clients connect very quickly to AMI and needs to know when
  all manager events embedded in modules are loaded
  Also, if this does not happen, something is seriously wrong.
  This could happen to chan_sip and other modules using DNS.
 Example:
  Event: ModuleLoad
  ModuleLoadStatus: Done
  ModuleSelection: All
  ModuleCount: 24

- Event: QueueSummary
 Modules: app_queue
 Purpose: To report a summary of queue information. This event is generated by
  issuing a QueueSummary AMI action.
 Example:
  Event: QueueSummary
  Queue: Sales
  LoggedIn: 12
  Available: 5
  Callers: 10
  HoldTime: 47
 If an actionID was specified for the QueueSummary action, it will be appended as the
 last line of the QueueSummary event.

- Event: AgentRingNoAnswer
 Modules: app_queue
 Purpose: Reports when a queue member was rung but there was no answer.
 Example:
  Event: AgentRingNoAnswer
  Queue: Support
  Uniqueid: 1177530895.2
  Channel: SIP/1000-53aee458
  Member: SIP/1000
  MemberName: Thaddeus McClintock
  Ringtime: 10

- Event: RegistryEntry
 Modules: chan_sip
 Purpose: Reports the state of the SIP registrations. This event is generated by
                issuing a QueueSummary AMI action.
  The RegistrationTime header is expressed as epoch.
 Example:
  Event: RegistryEntry
  Host: sip.myvoipprovider.com
  Port: 5060
  Username: guestuser
  Refresh: 105
  State: Registered
  RegistrationTime: 1219161830
 If an actionID was specified for the SipShowRegistry action, it will be appended as the
 last line of the RegistrationsComplete event.

- Event: ChanSpyStart
 Modules: app_chanspy
 Purpose: Reports when an active channel starts to be monitored by someone.
 Example:
  Event: ChanSpyStart
  SpyerChannel: SIP/4321-13bba124
  SpyeeChannel: SIP/1234-56ecc098

- Event: ChanSpyStop
 Modules: app_chanspy
 Purpose: Reports when an active channel stops to be monitored by someone.
 Example:
```

```
Event: ChanSpyStop
SpyeeChannel: SIP/1234-56ecc098
```

TODO

...

AMI Libraries and Frameworks

This page is under construction!

Name	Language	Website	Protocols
Asterisk-Java	Java	https://blogs.reucon.com/asterisk-java/	AMI/FastAGI
StarPy	Python+Twisted	https://github.com/asterisk/starpy	AMI/FastAGI
Panoramisk	Python+AsyncIO	https://github.com/gawel/panoramisk	AMI/FastAGI
PAMI	PHP	https://github.com/marcelog/PAMI	AMI
Pyst2	Python	https://github.com/rdegges/pyst2	AMI/AGI
Adhearsion	Ruby	http://www.adhearsion.com/	AMI/FastAGI
node-asterisk	Node.js	https://github.com/danjenkins/node-asterisk-ami	AMI
AMI-IO	Node.js	https://github.com/NumminorihSF/ami-io	AMI
NodeJS-AsteriskManager	Node.js	https://github.com/pipobscure/NodeJS-AsteriskManager	AMI
AsterNET	.NET	https://github.com/AsterNET/AsterNET	AMI/FastAGI
AmiClient	.NET	https://github.com/alexforster/AmiClient	AMI

Asterisk REST Interface (ARI)

The Evolution of Asterisk APIs

When Asterisk was first created back in 1999, its design was focussed on being a stand-alone Private Branch eXchange (PBX) that you could configure via static `.conf` files. Control of the calls that passed through it was done through a special `.conf` file, `extensions.conf`, known as the "dialplan". The dialplan script told Asterisk which applications to execute on the call, and made logical decisions based on what the users did through their phones. This model worked well for a long time - it was certainly more flexible than what existed at the time, and the plethora of dialplan applications provided a large suite of functionality.

These dialplan applications, however, were - and still are - written in C. Because the applications act directly on the raw primitives in Asterisk, they are incredibly powerful. They have access to the channels, media, bridges, endpoints, and all other objects that Asterisk uses to make phones communicate. However, while powerful, there are times when a business use case is not met by the existing suite of applications. In the past, if the functionality you needed wasn't met by the dialplan application, you really only had one solution: write a patch in C - possibly adding a parameter to the application to tweak the behaviour - and submit it to the project. If you could not write the feature in C, you were, unfortunately, stuck.

Enter AMI and AGI

Not long into the project, two application programming interfaces (APIs) were added to Asterisk: the Asterisk Gateway Interface (AGI) and the Asterisk Manager Interface (AMI). These interfaces have distinct purposes:

1. AGI is analogous to CGI in Apache. AGI provides an interface between the Asterisk dialplan and an external program that wants to manipulate a channel in the dialplan. In general, the interface is synchronous - actions taken on a channel from an AGI block and do not return until the action is completed.
2. AMI provides a mechanism to control where channels execute in the dialplan. Unlike AGI, AMI is an asynchronous, event driven interface. For the most part, AMI does not provide mechanisms to control channel execution - rather, it provides information about the state of the channels and controls about where the channels are executing.

Both of these interfaces are powerful and opened up a wide range of integration possibilities. Using AGI, remote dialplan execution could be enabled, which allowed developers to integrate Asterisk with PHP, Python, Java, and other applications. Using AMI, the state of Asterisk could be displayed, calls initiated, and the location of channels controlled. Using both APIs together, complex applications using Asterisk as the engine *could* be developed.

On This Page

ARI In More Detail

- Getting Started with ARI
- Asterisk Configuration for ARI
- Introduction to ARI and Channels
- Introduction to ARI and Bridges
- Introduction to ARI and Media Manipulation
- The Asterisk Resource
- ARI Versioning
- ARI Libraries

However, there are some drawbacks to using AMI and AGI to create custom communication applications:

1. AGI is synchronous and blocks the thread servicing the AGI when an Asterisk action is taken on the channel. When creating a communications application, you will often want to respond to changes in the channel (DTMF, channel state, etc.); this is difficult to do with AGI by itself. Coordinating with AMI events can be challenging.
2. The dialplan can be limiting. Even with AMI and AGI, your fundamental operations are limited to what can be executed on a channel. While powerful, there are other primitives in Asterisk that are not available through those APIs: bridges, endpoints, device state, message waiting indications, and the actual media on the channels themselves. Controlling those through AMI and AGI can be difficult, and can often involve complex dialplan manipulation to achieve.

3. Finally, both AMI and AGI were created early in the Asterisk project, and are products of their time. While both are powerful interfaces, technologies that are used today were not in heavy use at the time. Concepts such as SOAP, XML/JSON-RPC, and REST were not in heavy use. As such, newer APIs can be more intuitive and easier to adopt, leading to faster development for users of Asterisk.

And so, before Asterisk 12, if you wanted your own custom communication application, you could:

- Write an Asterisk module in C, **or**
- Write a custom application using both AGI and AMI, performing some herculean effort to marry the two APIs together (as well as some dialplan trickery)

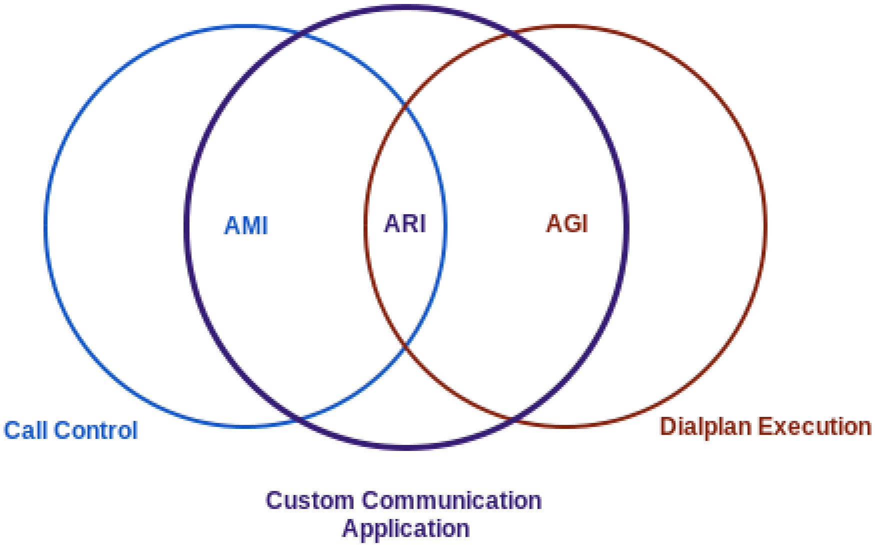
ARI: An Interface for Communications Applications

The Asterisk RESTful Interface (ARI) was created to address these concerns. While AMI is good at call control and AGI is good at allowing a remote process to execute dialplan applications, neither of these APIs was designed to let a developer build their own custom communications application. ARI is an asynchronous API that allows developers to build communications applications by exposing the raw primitive objects in Asterisk - channels, bridges, endpoints, media, etc. - through an intuitive REST interface. The state of the objects being controlled by the user are conveyed via JSON events over a WebSocket.

These resources were traditionally the purview of Asterisk's C modules. By handing control of these resources over to all developers - regardless of their language choice - Asterisk becomes an engine of communication, with the business logic of how things should communicate deferred to the application using Asterisk.

ARI is not about telling a channel to execute the VoiceMail dialplan application or redirecting a channel in the dialplan to VoiceMail.

It is about letting you build your own VoiceMail application.



ARI Fundamentals

ARI consists of three different pieces that are - for all intents and purposes - interrelated and used together. They are:

1. A RESTful interface that a client uses to control resources in Asterisk.
2. A WebSocket that conveys events in JSON about the resources in Asterisk to the client.
3. The Stasis dialplan application that hands over control of a channel from Asterisk to the client.

All three pieces work together, allowing a developer to manipulate and control the fundamental resources in Asterisk and build their own communications application.

Developer Documentation
You can find some historical documents on the wiki about the development and architecture of ARI.

What is REST?

Representational State Transfer (REST) is a software architectural style. It has several characteristics:

- Communication is performed using a client-server model.
- The communication is stateless. Servers do not store client state between requests.

- Connections are layered, allowing for intermediaries to assist in routing and load balancing.
- A uniform interface. Resources are identified in the requests, messages are self-descriptive, etc.

ARI *does not* strictly conform to a REST API. Asterisk, as a stand-alone application, has state that may change outside of a client request through ARI. For example, a SIP phone may be hung up, and Asterisk will hang up the channel - even though a client through ARI did not tell Asterisk to hang up the SIP phone. Asterisk lives in an asynchronous, state-ful world: hence, ARI is *RESTful*. It attempts to follow the tenants of REST as best as it can, without getting bogged down in philosophical constraints.

What is a WebSocket?

WebSockets are a relatively new protocol standard (RFC 6455) that enable two-way communication between a client and server. The protocol's primary purpose is to provide a mechanism for browser-based applications that need two-way communication with servers, without relying on HTTP long polling or other, non-standard, mechanisms.

In the case of ARI, a WebSocket connection is used to pass asynchronous events from Asterisk to the client. These events are related to the RESTful interface, but are technically independent of it. They allow Asterisk to inform the client of changes in resource state that may occur because of and in conjunction with the changes made by the client through ARI.

What is Stasis?

Stasis is a dialplan application in Asterisk. It is the mechanism that Asterisk uses to hand control of a channel over from the dialplan - which is the traditional way in which channels are controlled - to ARI and the client. Generally, ARI applications manipulate channels in the Stasis dialplan application, as well as other resources in Asterisk. Channels not in a Stasis dialplan application generally cannot be manipulated by ARI - the purpose of ARI, after all, is to build your own dialplan application, not manipulate an existing one.

Diving Deeper

This space has a number of pages that explore different resources available to you in ARI and examples of what you can build with them. Generally, the examples assume the following:

- That you have some phone registered to Asterisk, typically using `chan_pjsip` or `chan_sip`
- That you have some basic knowledge of configuring Asterisk
- A basic knowledge of Python, JavaScript, or some other higher level programming language (or a willingness to learn!)

Most of the examples will not directly construct the HTTP REST calls, as a number of very useful libraries have been written to encapsulate those mechanics. These libraries are listed below.

Where to get the examples

All of the examples on the pages below this one are available on github. Check them out!

ARI Libraries

See the ARI Libraries page for a list of Asterisk Rest Interface libraries and frameworks.

Recommended Practices

Don't access ARI directly from a web page

It's very convenient to use ARI directly from a web page for development, such as using Swagger-UI, or even abusing the WebSocket echo demo to get at the ARI WebSocket.

But, *please*, do not do this in your production applications. This would be akin to accessing your database directly from a web page. You need to hide Asterisk behind your own application server, where you can handle security, logging, multi-tenancy and other concerns that really don't belong in a communications engine.

Use an abstraction layer

One of the beautiful things about ARI is that it's so easy to just bang out a request. But what's good for development isn't necessarily what's good for production.

Please don't spread lots of direct HTTP calls throughout your application. There are cross-cutting concerns with accessing the API that you'll want to deal with in a central location. Today, the only concern is authentication. But as the API evolves, other concerns (such as versioning) will also be important.

Note that the abstraction layer doesn't (and shouldn't) be complicated. Your client side API can even be something as simple wrapper around GET, POST and DELETE that addresses the cross-cutting concerns. The Asterisk TestSuite has a very simple abstraction library that can be used like this:

```
ari = ARI('localhost', ('username', 'password'))

# Hang up all channels
channels = ari.get('channels')
for channel in channels:
    ari.delete('channels', channel['id'])
```

In other words: **use one of the aforementioned libraries or write your own!**

Getting Started with ARI

Overview

Asterisk 12 introduces the Asterisk REST Interface, a set of RESTful APIs for building Asterisk based applications. This article will walk you though getting ARI up and running.

There are three main components to building an ARI application.

The first, obviously, is **the RESTful API** itself. The API is documented using Swagger, a lightweight specification for documenting RESTful APIs. The Swagger API docs are used to generate validations and boilerplate in Asterisk itself, along with static wiki documentation, and interactive documentation using Swagger-UI.

Then, Asterisk needs to send asynchronous events to the application (new channel, channel left a bridge, channel hung up, etc). This is done using a **Web Socket on /ari/events**. Events are sent as JSON messages, and are documented on the REST Data Models page. (See the list of subtypes for the `Message` data model.)

Finally, connecting the dialplan to your application is the `Stasis()` dialplan application. From within the dialplan, you can send a channel to `Stasis()`, specifying the name of the external application, along with optional arguments to pass along to the application.

On This Page

Example: ARI Hello World!

In this example, we will:

- Configure Asterisk to enable ARI
- Send a channel into Stasis
- And playback "Hello World" to the channel

This example will **not** cover:

1. Installing Asterisk. We'll assume you have Asterisk 12 or later installed and running.
2. Configuring a SIP device in Asterisk. For the purposes of this example, we are going to assume you have a SIP softphone or hardphone registered to Asterisk, using either `chan_sip` or `chan_pjsip`.

Getting wscat

ARI needs a WebSocket connection to receive events. For the sake of this example, we're going to use wscat, an incredibly handy command line utility similar to netcat but based on a node.js websocket library. If you don't have wscat:

1. If you don't have it already, **install** npm

   ```
   $ apt-get install npm
   ```

2. **Install** the `ws` node package:

   ```
   $ npm install -g wscat
   ```

Some distributions repos (e.g. Ubuntu) may have older versions of nodejs and npm that will throw a wrench in your install of the ws package. You'll have to install a newer version from another repo or via source.

Installing Nodejs via packages

Installing npm in a variety of ways

Getting curl

In order to control a channel in the Stasis dialplan application through ARI, we also need an HTTP client. For the sake of this example, we'll use curl:

```
$ apt-get install curl
```

Configuring Asterisk

1. Enable the Asterisk HTTP service in `http.conf`:

   http.conf

   ```
   [general]
   enabled = yes
   bindaddr = 0.0.0.0
   ```

2. Configure an ARI user in `ari.conf`:

   ari.conf

   ```
   [general]
   enabled = yes
   pretty = yes

   [asterisk]
   type = user
   read_only = no
   password = asterisk
   ```

   This is just a demo
   Please use a more secure account user and password for production applications. Outside of examples and demos, asterisk/asterisk is a terrible, horrible, no-good choice...

3. Create a dialplan extension for your Stasis application. Here, we're choosing extension `1000` in context `default` - if your SIP phone is configured for a different context, adjust accordingly.

   extensions.conf

   ```
   [default]

   exten => 1000,1,NoOp()
    same =>      n,Answer()
    same =>      n,Stasis(hello-world)
    same =>      n,Hangup()
   ```

Hello World!

1. Connect to Asterisk using `wscat`:

   ```
   $ wscat -c
   "ws://localhost:8088/ari/events?api_key=asterisk:asterisk&app=hello-world"
   connected (press CTRL+C to quit)
   >
   ```

   In Asterisk, we should see a new WebSocket connection and a message telling us that our Stasis application has been created:

   ```
   == WebSocket connection from '127.0.0.1:37872' for protocol '' accepted using
   version '13'
    Creating Stasis app 'hello-world'
   ```

2. From your SIP device, dial extension 1000:

```
    -- Executing [1000@default:1] NoOp("PJSIP/1000-00000001", "") in new stack
    -- Executing [1000@default:2] Answer("PJSIP/1000-00000001", "") in new stack
    -- PJSIP/1000-00000001 answered
    -- Executing [1000@default:3] Stasis("PJSIP/1000-00000001", "hello-world") in
new stack
```

In wscat, we should see the `StasisStart` event, indicating that a channel has entered into our Stasis application:

```
< {
   "application":"hello-world",
   "type":"StasisStart",
   "timestamp":"2014-05-20T13:15:27.131-0500",
   "args":[],
   "channel":{
      "id":"1400609726.3",
      "state":"Up",
      "name":"PJSIP/1000-00000001",
      "caller":{
         "name":"",
         "number":""},
      "connected":{
         "name":"",
         "number":""},
      "accountcode":"",
      "dialplan":{
         "context":"default",
         "exten":"1000",
         "priority":3},
      "creationtime":"2014-05-20T13:15:26.628-0500"}
   }
>
```

3. Using `curl`, tell Asterisk to playback `hello-world`. Note that the identifier of the channel in the `channels` resource **must** match the channel `id` passed back in the `StasisStart` event:

```
$ curl -v -u asterisk:asterisk -X POST
"http://localhost:8088/ari/channels/1400609726.3/play?media=sound:hello-world"
```

The response to our HTTP request will tell us whether or not the request succeeded or failed (in our case, a success will queue the playback onto the channel), as well as return in JSON the Playback resource that was created for the operation:

```
* About to connect() to localhost port 8088 (#0)
*   Trying 127.0.0.1... connected
* Server auth using Basic with user 'asterisk'
> POST /ari/channels/1400609726.3/play?media=sound:hello-world HTTP/1.1
> Authorization: Basic YXN0ZXJpc2s6c2VjcmV0
> User-Agent: curl/7.22.0 (x86_64-pc-linux-gnu) libcurl/7.22.0 OpenSSL/1.0.1
zlib/1.2.3.4 libidn/1.23 librtmp/2.3
> Host: localhost:8088
> Accept: */*
>
< HTTP/1.1 201 Created
< Server: Asterisk/SVN-branch-12-r414137M
< Date: Tue, 20 May 2014 18:25:15 GMT
< Connection: close
< Cache-Control: no-cache, no-store
< Content-Length: 146
< Location: /playback/9567ea46-440f-41be-a044-6ecc8100730a
< Content-type: application/json
<
* Closing connection #0
{"id":"9567ea46-440f-41be-a044-6ecc8100730a",
 "media_uri":"sound:hello-world",
 "target_uri":"channel:1400609726.3",
 "language":"en",
 "state":"queued"}

$
```

In Asterisk, the sound file will be played back to the channel:

```
-- <PJSIP/1000-00000001> Playing 'hello-world.gsm' (language 'en')
```

And in our `wscat` WebSocket connection, we'll be informed of the start of the playback, as well as it finishing:

```
< {"application":"hello-world",
   "type":"PlaybackStarted",
   "playback":{
      "id":"9567ea46-440f-41be-a044-6ecc8100730a",
      "media_uri":"sound:hello-world",
      "target_uri":"channel:1400609726.3",
      "language":"en",
      "state":"playing"}
   }

< {"application":"hello-world",
   "type":"PlaybackFinished",
   "playback":{
      "id":"9567ea46-440f-41be-a044-6ecc8100730a",
      "media_uri":"sound:hello-world",
      "target_uri":"channel:1400609726.3",
      "language":"en",
      "state":"done"}
   }
```

4. Hang up the phone! This will cause the channel in Asterisk to be hung up, and the channel will leave the Stasis application, notifying the client via a `StasisEnd` event:

```
< {"application":"hello-world",
   "type":"StasisEnd",
   "timestamp":"2014-05-20T13:30:01.852-0500",
   "channel":{
      "id":"1400609726.3",
      "state":"Up",
      "name":"PJSIP/1000-00000001",
      "caller":{
         "name":"",
         "number":""},
      "connected":{
         "name":"",
         "number":""},
      "accountcode":"",
      "dialplan":{
         "context":"default",
         "exten":"1000",
         "priority":3},
      "creationtime":"2014-05-20T13:15:26.628-0500"}
   }
```

Using Swagger to Drive ARI

Using Swagger-UI

The REST API that makes up ARI is documented using Swagger, a lightweight specification for documenting RESTful API's. The Swagger API docs are used to generate validations and boilerplate in Asterisk itself, along with static wiki documentation, and interactive documentation using Swagger-UI.

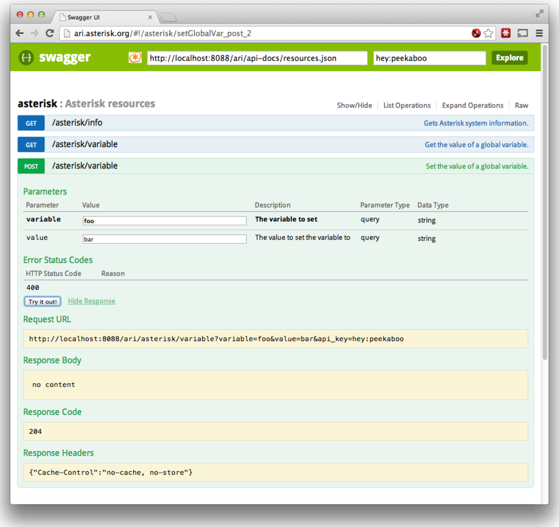
Swagger-UI is a pure HTML+JavaScript application which can download Swagger api-docs, and generate an interactive web page which allows you to view resources, their operations, and submit API requests directly from the documentation. A fork of Swagger-UI is hosted on ari.asterisk.org, which enables DELETE operations (which are disabled by default in Swagger-UI), and sets the default URL to what it would be running Asterisk on your local system.

In order to access ARI, you have to populate the `api_key` field with a `username:password` configured in `ari.conf`. You should also set `allowed_origins` in `ari.conf` to allow the site hosting Swagger-UI to access ARI.

ari.conf

```
[general]
enabled=yes
;pretty=yes     ; we don't need pretty-printing of the JSON responses in this
                ; example, but you might if you use curl a lot.
;
; In this example, we are going to use the version of Swagger-UI that is hosted
; at ari.asterisk.org. In order to get past CORS restrictions in the browser,
; That origin needs to be added to the allowed_origins list.
;
allowed_origins=http://ari.asterisk.org

[hey]
type=user
password=peekaboo
;read_only=no   ; Set to yes for read-only applications
```



asterisk : Asterisk resources

GET	/asterisk/info	Gets Asterisk system information.
GET	/asterisk/variable	Get the value of a global variable.
POST	/asterisk/variable	Set the value of a global variable.

Parameters

Parameter	Value	Description	Parameter Type	Data Type
variable	foo	**The variable to set**	query	string
value	bar	The value to set the variable to	query	string

Error Status Codes

HTTP Status Code	Reason
400	

Request URL

```
http://localhost:8088/ari/asterisk/variable?variable=foo&value=bar&api_key=hey:peekaboo
```

Response Body

```
no content
```

Response Code

```
204
```

Response Headers

```
{"Cache-Control":"no-cache, no-store"}
```

Asterisk Configuration for ARI

Overview

ARI has a number of parts to it - the HTTP server in Asterisk servicing requests, the dialplan application handing control of channels over to a connected client, and the websocket sharing state in Asterisk with the external application. This page provides the configuration files in Asterisk that can be altered to suit deployment considerations.

This page does not include all of the configuration options available to a system administrator. It does cover some of the basics that you might be interested in when setting up your Asterisk system for ARI.

Asterisk Configuration Options for ARI

HTTP Server

The HTTP server in Asterisk is configured via `http.conf`. Note that this does not describe all of the options available via `http.conf` - rather, it lists the most useful ones for ARI.

On This Page

Section	Parameter	Type	Default	Description
`general`				
	`enabled`	Boolean	False	Enable the HTTP server. **The HTTP server in Asterisk is disabled by default**. Unless it is enabled, ARI will not function!
	`bindaddr`	IP Address		The IP address to bind the HTTP server to. This can either be an explicit local address, or `0.0.0.0` to bind to all available interfaces.
	`bindport`	Port	8088	The port to bind the HTTP server to. Client making HTTP requests should specify 8088 as the port to send the request to.
	`prefix`	String		A prefix to require for all requests. If specified, requests must begin with the specified prefix.
	`tlsenable`	Boolean	False	Enable HTTPS
	`tlsbindaddr`	IP Address/Port		The IP address and port to bind the HTTPS server to. This should be an IP address and port, e.g., `0.0.0.0:8089`
	`tlscertfile`	Path		The full path to the certificate file to use. Asterisk only supports the `.pem` format.
	`tlsprivatekey`	Path		The full path to the private key file. Asterisk only supports the `.pem` format. If this is not specified, the certificate specified in `tlscertfile` will be searched for the private key.

Example http.conf

http.conf

```
[general]
enabled = yes
bindaddr = 0.0.0.0
bindport = 8088
```

Use TLS!
It is **highly** recommended that you encrypt your HTTP signalling with TLS, and use secure WebSockets (WSS) for your events. This requires configuring the TLS information in `http.conf`, and establishing secure websocket/secure HTTP connections from your ARI application.

ARI Configuration

ARI users and properties are configured via `ari.conf`. Note that all options may not be listed here; this listing includes the most useful ones for configuring users in ARI. For a full description, see the ARI configuration documentation.

Section	Parameter	Type	Default	Description
`general`				
	`enabled`	Boolean	Yes	Enable/disable ARI.
	`pretty`	Boolean	No	Format JSON responses and events in a human readable form. This makes the output easier to read, at the cost of some additional bytes.
	`allowed_origins`	String		A comma separated list of allowed origins for Cross-Origin Resource Sharing.
[user_name]				
	`type`	String		Must be `user`. Specifies that this configuration section defines a user for ARI.
	`read_only`	Boolean	No	Whether or not the user can issue requests that alter the Asterisk system. If set to Yes, then only `GET` and `OPTIONS` HTTP requests will be serviced.
	`password_format`	String	plain	Can be either `plain` or `crypt`. When the password is plain, Asterisk will expect the user's password to be in plain text in the `password` field. When set to `crypt`, Asterisk will use `crypt(3)` to decrypt the password. A crypted password can be generated using `mkpasswd -m sha-512`.
	`password`	String		The password for the user.

Example ari.conf

ari.conf

```
[general]
enabled = yes
pretty = yes
allowed_origins = localhost:8088,http://ari.asterisk.org

[asterisk]
type = user
read_only = no
password = asterisk

; password_format may be set to plain (the default) or crypt. When set to crypt,
; crypt(3) is used to validate the password. A crypted password can be generated
; using mkpasswd -m sha-512.
;
[asterisk-supersecret]
type = user
read_only = no
password =
$6$nqvAB8Bvs1dJ4V$8zCUygFXuXXp8EU3t2M8i.N8iCsY4WRchxe2AYgGOzHAQrmjIPif3DYrvdj5U2CilLLMCht
mFyvFa3XHSxBlB/
password_format = crypt
```

Configuring the Dialplan for ARI

By default, ARI cannot just manipulate any arbitrary channel in Asterisk. That channel may be in a long running dialplan application; it may be controlled by an AGI; it may be hung up! Many operations that ARI exposes would be fundamentally unsafe if Asterisk did not hand control of the channel over to ARI in a safe fashion.

To hand a channel over to ARI, Asterisk uses a dialplan application called Stasis. Stasis acts as any other dialplan application in Asterisk, except that it does not do anything to the channel other than safely pass control over to an ARI application. The Stasis dialplan application takes in two parameters:

1. The name of the ARI application to hand the channel over to. Multiple ARI applications can exist with a single instance of Asterisk, and each ARI application will only be able to manipulate the channels that it controls.
2. Optionally, arguments to pass to the ARI application when the channel is handed over.

Example: Two ARI Applications

This snippet of dialplan, taken from `extensions.conf`, illustrates two ARI applications. The first hands a channel over to an ARI application "Intro-IVR" without any additional parameters; the second hands a channel over to an ARI application "Super-Conference" with a parameter that specifies a conference room to enter.

extensions.conf

```
[default]

exten => ivr,1,NoOp()
 same =>     n,Stasis(Intro-IVR)
 same =>     n,Hangup()

exten => conference,1,NoOp()
 same =>            n,Stasis(Super-Conference,100)
 same =>            n,Hangup()
```

When a channel enters into a Stasis application, Asterisk will check to see if a WebSocket connection has been established for that application. If so, the channel is handed over to ARI for control, a subscription for the channel is made for the WebSocket, and a StasisStart event is sent to the WebSocket notifying it that a channel has entered into its application.

A WebSocket connection is necessary!
If you have not connected a WebSocket to Asterisk for a particular application, when a channel enters into Stasis for that application, Asterisk will immediately eject the channel from the application and return back to the dialplan. This is to prevent channels from entering into an application before something is ready to handle them.

Note that if a connection is broken, Asterisk will know that a connection previously existed and will allow channels to enter (although you may got warned that events are about to get missed...)

Introduction to ARI and Channels

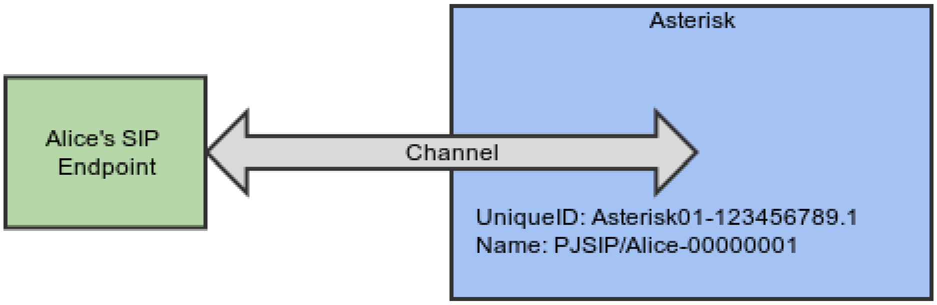
Channels: An Overview

In Asterisk, a channel is a patch of communication between some endpoint and Asterisk itself. The path of communication encompasses all information passed to and from the endpoint. That includes both the signalling (such as "change the state of the device to ringing" or "hangup this call") as well as media (the actual audio or video being sent/received to/from the endpoint).

When a channel is created in Asterisk to represent this path of communication, Asterisk assigns it both a **UniqueID** - which acts as a handle to the channel for its entire lifetime - as well as a unique **Name**. The UniqueID can be any globally unique identifier provided by the ARI client. If the ARI client does not provide a UniqueID to the channel, then Asterisk will assign one to the channel itself. By default, it uses an epoch timestamp with a monotonically increasing integer, optionally along with the Asterisk system name.

Channels to Endpoints

The channel name consists of two parts: the type of channel being created, along with a descriptive identifier determined by the channel type. What channel types are available depends on how the Asterisk system is configured; for the purposes of most examples, we will use "PJSIP" channels to communicate with SIP devices.



In the above diagram, Alice's SIP device has called into Asterisk, and Asterisk has assigned the resulting channel a UniqueID of **Asterisk01-123456789.1**, while the PJSIP channel driver has assigned a name of **PJSIP/Alice-00000001**. In order to manipulate this channel, ARI operations would use the UniqueID Asterisk01-123456789.1 as the handle to the channel.

On This Page

Channels In Depth

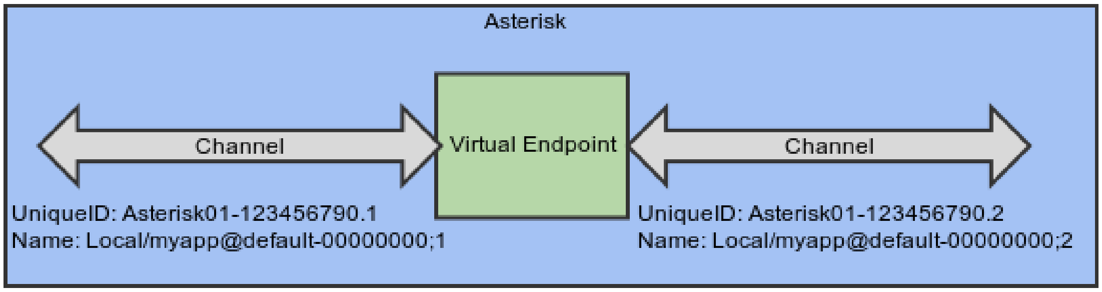
- ARI and Channels: Manipulating Channel State
- ARI and Channels: Simple Media Manipulation
- ARI and Channels: Handling DTMF

Internal Channels - Local Channels

While most channels are between some external endpoint and Asterisk, Asterisk can also create channels that are completely internal within itself. These channels - called **Local** channels - help to move media between various resources within Asterisk.

Local channel are special in that Local channels always come in pairs of channels. Creating a single Local "channel" will *always* result in two channels being created in Asterisk. Sitting between the Local channel pairs is a special virtual endpoint that forwards media back and forth between the two Local channel pairs. One end of each Local channel is permanently tied to this virtual endpoint and cannot be moved about - however, the other end of each

Local channel can be manipulated in any fashion. All media that enters into one of the Local channel halves is passed out through the other Local channel half, and vice versa.


Asterisk
Channel
Virtual Endpoint
Channel
UniqueID: Asterisk01-123456790.1
Name: Local/myapp@default-00000000;1
UniqueID: Asterisk01-123456790.2
Name: Local/myapp@default-00000000;2

In the above diagram, ARI has created a Local channel, `Local/myapp@default`. As a result, Asterisk has created a pair of Local channels with the UniqueIDs of **Asterisk01-123456790.1** and **Asterisk01-123456790.2**. The names of the Local channel halves are **Local/myapp@default-00000000;1** and **Local/myapp@default-00000000;2** - where the ;1 and ;2 denote the two halves of the Local channel.

Channels in a Stasis Application

When a channel is created in Asterisk, it begins to execute dialplan. All channels enter into the dialplan at some location defined by a **context/extension/priority** tuple. Each tuple location in the dialplan defines some Asterisk application that the channel should go execute. When the application completes, the priority in the tuple is increased by one, and the next location in the dialplan is executed. This continues until the dialplan runs out of things to execute, the dialplan application tells the channel to hangup, or until the device itself hangs up.

Channels are handed over to ARI through the Stasis dialplan application. This special application takes control of the channel from the dialplan, and indicates to an ARI client with a connected websocket that a channel is now ready to be controlled. When this occurs, a StasisStart event is emitted; when the channel leaves the Stasis dialplan application - either because it was told to leave or because the device hung up - a StasisEnd event is emitted. When the StasisEnd event is emitted, ARI no longer controls the channel and the channel is handed back over to the dialplan.

Resources in Asterisk do not, by default, send events about themselves to a connected ARI application. In order to get events about resources, one of three things must occur:

1. The resource must be a channel that entered into a Stasis dialplan application. A subscription is implicitly created in this case. The subscription is implicitly destroyed when the channel leaves the Stasis dialplan application.
2. While a channel is in a Stasis dialplan application, the channel may interact with other resources - such as a bridge. While channels interact with the resource, a subscription is made to that resource. When no more channels in a Stasis dialplan application are interacting with the resource, the implicit subscription is destroyed.
3. At any time, an ARI application may make a subscription to a resource in Asterisk through application operations. While that resource exists, the ARI application owns the subscription.

Example: Interacting with Channels

For this example, we're going to write an ARI application that will do the following:

1. When it connects, it will print out the names of all existing channels. If there are no existing channels, it will tell us that as well.
2. When a channel enters into its Stasis application, it will print out all of the specific information about that channel.
3. When a channel leaves its Stasis application, it will print out that the channel has left.

Dialplan

The dialplan for this will be very straight forward: a simple extension that drops a channel into Stasis.

extensions.conf

```
[default]

exten => 1000,1,NoOp()
 same =>      n,Answer()
 same =>      n,Stasis(channel-dump)
 same =>      n,Hangup()
```

Python

For our Python examples, we will rely primarily on the ari-py library. Because the `ari` library will emit useful information using Python logging, we should go ahead and set that up as well - for now, a `basicConfig` with `ERROR` messages displayed should be sufficient. Finally, we'll need to get a client made by initiating a connection to Asterisk. This occurs using the `ari.connect` method, where we have to specify three things:

1. The HTTP base URI of the Asterisk server to connect to. Here, we assume that this is running on the same machine as the script, and that we're using the default port for Asterisk's HTTP server - `8088`.
2. The username of the ARI user account to connect as. In this case, we're specifying it as `asterisk`.
3. The password for the ARI user account. In this case, that's asterisk.

> Modify the connection credentials as appropriate for your server, although many examples will use these credentials.
>
> **Please don't use these credentials in production systems!**

```
 #!/usr/bin/env python

import ari
import logging

logging.basicConfig(level=logging.ERROR)

client = ari.connect('http://localhost:8088', 'asterisk', 'asterisk')
```

Once we've made our connection, our first task is to print out all existing channels or - if there are no channels - print out that there are no channels. The `channels` resource has an operation for this - `GET /channels`. Since the `ari-py` library will dynamically construct operations on objects that map to resource calls using the nickname of an operation, we can use the `list` method on the `channels` resource to get all current channels in Asterisk:

```
current_channels = client.channels.list()
if (len(current_channels) == 0):
    print "No channels currently :-("
else:
    print "Current channels:"
    for channel in current_channels:
        print channel.json.get('name')
```

The `GET /channels` operation returns back a list of `Channel` resources. Those resources, however, are returned as JSON from the operation, and while the `ari-py` library converts the `uniqueid` of those into an attribute on the object, it leaves the rest of them in the JSON dictionary. Since what we want is the name, we can just extract it ourselves out of the JSON and print it out.

Our next step involves a bit more - we want to print out all the information about a channel when it enters into our Stasis dialplan application "channel-dump" and print the channel name when it leaves. To do that, we need to subscribe for the `StasisStart` and `StasisEnd` events:

```
client.on_channel_event('StasisStart', stasis_start_cb)
client.on_channel_event('StasisEnd', stasis_end_cb)
```

We need two handler functions - `stasis_start_cb` for the `StasisStart` event and `stasis_end_cb` for the `StasisEnd` event:

```
def stasis_start_cb(channel_obj, ev):
    """Handler for StasisStart event"""

    channel = channel_obj.get('channel')
    print "Channel %s has entered the application" % channel.json.get('name')

    for key, value in channel.json.items():
        print "%s: %s" % (key, value)

def stasis_end_cb(channel, ev):
    """Handler for StasisEnd event"""

    print "%s has left the application" % channel.json.get('name')
```

Finally, we need to tell the `client` to run our application. Once we call `client.run`, the websocket connection will be made and our application will wait on events infinitely. We can use `Ctrl+C` to kill it and break the connection.

```
client.run(apps='channel-dump')
```

channel-dump.py

The full source code for `channel-dump.py` is shown below:

channel-dump.py

```
#!/usr/bin/env python

import ari
import logging

logging.basicConfig(level=logging.ERROR)

client = ari.connect('http://localhost:8088', 'asterisk', 'asterisk')

current_channels = client.channels.list()
if (len(current_channels) == 0):
    print "No channels currently :-("
else:
    print "Current channels:"
    for channel in current_channels:
        print channel.json.get('name')

def stasis_start_cb(channel_obj, ev):
    """Handler for StasisStart event"""

    channel = channel_obj.get('channel')
    print "Channel %s has entered the application" % channel.json.get('name')

    for key, value in channel.json.items():
        print "%s: %s" % (key, value)

def stasis_end_cb(channel, ev):
    """Handler for StasisEnd event"""

    print "%s has left the application" % channel.json.get('name')

client.on_channel_event('StasisStart', stasis_start_cb)
client.on_channel_event('StasisEnd', stasis_end_cb)

client.run(apps='channel-dump')
```

channel-dump.py in action

Here's sample output from `channel-dump.py`. When it first connects there are no channels in Asterisk - ☹ - but afterwards a PJSIP channel from Alice enters into extension 1000. This prints out all the information about her channels. After hearing silence for awhile, she hangs up - and our script notifies us that her channel has left the application.

```
asterisk:~$ python channel-dump.py
No channels currently :-(
Channel PJSIP/alice-00000001 has entered the application
accountcode:
name: PJSIP/alice-00000001
caller: {u'Alice': u'', u'6575309': u''}
creationtime: 2014-06-09T17:36:31.698-0500
state: Up
connected: {u'name': u'', u'number': u''}
dialplan: {u'priority': 3, u'exten': u'1000', u'context': u'default'}
id: asterisk-01-1402353503.1
PJSIP/alice-00000001 has left the application
```

JavaScript (Node.js)

For our JavaScript examples, we will rely primarily on the Node.js ari-client library. We'll need to get a client made by initiating a connection to Asterisk. This occurs using the `ari.connect` method, where we have to specify four things:

1. The HTTP base URI of the Asterisk server to connect to. Here, we assume that this is running on the same machine as the script, and that we're using the default port for Asterisk's HTTP server - `8088`.
2. The username of the ARI user account to connect as. In this case, we're specifying it as `asterisk`.
3. The password for the ARI user account. In this case, that's asterisk.
4. A callback that will be called with an error if one occurred, followed by an instance of an ARI client.

Modify the connection credentials as appropriate for your server, although many examples will use these credentials.

Please don't use these credentials in production systems!

```
/*jshint node:true*/
'use strict';

var ari = require('ari-client');
var util = require('util');

ari.connect('http://localhost:8088', 'asterisk', 'asterisk', clientLoaded);

// handler for client being loaded
function clientLoaded (err, client) {
  if (err) {
    throw err;
  }
}
```

Once we've made our connection, our first task is to print out all existing channels or - if there are no channels - print out that there are no channels. The `channels` resource has an operation for this - `GET /channels`. Since the `ari-client` library will dynamically construct a client with operations on objects that map to resource calls using the nickname of an operation, we can use the `list` method on the `channels` resource to get all current channels in Asterisk:

```
client.channels.list(function(err, channels) {
  if (!channels.length) {
    console.log('No channels currently :-(');
  } else {
    console.log('Current channels:');
    channels.forEach(function(channel) {
      console.log(channel.name);
    });
  }
});
```

The `GET /channels` operation expects a callback that will be called with an error if one occurred and a list of `Channel` resources. `ari-client` will return a JavaScript object for each `Channel` resource. Properties such as `name` can be accessed on the object directly.

Our next step involves a bit more - we want to print out all the information about a channel when it enters into our Stasis dialplan application "channel-dump" and print the channel name when it leaves. To do that, we need to subscribe for the `StasisStart` and `StasisEnd` events:

```
client.on('StasisStart', stasisStart);
client.on('StasisEnd', stasisEnd);
```

We need two callback functions - `stasisStart` for the `StasisStart` event and `stasisEnd` for the `StasisEnd` event:

```
// handler for StasisStart event
function stasisStart(event, channel) {
  console.log(util.format(
      'Channel %s has entered the application', channel.name));
  // use keys on event since channel will also contain channel operations
  Object.keys(event.channel).forEach(function(key) {
    console.log(util.format('%s: %s', key, JSON.stringify(channel[key])));
  });
}
// handler for StasisEnd event
function stasisEnd(event, channel) {
  console.log(util.format(
      'Channel %s has left the application', channel.name));
}
```

Finally, we need to tell the `client` to start our application. Once we call `client.start`, a websocket connection will be established and the client will emit Node.js events as events come in through the websocket. We can use `Ctrl+C` to kill it and break the connection.

```
client.start('channel-dump');
```

channel-dump.js

The full source code for `channel-dump.js` is shown below:

```
/*jshint node:true*/
'use strict';

var ari = require('ari-client');
var util = require('util');

ari.connect('http://localhost:8088', 'asterisk', 'asterisk', clientLoaded);

// handler for client being loaded
function clientLoaded (err, client) {
  if (err) {
    throw err;
  }

  client.channels.list(function(err, channels) {
    if (!channels.length) {
      console.log('No channels currently :-(');
    } else {
      console.log('Current channels:');
      channels.forEach(function(channel) {
        console.log(channel.name);
      });
    }
  });

  // handler for StasisStart event
  function stasisStart(event, channel) {
    console.log(util.format(
        'Channel %s has entered the application', channel.name));

    // use keys on event since channel will also contain channel operations
    Object.keys(event.channel).forEach(function(key) {
      console.log(util.format('%s: %s', key, JSON.stringify(channel[key])));
    });
  }

  // handler for StasisEnd event
  function stasisEnd(event, channel) {
    console.log(util.format(
        'Channel %s has left the application', channel.name));
  }

  client.on('StasisStart', stasisStart);
  client.on('StasisEnd', stasisEnd);

  client.start('channel-dump');
}
```

channel-dump.js in action

Here's sample output from `channel-dump.js`. When it first connects there are no channels in Asterisk - ☹ - but afterwards a PJSIP channel from Alice enters into extension 1000. This prints out all the information about her channels. After hearing silence for a while, she hangs up - and our script notifies us that her channel has left the application.

```
asterisk:~$ node channel-dump.js
No channels currently :-(
Channel PJSIP/alice-00000001 has entered the application
accountcode:
name: PJSIP/alice-00000001
caller: {u'Alice': u'', u'6575309': u''}
creationtime: 2014-06-09T17:36:31.698-0500
state: Up
connected: {u'name': u'', u'number': u''}
dialplan: {u'priority': 3, u'exten': u'1000', u'context': u'default'}
id: asterisk-01-1402353503.1
PJSIP/alice-00000001 has left the application
```

ARI and Channels: Manipulating Channel State

Channel State

A channel's state reflects the current state of the path of communication between Asterisk and a device. What state a channel is in also affects what operations are allowed on it and/or how certain operations will affect a device.

While there are many states a channel can be in, the following are the most common:

- **Down** - a path of communication exists or used to exist between Asterisk and the device, but no media can flow between the two.
- **Ringing** - the device is ringing. Media may or may not be able to flow from Asterisk to the device.
- **Up** - the device has been answered. When in the up state, media can flow bidirectionally between Asterisk and the device.

⚠ **More Channel States**
Certain channel technologies, such as DAHDI analog channels, may have additional channel states (such as "Pre-ring" or "Dialing Offhook"). When handling channel state, consult the Channel data model for all possible values.

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Indicating Ringing

Asterisk can inform a device that it should start playing a ringing tone back to the caller using the `POST /channels/{channel_id}/ring` operation. Likewise, ringing can be stopped using the `DELETE /channels/{channel_id}/ring` operation. Note that indicating ringing typically does not actually transmit media from Asterisk to the device in question - Asterisk merely signals the device to ring. It is up to the device itself to actually play something back for the user.

Answering a Channel

When a channel isn't answered, Asterisk has typically not yet informed the device how it will communicate with it. Answering a channel will cause Asterisk to complete the path of communication, such that media flows bi-directionally between the device and Asterisk.

You can answer a channel using the `POST /channels/{channel_id}/answer` operation.

Hanging up a channel

You can hang up a channel using the `DELETE /channels/{channel_id}` operation. When this occurs, the path of communication between Asterisk and the device is terminated, and the channel will leave the Stasis application. Your application will be notified of this via a StasisEnd event.

The same is true if the device initiates the hang up. In the same fashion, the path of communication between Asterisk and the device is terminated, the channel is hung up, and your application is informed that the channel is leaving your application via a StasisEnd event.

Generally, once a channel leaves your application, you won't receive any more events about the channel. There are times, however, when you may be subscribed to all events coming from a channel - regardless if that channel is in your application or not. In that case, a ChannelDestroyed event will inform you when the channel is well and truly dead.

Example: Manipulating Channel State

For this example, we're going to write an ARI application that will do the following:

1. Wait for a channel to enter its Stasis application.
2. When a channel enters its Stasis application, it will indicate ringing to the channel. If the channel wasn't already ringing, it will now!
3. After a few seconds, it will answer the channel.
4. Once the channel is answered, we'll start silence on the channel so that the user feels a comfortable whishing noise. Then, after a few more seconds, we'll hangup the channel.
5. If at any point in time the phone hangs up first, we'll gracefully handle that.

Dialplan

For this example, we need to just drop the channel into Stasis, specifying our application:

extensions.conf

```
exten => 1000,1,NoOp()
 same =>      n,Stasis(channel-state)
 same =>      n,Hangup()
```

Python

This example will use the ari-py library. The basic structure is very similar to the channel-dump Python example - see that example for more information on the basics of setting up an ARI connection using this library.

To start, once our ARI client has been set up, we will want to register handlers for three different events - `StasisStart`, `ChannelStateChange`, and `StasisEnd`.

1. The bulk of the work will be done in `StasisStart`, which is called when the channel enters our application. For the most part, this will involve setting up Python timers to initiate actions on the channel.
2. The `ChannelStateChange` handler will merely print out the channel state changes for us, which is informative as it will tell us when the channel is answered.
3. Finally, the `StasisEnd` event will clean up for us by cancelling any pending timers that we initiated. This will get called when the channel leaves our application - which will happen when the user hangs up the channel, or when we hang up the channel.

We can store the timers that we've set up for a channel using a dictionary of channel IDs to timer instances:

```
channel_timers = {}
```

And we can register for our three events:

```
client.on_channel_event('StasisStart', stasis_start_cb)
client.on_channel_event('ChannelStateChange', channel_state_change_cb)
client.on_channel_event('StasisEnd', stasis_end_cb)
```

The `StasisStart` event is the most interesting part.

1. First, we tell the channel to ring, and after two seconds, to answer the channel:

   ```
   channel.ring()
       # Answer the channel after 2 seconds
       timer = threading.Timer(2, answer_channel, [channel])
       channel_timers[channel.id] = timer
       timer.start()
   ```

   If we didn't have that there, then the caller would probably just have dead space to listen to! Not very enjoyable. We store the timer in the `channel_timers` dictionary so that our `StasisEnd` event can cancel it for us if the user hangs up the phone.
2. Once we're in the `answer_channel` handler, we answer the channel and start silence on the channel. That (hopefully) gives them a slightly more ambient silence noise. Note that we'll go ahead and declare `answer_channel` as a nested function inside our `StasisStart` handler, `stasis_start_cb`:

   ```
   def stasis_start_cb(channel_obj, ev):
       """Handler for StasisStart event"""

       def answer_channel(channel):
           """Callback that will actually answer the channel"""
           print "Answering channel %s" % channel.json.get('name')
           channel.answer()
           channel.startSilence()
   ```

3. After we've answered the channel, we kick off another Python timer to hang up the channel in 4 seconds. When that timer fires, it will call `hangup_channel`. This does the final action on the channel by hanging it up. Again, we'll declare `hangup_channel` as a nested function inside our `Sta`

`sisStart` handler:

```
def hangup_channel(channel):
        """Callback that will actually hangup the channel"""

        print "Hanging up channel %s" % channel.json.get('name')
        channel.hangup()
```

When we create a timer - such as when we started ringing on the channel - we stored it in our `channel_timers` dictionary. In our `StasisEnd` event handler, we'll want to cancel any pending timers. Otherwise, our timers may fire and try to perform an action on channel that has already left our Stasis application, which is a good way to get an HTTP error response code.

```
def stasis_end_cb(channel, ev):
   """Handler for StasisEnd event"""

   print "Channel %s just left our application" % channel.json.get('name')

   # Cancel any pending timers
   timer = channel_timers.get(channel.id)
   if timer:
       timer.cancel()
       del channel_timers[channel.id]
```

Finally, we want to print out the state of the channel in the `ChannelStateChanged` handler. This will tell us exactly when our channel has been answered:

```
def channel_state_change_cb(channel, ev):
   """Handler for changes in a channel's state"""
   print "Channel %s is now: %s" % (channel.json.get('name'),
                                    channel.json.get('state'))
```

channel-state.py

The full source code for `channel-state.py` is shown below:

channel-state.py

```
#!/usr/bin/env python

import ari
import logging
import threading

logging.basicConfig(level=logging.ERROR)

client = ari.connect('http://localhost:8088', 'asterisk', 'asterisk')

channel_timers = {}

def stasis_end_cb(channel, ev):
    """Handler for StasisEnd event"""

    print "Channel %s just left our application" % channel.json.get('name')

    # Cancel any pending timers
    timer = channel_timers.get(channel.id)
    if timer:
        timer.cancel()
        del channel_timers[channel.id]
```

```
def stasis_start_cb(channel_obj, ev):
    """Handler for StasisStart event"""

    def answer_channel(channel):
        """Callback that will actually answer the channel"""
        print "Answering channel %s" % channel.json.get('name')
        channel.answer()
        channel.startSilence()

        # Hang up the channel in 4 seconds
        timer = threading.Timer(4, hangup_channel, [channel])
        channel_timers[channel.id] = timer
        timer.start()

     def hangup_channel(channel):
        """Callback that will actually hangup the channel"""

        print "Hanging up channel %s" % channel.json.get('name')
        channel.hangup()

    channel = channel_obj.get('channel')
    print "Channel %s has entered the application" % channel.json.get('name')

    channel.ring()
    # Answer the channel after 2 seconds
    timer = threading.Timer(2, answer_channel, [channel])
    channel_timers[channel.id] = timer
    timer.start()

def channel_state_change_cb(channel, ev):
    """Handler for changes in a channel's state"""
    print "Channel %s is now: %s" % (channel.json.get('name'),
                                     channel.json.get('state'))

client.on_channel_event('StasisStart', stasis_start_cb)
client.on_channel_event('ChannelStateChange', channel_state_change_cb)
client.on_channel_event('StasisEnd', stasis_end_cb)

client.run(apps='channel-state')
```

channel-state.py in action

Here, we see the output from the `channel-state.py` script when a PJSIP channel for endpoint 'alice' enters into the application:

```
Channel PJSIP/alice-00000001 has entered the application
Answering channel PJSIP/alice-00000001
Channel PJSIP/alice-00000001 is now: Up
Hanging up channel PJSIP/alice-00000001
Channel PJSIP/alice-00000001 just left our application
```

JavaScript (Node.js)

This example will use the ari-client library. The basic structure is very similar to the channel-dump JavaScript example - see that example for more information on the basics of setting up an ARI connection using this library.

To start, once our ARI client has been set up, we will want to register callbakcs for three different events - `StasisStart`, `ChannelStateChange`, and `StasisEnd`.

1. The bulk of the work will be done in `StasisStart`, which is called when the channel enters our application. For the most part, this will involve setting up JavaScript timeouts to initiate actions on the channel.
2. The `ChannelStateChange` handler will merely print out the channel state changes for us, which is informative as it will tell us when the channel is answered.
3. Finally, the `StasisEnd` event will clean up for us by cancelling any pending timeouts that we initiated. This will get called when the channel leaves our application - which will happen when the user hangs up the channel, or when we hang up the channel.

We can store the timeouts that we've set up for a channel using an object of channel IDs to timer instances:

```
timers = {}
```

And we can register for our three events:

```
client.on_channel_event('StasisStart', stasis_start_cb)
client.on_channel_event('ChannelStateChange', channel_state_change_cb)
client.on_channel_event('StasisEnd', stasis_end_cb)
```

The `StasisStart` event is the most interesting part.

1. First, we tell the channel to ring, and after two seconds, to answer the channel:

   ```
   channel.ring(function(err) {
     if (err) {
       throw err;
     }
   });
   // answer the channel after 2 seconds
   var timer = setTimeout(answer, 2000);
   timers[channel.id] = timer;
   ```

   If we didn't have that there, then the caller would probably just have dead space to listen to! Not very enjoyable. We store the timer in the `timers` object so that our `StasisEnd` event can cancel it for us if the user hangs up the phone.
2. Once we're in the `answer` callback, we answer the channel and start silence on the channel. That (hopefully) gives them a slightly more ambient silence noise:

```
// callback that will answer the channel
function answer() {
  console.log(util.format('Answering channel %s', channel.name));
  channel.answer(function(err) {
    if (err) {
      throw err;
    }
  });
  channel.startSilence(function(err) {
    if (err) {
      throw err;
    }
  });
  // hang up the channel in 4 seconds
  var timer = setTimeout(hangup, 4000);
  timers[channel.id] = timer;
}
```

3. After we've answered the channel, we kick off another timer to hang up the channel in 4 seconds. When that timer fires, it will call `the hangup callback`. This does the final action on the channel by hanging it up:

```
// callback that will hangup the channel
function hangup() {
  console.log(util.format('Hanging up channel %s', channel.name));
  channel.hangup(function(err) {
    if (err) {
      throw err;
    }
  });
}
```

When we create a timer - such as when we started ringing on the channel - we stored it in our `timers` object. In our `StasisEnd` event handler, we'll want to cancel any pending timers. Otherwise, our timers may fire and try to perform an action on channel that has already left our Stasis application, which is a good way to get an HTTP error response code.

```
// handler for StasisEnd event
function stasisEnd(event, channel) {
  console.log(util.format(
        'Channel %s just left our application', channel.name));
  var timer = timers[channel.id];
  if (timer) {
    clearTimeout(timer);
    delete timers[channel.id];
  }
}
```

Finally, we want to print out the state of the channel in the `ChannelStateChanged` callback. This will tell us exactly when our channel has been answered:

```
// handler for ChannelStateChange event
function channelStateChange(event, channel) {
  console.log(util.format(
        'Channel %s is now: %s', channel.name, channel.state));
}
```

channel-state.js

The full source code for `channel-state.js` is shown below:

channel-state.js

```
/*jshint node: true*/
'use strict';

var ari = require('ari-client');
var util = require('util');

var timers = {};
ari.connect('http://localhost:8088', 'asterisk', 'asterisk', clientLoaded);

// handler for client being loaded
function clientLoaded (err, client) {
  if (err) {
    throw err;
  }

  // handler for StasisStart event
  function stasisStart(event, channel) {
    console.log(util.format(
          'Channel %s has entered the application', channel.name));

    channel.ring(function(err) {
      if (err) {
        throw err;
      }
    });
    // answer the channel after 2 seconds
    var timer = setTimeout(answer, 2000);
    timers[channel.id] = timer;

    // callback that will answer the channel
    function answer() {
      console.log(util.format('Answering channel %s', channel.name));
      channel.answer(function(err) {
        if (err) {
          throw err;
        }
      });
      channel.startSilence(function(err) {
        if (err) {
          throw err;
        }
      });
      // hang up the channel in 4 seconds
      var timer = setTimeout(hangup, 4000);
      timers[channel.id] = timer;
    }

    // callback that will hangup the channel
    function hangup() {
      console.log(util.format('Hanging up channel %s', channel.name));
      channel.hangup(function(err) {
        if (err) {
          throw err;
        }
      });
    }
  }

  // handler for StasisEnd event
  function stasisEnd(event, channel) {
    console.log(util.format(
          'Channel %s just left our application', channel.name));
    var timer = timers[channel.id];
    if (timer) {
      clearTimeout(timer);
      delete timers[channel.id];
    }
  }

  // handler for ChannelStateChange event
  function channelStateChange(event, channel) {
    console.log(util.format(
          'Channel %s is now: %s', channel.name, channel.state));
  }

  client.on('StasisStart', stasisStart);
  client.on('StasisEnd', stasisEnd);
  client.on('ChannelStateChange', channelStateChange);
```

```

  client.start('channel-state');
}
```

channel-state.js in action

Here, we see the output from the `channel-state.js` script when a PJSIP channel for endpoint 'alice' enters into the application:

```
Channel PJSIP/alice-00000001 has entered the application
Answering channel PJSIP/alice-00000001
Channel PJSIP/alice-00000001 is now: Up
Hanging up channel PJSIP/alice-00000001
Channel PJSIP/alice-00000001 just left our application
```

ARI and Channels: Simple Media Manipulation

Simple media playback

Almost all media is played to a channel using the `POST /channels/{channel_id}/play` operation. This will do the following:

1. Create a new `Playback` object for the channel. If a media operation is currently in progress on the channel, the new `Playback` object will be queued up for the channel.
2. The `media` URI passed to the `play` operation will be inspected, and Asterisk will attempt to find the media requested. Currently, the following media schemes are supported:

URI Scheme	Description
`sound`	A sound file located on the Asterisk system. You can use the `/sounds` resource to query for available sounds on the system. You can also use specify a media file which is consumed via HTTP (e.g sound:http://foo.com/sound.wav)
`recording`	A `StoredRecording` stored on the Asterisk system. You can use the `/recordings/stored` resource to query for available `StoredRecordings` on the system.
`number`	Play back the specified number. This uses the same mechanism as Asterisk's `Say` family of applications.
`digits`	Play back the specified digits. This uses the same mechanism as Asterisk's `Say` family of applications.
`characters`	Play back the specified characters. This uses the same mechanism as Asterisk's `Say` family of applications.
`tone`	Play a particular tone sequence until stopped. This can be used to play locale specific ringing, stutter, busy, congestion, or other tones to a device.

3. Once the media operation is started or enqueued, the `Playback` object will be returned to the caller in the `HTTP` response to the request. The caller can use that playback object to manipulate the media operation.

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Specify your own ID!
It is **highly** recommended that the `POST /channels/{channel_id}/play/{playback_id}` operation be used instead of the `POST /channels/{channel_id}/play` variant. Asterisk lives in an asynchronous world - which is the same world you and I live in. Sometimes, if things happen *very* quickly, you may get notifications over the WebSocket about things you have started before the HTTP response completes!

When you specify your own ID, you have the ability to tie information coming from events back to whatever operation you initiated - if you so choose to. If you use the non-ID providing variant, Asterisk will happily generate a UUID for your `Playback` object - but then it is up to you to deal with whatever information comes back from the WebSocket.

Early Media

Generally, before a channel has been answered and transitioned to the Up state, media cannot pass between Asterisk and a device. For example, if Asterisk is placing an outbound call to a device, the device may be ringing but no one has picked up a handset yet! In such circumstances, media cannot be successfully played to the ringing device - after all, who could listen to it?

However, with inbound calls, Asterisk is the entity that decides when the path of communication between itself and the device is answered - not the user on the other end. This can be useful when answering the channel may trigger billing times or other mechanisms that we don't want to fire yet. This is called "early media". For the channel technologies that support this, ARI and Asterisk will automatically handle sending the correct indications to the ringing phone before sending it media. The same `play` operation can be used both for "regular" playback of media, as well as for "early media" scenarios.

Example: Playing back tones

This example ARI application will do the following:

1. When a channel enters into the Stasis application, it will start a playback of a French ringing tone.
2. After 8 seconds, the channel will be answered.
3. After 1 second, the channel will be rudely hung up on - we didn't want to talk to them anyway!

Dialplan

For this example, we need to just drop the channel into Stasis, specifying our application:

extensions.conf

```
exten => 1000,1,NoOp()
 same =>      n,Stasis(channel-tones)
 same =>      n,Hangup()
```

Python

This example will use a very similar structure as the `channel-state.py` example. Instead of performing a `ring` operation in our `StasisStart` handler, we'll instead initiate a playback using the `playWithId` operation on the channel. Note that our URI uses the `tone` scheme, which supports an optional `tonezone` parameter. We specify our `tonezone` as `fr`, so that we get an elegant French ringing tone. Much like the `channel-state.py` example, we then use a Python timer to schedule a callback that will answer the channel. Since we care about both the `channel` and the `playback` initiated on it, we pass both parameters as `*args` parameters to the callback function.

```
playback_id = str(uuid.uuid4())
    playback = channel.playWithId(playbackId=playback_id,
                                  media='tone:ring;tonezone=fr')
    timer = threading.Timer(8, answer_channel, [channel, playback])
```

Since this is a media operation and not *technically* a ringing indication, when we `answer` the channel, the tone playback will not stop! To stop playing back our French ringing tone, we issue a `stop` operation on the `playback` object. This actually maps to a `DELETE /playbacks/{playback_id}` operation.

```
def answer_channel(channel, playback):
        """Callback that will actually answer the channel"""

        print "Answering channel %s" % channel.json.get('name')
        playback.stop()
        channel.answer()
```

Once answered, we'll schedule another Python timer that will do the actual hanging up of the channel.

channel-tones.py

The full source code for `channel-tones.py` is shown below:

channel-tones.py

```
#!/usr/bin/env python
```

```
import ari
import logging
import threading
import uuid

logging.basicConfig(level=logging.ERROR)

client = ari.connect('http://localhost:8088', 'asterisk', 'asterisk')

channel_timers = {}

def stasis_end_cb(channel, ev):
    """Handler for StasisEnd event"""

    print "Channel %s just left our application" % channel.json.get('name')
    timer = channel_timers.get(channel.id)
    if timer:
        timer.cancel()
        del channel_timers[channel.id]

def stasis_start_cb(channel_obj, ev):
    """Handler for StasisStart event"""

    def answer_channel(channel, playback):
        """Callback that will actually answer the channel"""

        print "Answering channel %s" % channel.json.get('name')
        playback.stop()
        channel.answer()

        timer = threading.Timer(1, hangup_channel, [channel])
        channel_timers[channel.id] = timer
        timer.start()

    def hangup_channel(channel):
        """Callback that will actually hangup the channel"""

        print "Hanging up channel %s" % channel.json.get('name')
        channel.hangup()

    channel = channel_obj.get('channel')
    print "Channel %s has entered the application" % channel.json.get('name')

    playback_id = str(uuid.uuid4())
    playback = channel.playWithId(playbackId=playback_id,
                                  media='tone:ring;tonezone=fr')
    timer = threading.Timer(8, answer_channel, [channel, playback])
    channel_timers[channel.id] = timer
    timer.start()


client.on_channel_event('StasisStart', stasis_start_cb)
client.on_channel_event('StasisEnd', stasis_end_cb)


client.run(apps='channel-tones')
```

channel-tones.py in action

The following shows the output of the `channel-tones.js` script when a `PJSIP` channel for `alice` enters the application:

```
Channel PJSIP/alice-00000000 has entered the application
Answering channel PJSIP/alice-00000000
Hanging up channel PJSIP/alice-00000000
Channel PJSIP/alice-00000000 just left our application
```

JavaScript (Node.js)

This example will use a very similar structure as the `channel-state.js` example. Instead of performing a `ring` operation in our `StasisStart` handler, we'll instead initiate a playback using the `play` operation on the channel. Note that our URI uses the `tone` scheme, which supports an optional `tonezone` parameter. We specify our `tonezone` as `fr`, so that we get an elegant French ringing tone. Much like the `channel-state.js` example, we then use a JavaScript timeout to schedule a callback that will answer the channel.

```
var playback = client.Playback();
channel.play({media: 'tone:ring;tonezone=fr'},
             playback, function(err, newPlayback) {
  if (err) {
    throw err;
  }
});
// answer the channel after 8 seconds
var timer = setTimeout(answer, 8000);
timers[channel.id] = timer;
```

Since this is a media operation and not *technically* a ringing indication, when we `answer` the channel, the tone playback will not stop! To stop playing back our French ringing tone, we issue a `stop` operation on the `playback` object. This actually maps to a `DELETE /playbacks/{playback_id}` operation. Notice that we use the fact that the answer callback closes on the original channel and playback variables to access them from the callback.

```
function answer() {
  console.log(util.format('Answering channel %s', channel.name));
  playback.stop(function(err) {
    if (err) {
      throw err;
    }
  });
  channel.answer(function(err) {
    if (err) {
      throw err;
    }
  });
  // hang up the channel in 1 seconds
  var timer = setTimeout(hangup, 1000);
  timers[channel.id] = timer;
}
```

Once answered, we'll schedule another timeout that will do the actual hanging up of the channel.

channel-tones.js

The full source code for `channel-tones.js` is shown below:

channel-tones.js

```
/*jshint node: true*/
'use strict';

var ari = require('ari-client');
var util = require('util');

var timers = {};
ari.connect('http://localhost:8088', 'asterisk', 'asterisk', clientLoaded);

// handler for client being loaded
function clientLoaded (err, client) {
  if (err) {
    throw err;
  }

  // handler for StasisStart event
  function stasisStart(event, channel) {
    console.log(util.format(
          'Channel %s has entered the application', channel.name));

    var playback = client.Playback();
    channel.play({media: 'tone:ring;tonezone=fr'},
                 playback, function(err, newPlayback) {
      if (err) {
        throw err;
      }
    });
    // answer the channel after 8 seconds
    var timer = setTimeout(answer, 8000);
    timers[channel.id] = timer;

    // callback that will answer the channel
    function answer() {
      console.log(util.format('Answering channel %s', channel.name));
      playback.stop(function(err) {
        if (err) {
          throw err;
        }
      });
      channel.answer(function(err) {
        if (err) {
          throw err;
        }
      });
      // hang up the channel in 1 seconds
      var timer = setTimeout(hangup, 1000);
      timers[channel.id] = timer;
    }

    // callback that will hangup the channel
    function hangup() {
      console.log(util.format('Hanging up channel %s', channel.name));
      channel.hangup(function(err) {
        if (err) {
          throw err;
        }
      });
    }
  }

  // handler for StasisEnd event
  function stasisEnd(event, channel) {
    console.log(util.format(
          'Channel %s just left our application', channel.name));
    var timer = timers[channel.id];
    if (timer) {
      clearTimeout(timer);
      delete timers[channel.id];
    }
  }

  client.on('StasisStart', stasisStart);
  client.on('StasisEnd', stasisEnd);
```

```
  client.start('channel-tones');
}
```

channel-tones.js in action

The following shows the output of the `channel-tones.js` script when a `PJSIP` channel for `alice` enters the application:

```
Channel PJSIP/alice-00000000 has entered the application
Answering channel PJSIP/alice-00000000
Hanging up channel PJSIP/alice-00000000
Channel PJSIP/alice-00000000 just left our application
```

Example: Playing back a sound file

This example ARI application will do the following:

1. When a channel enters the Stasis application, initiate a playback of howler monkeys on the channel. Fly my pretties, FLY!
2. If the user has not hung up their phone in panic, it will hang up the channel when the howler monkeys return victorious - or rather, when ARI notifies the application that the playback has finished via the `PlaybackFinished` event.

Dialplan

For this example, we need to just drop the channel into Stasis, specifying our application:

extensions.conf

```
exten => 1000,1,NoOp()
 same =>      n,Stasis(channel-playback-monkeys)
 same =>      n,Hangup()
```

Python

Much like the `channel-tones.py` example, we'll start off by initiating a playback on the channel. Instead of specifying a `tone` scheme, however, we'll specify a scheme of `sound` with a resource of `tt-monkeys`. Unlike the tones, this media *does* have a well defined ending - the end of the sound file! So we'll subscribe for the `PlaybackFinished` event and tell `ari-py` to call `playback_finished` when our monkeys are done attacking.

```
    playback_id = str(uuid.uuid4())
    playback = channel.playWithId(playbackId=playback_id,
                                  media='sound:tt-monkeys')
    playback.on_event('PlaybackFinished', playback_finished)
```

Unfortunately, `ari-py` doesn't let us pass arbitrary data to a callback function in the same fashion as a Python timer. Nuts. Luckily, the `Playback` object has a property, `target_uri`, that tells us which object it just finished playing to. Using that, we can get the `channel` object back from Asterisk so we can hang it up.

```
def playback_finished(playback, ev):
        """Callback when the monkeys have finished howling"""

        target_uri = playback.json.get('target_uri')
        channel_id = target_uri.replace('channel:', '')
        channel = client.channels.get(channelId=channel_id)

        print "Monkeys successfully vanquished %s; hanging them up" %
channel.json.get('name')
        channel.hangup()
```

Note that unlike the `channel-tones.py` example, this application eschews the use of Python timers and simply responds to ARI events as they happen. This means we don't have to do much in our `StasisEnd` event, and we have to track less state.

channel-playback-monkeys.py

The full source code for `channel-playback-monkeys.py` is shown below:

```
 #!/usr/bin/env python

import ari
import logging
import uuid

logging.basicConfig(level=logging.ERROR)

client = ari.connect('http://localhost:8088', 'asterisk', 'asterisk')

def stasis_end_cb(channel, ev):
    """Handler for StasisEnd event"""

    print "Channel %s just left our application" % channel.json.get('name')

def stasis_start_cb(channel_obj, ev):
    """Handler for StasisStart event"""

    def playback_finished(playback, ev):
        """Callback when the monkeys have finished howling"""

        target_uri = playback.json.get('target_uri')
        channel_id = target_uri.replace('channel:', '')
        channel = client.channels.get(channelId=channel_id)

        print "Monkeys successfully vanquished %s; hanging them up" %
channel.json.get('name')
        channel.hangup()

    channel = channel_obj.get('channel')
    print "Monkeys! Attack %s!" % channel.json.get('name')

    playback_id = str(uuid.uuid4())
    playback = channel.playWithId(playbackId=playback_id,
                                  media='sound:tt-monkeys')
    playback.on_event('PlaybackFinished', playback_finished)


client.on_channel_event('StasisStart', stasis_start_cb)
client.on_channel_event('StasisEnd', stasis_end_cb)


client.run(apps='channel-playback-monkeys')
```

channel-playback-monkeys.py in action

The following shows the output of the `channel-playback-monkeys`.py script when a `PJSIP` channel for `alice` enters the application:

```
Monkeys! Attack PJSIP/alice-00000000!
Monkeys successfully vanquished PJSIP/alice-00000000; hanging them up
Channel PJSIP/alice-00000000 just left our application
```

JavaScript (Node.js)

Much like the `channel-tones.js` example, we'll start off by initiating a playback on the channel. Instead of specifying a `tone` scheme, however, we'll specify a scheme of `sound` with a resource of `tt-monkeys`. Unlike the tones, this media *does* have a well defined ending - the end of the sound file! So we'll subscribe for the `PlaybackFinished` event and tell `ari-client` to call `playbackFinished` when our monkeys are done attacking. Notice that we use `client.Playback()` to generate a playback object with a pre-existing Id so we can scope the PlaybackFinished event to the playback we just created.

```
var playback = client.Playback();
channel.play({media: 'sound:tt-monkeys'},
              playback, function(err, newPlayback) {
  if (err) {
    throw err;
  }
});
playback.on('PlaybackFinished', playbackFinished);
```

Notice that we use the fact that the playbackFinished callback closes over the original channel variable to perform a hangup operation using that object directly.

```
function playbackFinished(event, completedPlayback) {
  console.log(util.format(
      'Monkeys successfully vanquished %s; hanging them up',
      channel.name));
  channel.hangup(function(err) {
    if (err) {
      throw err;
    }
  });
}
```

Note that unlike the `channel-tones.js` example, this application eschews the use of JavaScript timeouts and simply responds to ARI events as they happen. This means we don't have to do much in our `StasisEnd` event, and we have to track less state.

channel-playback-monkeys.js

The full source code for `channel-playback-monkeys.js` is shown below:

```
/*jshint node: true*/
'use strict';

var ari = require('ari-client');
var util = require('util');

ari.connect('http://localhost:8088', 'asterisk', 'asterisk', clientLoaded);

// handler for client being loaded
function clientLoaded (err, client) {
  if (err) {
    throw err;
  }

  // handler for StasisStart event
  function stasisStart(event, channel) {
    console.log(util.format(
          'Monkeys! Attack %s!', channel.name));

    var playback = client.Playback();
    channel.play({media: 'sound:tt-monkeys'},
                  playback, function(err, newPlayback) {
      if (err) {
        throw err;
      }
    });
    playback.on('PlaybackFinished', playbackFinished);

    function playbackFinished(event, completedPlayback) {
      console.log(util.format(
          'Monkeys successfully vanquished %s; hanging them up',
          channel.name));
      channel.hangup(function(err) {
        if (err) {
          throw err;
        }
      });
    }
  }

  // handler for StasisEnd event
  function stasisEnd(event, channel) {
    console.log(util.format(
          'Channel %s just left our application', channel.name));
  }

  client.on('StasisStart', stasisStart);
  client.on('StasisEnd', stasisEnd);

  client.start('channel-playback-monkeys');
}
```

channel-playback-monkeys.js in action

The following shows the output of the `channel-playback-monkeys`.js script when a `PJSIP` channel for `alice` enters the application:

```
Monkeys! Attack PJSIP/alice-00000000!
Monkeys successfully vanquished PJSIP/alice-00000000; hanging them up
Channel PJSIP/alice-00000000 just left our application
```

ARI and Channels: Handling DTMF

Handling DTMF events

DTMF events are conveyed via the `ChannelDtmfReceived` event. The event contains the channel that pressed the DTMF key, the digit that was pressed, and the duration of the digit.

While this concept is relatively straight forward, handling DTMF is quite common in applications, as it is the primary mechanism that phones have to inform a server to perform some action. This includes manipulating media, initiating call features, performing transfers, dialling, and just about every thing in between. As such, the examples on this page focus less on simply handling the event and more on using the DTMF in a relatively realistic fashion.

On This Page

Example: A simple automated attendant

This example mimics the automated attendant/IVR dialplan example. It does the following:

- Plays a menu to the user which is cancelled when the user takes some action.
- If the user presses 1 or 2, the digit is repeated to the user and the menu restarted.
- If the user presses an invalid digit, a prompt informing the user that the digit was invalid is played to the user and the menu restarted.
- If the user fails to press anything within some period of time, a prompt asking the user if they are still present is played to the user and the menu restarted.

For this example, you will need the following:

1. The **extra** sound package from Asterisk. You can install this using the `menuselect` tool.
2. If using the Python example, `ari-py` version 0.1.3 or later.
3. If using the JavaScript example, ari-client version 0.1.4 or later.

Dialplan

As usual, a very simple dialplan is sufficient for this example. The dialplan takes the channel and places it into the Stasis application `channel-aa`.

extensions.conf

```
exten => 1000,1,NoOp()
 same =>      n,Stasis(channel-aa)
 same =>      n,Hangup()
```

Python

As this example is a bit larger, how the code is written and structured is broken up into two phases:

1. Constructing the menu and handling its state as the user presses buttons.
2. Actually handling the button presses from the user.

The full source code for this example immediately follows the walk through.

Playing the menu

Unlike Playback, which can chain multiple sounds together and play them back in one continuous operation, ARI treats all sound files being played as separate operations. It will queue each sound file up to be played on the channel, and hand back the caller an object to control the operation of that single sound file. The menu announcement for the attendant has the following requirements:

1. Playback the options for the user
2. If the user presses a DTMF key, cancel the playback of the options and handle the request
3. If the user presses an invalid DTMF key, let them know and restart the menu
4. If the user doesn't press anything, wait 10 seconds, ask them if they are still present, and restart the menu

The second requirement makes this a bit more challenging: when the user presses a DTMF key, we want to cancel whatever sound file is currently being played back and immediately handle their request. We thus have to maintain some state in our application about what sound file is currently being played so that we can cancel the correct playback. We also don't want to queue up all of the sounds immediately - we'd have to walk through all of the queued up sounds and cancel each one - that'd be annoying! Instead, we only want to start the next sound in our prompt when the previous has completed.

To start, we'll define in a list at the top of our script the sounds that make up the initial menu prompt:

```
sounds = ['press-1', 'or', 'press-2']
```

Since we'll want to maintain some state, we'll create a small object to do that for us. In Python, tuples are immutable - and we'll want to mutate the state in callbacks when certain operations happen. As such, it makes sense to use a small class for this with two properties:

1. The current sound being played
2. Whether or not we should consider the menu complete

It's useful to have both pieces of data, as we may cancel the menu half-way through and want to take one set of actions, or we may complete the menu and all the sounds and start a different set of actions.

```
class MenuState(object):
    """A small tracking object for the channel in the menu"""

    def __init__(self, current_sound, complete):
        self.current_sound = current_sound
        self.complete = complete
```

To start, we'll write a function, `play_intro_menu`, that starts the menu on a channel. It will simply initialize the state of the menu, and get the ball rolling on the channel by calling `queue_up_sound`.

```
def play_intro_menu(channel):
    """Play our intro menu to the specified channel
    Since we want to interrupt the playback of the menu when the user presses
    a DTMF key, we maintain the state of the menu via the MenuState object.
    A menu completes in one of two ways:
    (1) The user hits a key
    (2) The menu finishes to completion
    In the case of (2), a timer is started for the channel. If the timer pops,
    a prompt is played back and the menu restarted.
    Keyword Arguments:
    channel  The channel in the IVR
    """
    menu_state = MenuState(0, False)

    def queue_up_sound(channel, menu_state):
        ...

    queue_up_sound(channel, menu_state)
```

queue_up_sound will be responsible for starting the next sound file on the channel and handling the manipulation of that sound file. Since there's a fair amount of checking that goes into this, we'll put the actual act of starting the sound in play_next_sound, which will return the Playback object from ARI. We'll prep the menu_state object for the next sound file playback, and pass it to the PlaybackFinished handler for the current sound being played back to the channel.

```
def queue_up_sound(channel, menu_state):
        """Start up the next sound and handle whatever happens

        Keywords Arguments:
        channel    The channel in the IVR
        menu_state The current state of the menu
        """

        current_playback = play_next_sound(menu_state)

        if not current_playback:
            return
        menu_state.current_sound += 1
        current_playback.on_event('PlaybackFinished', on_playback_finished,
                                  callback_args=[menu_state])
```

play_next_sound will do two things:

1. If we shouldn't play another sound - either because we've run out of sounds to play or because the menu is now "complete", we bail and return None.
2. If we should play back a sound, start it up on the channel and return the Playback object.

```
def play_next_sound(menu_state):
        """Play the next sound, if we should

        Keyword Arguments:
        menu_state The current state of the IVR

        Returns:
        None if no playback should occur
        A playback object if a playback was started
        """
        if (menu_state.current_sound == len(sounds) or menu_state.complete):
            return None
        try:
            current_playback = channel.play(media='sound:%s' %
sounds[menu_state.current_sound])
        except:
            current_playback = None
        return current_playback
```

Our playback finished handler is very simple: since we've already incremented the state of the menu, we just call queue_up_sound again:

```
def on_playback_finished(playback, ev, menu_state):
        """Callback handler for when a playback is finished
        Keyword Arguments:
        playback   The playback object that finished
        ev         The PlaybackFinished event
        menu_state The current state of the menu
        """
        queue_up_sound(channel, menu_state)
```

To recap, our play_intro_menu function has three nested functions:

1. queue_up_sound - starts a sound on a channel, increments the state of the menu, and subscribes for the PlaybackFinished event.
2. play_next_sound - if possible, actually starts the sound. Called from queue_up_sound.

3. `on_playback_finished` - called when `PlaybackFinished` is received for the current playback, and call `queue_up_sound` to start the next sound in the menu.

This will play back the menu sounds, but it doesn't handle cancelling the menu, time-outs, or other conditions. To do that, we're going to need more information from Asterisk.

Cancelling the menu

When the user presses a DTMF key, we want to stop the current playback and end the menu. To do that, we'll need to subscribe for DTMF events from the channel. We'll define a new handler function, `cancel_menu`, and tell `ari-py` to call it when a DTMF key is received via the `ChannelDtmfReceived` event. We don't really care about the digit here - we just want to cancel the menu. In the handler function, we'll set `menu_state.complete` to `True`, then tell the `current_playback` to stop.

We should also stop the menu when the channel is hung up. Since the `cancel_menu` , so we'll subscribe to the `StasisEnd` event here and call `cancel_menu` from it as well:

```
def queue_up_sound(channel, menu_state):
        """Start up the next sound and handle whatever happens

        Keywords Arguments:
        channel    The channel in the IVR
        menu_state The current state of the menu
        """

        current_playback = play_next_sound(menu_state)

        def cancel_menu(channel, ev, current_playback, menu_state):
            """Cancel the menu, as the user did something"""
            menu_state.complete = True
            try:
                current_playback.stop()
            except:
                pass
            return

        if not current_playback:
            return
        menu_state.current_sound += 1
        current_playback.on_event('PlaybackFinished', on_playback_finished,
                                  callback_args=[menu_state])

        # If the user hits a key or hangs up, cancel the menu operations
        channel.on_event('ChannelDtmfReceived', cancel_menu,
                         callback_args=[current_playback, menu_state])
        channel.on_event('StasisEnd', cancel_menu,
                         callback_args=[current_playback, menu_state])
```

Timing out

Now we can cancel the menu, but we also need to restart it if the user doesn't do anything. We can use a Python timer to start a timer if we're finished playing sounds *and* we got to the end of the sound prompt list. We don't want to start the timer if the user pressed a DTMF key - in that case, we would have stopped the menu early and we should be off handling their DTMF key press. The timer will call `menu_timeout`, which will play back a "are you still there?" prompt, then restart the menu.

```
def queue_up_sound(channel, menu_state):
        """Start up the next sound and handle whatever happens
        Keywords Arguments:
        channel    The channel in the IVR
        menu_state The current state of the menu
        """

        def menu_timeout(channel):
            """Callback called by a timer when the menu times out"""
            print 'Channel %s stopped paying attention...' % channel.json.get('name')
            channel.play(media='sound:are-you-still-there')
            play_intro_menu(channel)

        def cancel_menu(channel, ev, current_playback, menu_state):
            """Cancel the menu, as the user did something"""
            menu_state.complete = True
            try:
                current_playback.stop()
            except:
                pass
            return

        current_playback = play_next_sound(menu_state)
        if not current_playback:
            if menu_state.current_sound == len(sounds):
                # Menu played, start a timer!
                timer = threading.Timer(10, menu_timeout, [channel])
                channel_timers[channel.id] = timer
                timer.start()
            return

        menu_state.current_sound += 1
        current_playback.on_event('PlaybackFinished', on_playback_finished,
                                  callback_args=[menu_state])

        # If the user hits a key or hangs up, cancel the menu operations
        channel.on_event('ChannelDtmfReceived', cancel_menu,
                         callback_args=[current_playback, menu_state])
        channel.on_event('StasisEnd', cancel_menu,
                         callback_args=[current_playback, menu_state])
```

Now that we've introduced timers, we know we're going to need to stop them if the user does something. We'll store the timers in a dictionary indexed by channel ID, so we can get them from various parts of the script:

```
channel_timers = {}
```

Handling the DTMF options

While we now have code that plays back the menu to the user, we actually have to implement the attendant menu still. This is slightly easier than playing the menu. We can register for the `ChannelDtmfReceived` event in the `StasisStart` event handler. In that callback, we need to do the following:

1. Cancel any timers associated with the channel. Note that we don't need to stop the playback of the menu, as the menu function `queue_up_sound` already registers a handler for that event and cancels the menu when it gets any digit.
2. Actually handle the digit, if the digit is a `1` or a `2`.
3. If the digit isn't supported, play a prompt informing the user that their option was invalid, and re-play the menu.

The following implements these three items, deferring processing of the valid options to separate functions.

```
def on_dtmf_received(channel, ev):
    """Our main DTMF handler for a channel in the IVR

    Keyword Arguments:
    channel The channel in the IVR
    digit   The DTMF digit that was pressed
    """

    # Since they pressed something, cancel the timeout timer
    cancel_timeout(channel)
    digit = int(ev.get('digit'))

    print 'Channel %s entered %d' % (channel.json.get('name'), digit)
    if digit == 1:
        handle_extension_one(channel)
    elif digit == 2:
        handle_extension_two(channel)
    else:
        print 'Channel %s entered an invalid option!' % channel.json.get('name')
        channel.play(media='sound:option-is-invalid')
        play_intro_menu(channel)


def stasis_start_cb(channel_obj, ev):
    """Handler for StasisStart event"""

    channel = channel_obj.get('channel')
    print "Channel %s has entered the application" % channel.json.get('name')

    channel.on_event('ChannelDtmfReceived', on_dtmf_received)
    play_intro_menu(channel)
```

Cancelling the timer is done in a fashion similar to other examples. If the channel has a Python timer associated with it, we cancel the timer and remove it from the dictionary.

```
def cancel_timeout(channel):
    """Cancel the timeout timer for the channel

    Keyword Arguments:
    channel The channel in the IVR
    """
    timer = channel_timers.get(channel.id)
    if timer:
        timer.cancel()
        del channel_timers[channel.id]
```

Finally, we need to actually do *something* when the user presses a `1` or a `2`. We could do anything here - but in our case, we're merely going to play back the number that they pressed and restart the menu.

```
def handle_extension_one(channel):
    """Handler for a channel pressing '1'

    Keyword Arguments:
    channel The channel in the IVR
    """
    channel.play(media='sound:you-entered')
    channel.play(media='digits:1')
    play_intro_menu(channel)


def handle_extension_two(channel):
    """Handler for a channel pressing '2'

    Keyword Arguments:
    channel The channel in the IVR
    """
    channel.play(media='sound:you-entered')
    channel.play(media='digits:2')
    play_intro_menu(channel)
```

channel-aa.py

The full source for `channel-aa.py` is shown below:

channel-aa.py

```
#!/usr/bin/env python

import ari
import logging
import threading

logging.basicConfig(level=logging.ERROR)

client = ari.connect('http://localhost:8088', 'asterisk', 'asterisk')

# Note: this uses the 'extra' sounds package
sounds = ['press-1', 'or', 'press-2']

channel_timers = {}

class MenuState(object):
    """A small tracking object for the channel in the menu"""

    def __init__(self, current_sound, complete):
        self.current_sound = current_sound
        self.complete = complete


def play_intro_menu(channel):
    """Play our intro menu to the specified channel

    Since we want to interrupt the playback of the menu when the user presses
    a DTMF key, we maintain the state of the menu via the MenuState object.
    A menu completes in one of two ways:
    (1) The user hits a key
    (2) The menu finishes to completion
    In the case of (2), a timer is started for the channel. If the timer pops,
    a prompt is played back and the menu restarted.

    Keyword Arguments:
    channel  The channel in the IVR
    """

    menu_state = MenuState(0, False)

    def play_next_sound(menu_state):
        """Play the next sound, if we should

        Keyword Arguments:
        menu_state The current state of the IVR

        Returns:
        None if no playback should occur
        A playback object if a playback was started
        """
        if (menu_state.current_sound == len(sounds) or menu_state.complete):
            return None
        try:
            current_playback = channel.play(media='sound:%s' %
sounds[menu_state.current_sound])
        except:
            current_playback = None
        return current_playback

    def on_playback_finished(playback, ev, menu_state):
        """Callback handler for when a playback is finished

        Keyword Arguments:
        playback   The playback object that finished
        ev         The PlaybackFinished event
        menu_state The current state of the menu
        """
        queue_up_sound(channel, menu_state)

    def queue_up_sound(channel, menu_state):
        """Start up the next sound and handle whatever happens

        Keywords Arguments:
        channel    The channel in the IVR
        menu_state The current state of the menu
        """

        def menu_timeout(channel):
            """Callback called by a timer when the menu times out"""
            print 'Channel %s stopped paying attention...' % channel.json.get('name')
            channel.play(media='sound:are-you-still-there')
            play_intro_menu(channel)

        def cancel_menu(channel, ev, current_playback, menu_state):
            """Cancel the menu, as the user did something"""
            menu_state.complete = True
            try:
                current_playback.stop()
            except:
                pass
            return

        current_playback = play_next_sound(menu_state)
        if not current_playback:
            if menu_state.current_sound == len(sounds):
                # Menu played, start a timer!
                timer = threading.Timer(10, menu_timeout, [channel])
                channel_timers[channel.id] = timer
                timer.start()
            return

        menu_state.current_sound += 1
        current_playback.on_event('PlaybackFinished', on_playback_finished,
                                  callback_args=[menu_state])

        # If the user hits a key or hangs up, cancel the menu operations
        channel.on_event('ChannelDtmfReceived', cancel_menu,
                         callback_args=[current_playback, menu_state])
        channel.on_event('StasisEnd', cancel_menu,
                         callback_args=[current_playback, menu_state])

    queue_up_sound(channel, menu_state)


def handle_extension_one(channel):
    """Handler for a channel pressing '1'

    Keyword Arguments:
    channel The channel in the IVR
    """
    channel.play(media='sound:you-entered')
    channel.play(media='digits:1')
    play_intro_menu(channel)


def handle_extension_two(channel):
    """Handler for a channel pressing '2'

    Keyword Arguments:
    channel The channel in the IVR
    """
    channel.play(media='sound:you-entered')
    channel.play(media='digits:2')
    play_intro_menu(channel)


def cancel_timeout(channel):
    """Cancel the timeout timer for the channel

    Keyword Arguments:
    channel The channel in the IVR
    """
    timer = channel_timers.get(channel.id)
    if timer:
        timer.cancel()
        del channel_timers[channel.id]
```

```
def on_dtmf_received(channel, ev):
    """Our main DTMF handler for a channel in the IVR

    Keyword Arguments:
    channel The channel in the IVR
    digit   The DTMF digit that was pressed
    """

    # Since they pressed something, cancel the timeout timer
    cancel_timeout(channel)
    digit = int(ev.get('digit'))

    print 'Channel %s entered %d' % (channel.json.get('name'), digit)
    if digit == 1:
        handle_extension_one(channel)
    elif digit == 2:
        handle_extension_two(channel)
    else:
        print 'Channel %s entered an invalid option!' % channel.json.get('name')
        channel.play(media='sound:option-is-invalid')
        play_intro_menu(channel)


def stasis_start_cb(channel_obj, ev):
    """Handler for StasisStart event"""

    channel = channel_obj.get('channel')
    print "Channel %s has entered the application" % channel.json.get('name')

    channel.on_event('ChannelDtmfReceived', on_dtmf_received)
    play_intro_menu(channel)


def stasis_end_cb(channel, ev):
    """Handler for StasisEnd event"""

    print "%s has left the application" % channel.json.get('name')
    cancel_timeout(channel)


client.on_channel_event('StasisStart', stasis_start_cb)
client.on_channel_event('StasisEnd', stasis_end_cb)
client.run(apps='channel-aa')
```

channel-aa.py in action

The following shows the output of `channel-aa.py` when a PJSIP channel presses `1`, `2`, `8`, then times out. Finally they hang up.

```
Channel PJSIP/alice-00000001 has entered the application
Channel PJSIP/alice-00000001 entered 1
Channel PJSIP/alice-00000001 entered 2
Channel PJSIP/alice-00000001 entered 8
Channel PJSIP/alice-00000001 entered an invalid option!
Channel PJSIP/alice-00000001 stopped paying attention...
PJSIP/alice-00000001 has left the application
```

JavaScript (Node.js)

As this example is a bit larger, how the code is written and structured is broken up into two phases:

1. Constructing the menu and handling its state as the user presses buttons.
2. Actually handling the button presses from the user.

The full source code for this example immediately follows the walk through.

Playing the menu

Unlike Playback, which can chain multiple sounds together and play them back in one continuous operation, ARI treats all sound files being played as separate operations. It will queue each sound file up to be played on the channel, and hand back the caller an object to control the operation of that single sound file. The menu announcement for the attendant has the following requirements:

1. Playback the options for the user
2. If the user presses a DTMF key, cancel the playback of the options and handle the request
3. If the user presses an invalid DTMF key, let them know and restart the menu
4. If the user doesn't press anything, wait 10 seconds, ask them if they are still present, and restart the menu

The second requirement makes this a bit more challenging: when the user presses a DTMF key, we want to cancel whatever sound file is currently being played back and immediately handle their request. We thus have to maintain some state in our application about what sound file is currently being played so that we can cancel the correct playback. We also don't want to queue up all of the sounds immediately - we'd have to walk through all of the queued up sounds and cancel each one - that'd be annoying! Instead, we only want to start the next sound in our prompt when the previous has completed.

To start, we'll define an object to represent the menu at the top of our script that defines sounds that make up the initial menu prompt as well as valid DTMF options for the menu:

```
var menu = {
  // valid menu options
  options: [1, 2],
  // note: this uses the 'extra' sounds package
  sounds: ['sound:press-1', 'sound:or', 'sound:press-2']
};
```

To start with, well register a callback to handle a StasisStart and StasisEnd event on any channel that enters into our application:

```
function stasisStart(event, channel) {
  console.log('Channel %s has entered the application', channel.name);

  channel.on('ChannelDtmfReceived', dtmfReceived);

  channel.answer(function(err) {
    if (err) {
      throw err;
    }
    playIntroMenu(channel);
  });
}

// Handler for StasisEnd event
function stasisEnd(event, channel) {
  console.log('Channel %s has left the application', channel.name);

  // clean up listeners
  channel.removeListener('ChannelDtmfReceived', dtmfReceived);
  cancelTimeout(channel);
}
```

Note that we register a callback to handle ChannelDtmfReceived events on a channel entering our application in StasisStart and then unregister that callback on StasisEnd. For long running, non-trivial applications, this allows the JavaScript garbage collector to clean up our callback. This is important since every channel entering into our application will register its own copy of the callback which is not be garbage collected until it is unregistered.

We'll cover the DTMF callback handler shortly, but first we'll cover writting functions to handle playing the menu prompt

First we'll write a function to initialize a new instance of our menu; playIntroMenu.

Since we'll want to maintain some state, we'll create a small object to do that for us. This object will keep track of the following:

1. The current sound being played
2. The current Playback object being played
3. Whether or not this menu instance is done

It's useful to have this data, as we may cancel the menu half-way through and want to take one set of actions, or we may play all the sounds that make up the menu prompt and start a different set of actions.

```
var state = {
  currentSound: menu.sounds[0],
  currentPlayback: undefined,
  done: false
};
```

`playIntroMenu will` start the menu on a channel. It will simply initialize the state of the menu, and get the ball rolling on the channel by calling `queueUpSound` which is a nested function within playIntroMenu.

```
function playIntroMenu(channel) {
  var state = {
    currentSound: menu.sounds[0],
    currentPlayback: undefined,
    done: false
  };

  channel.on('ChannelDtmfReceived', cancelMenu);
  channel.on('StasisEnd', cancelMenu);
  queueUpSound();
  ...
```

We'll cover cancelMenu shortly, but first let's discuss queueUpSound. `queueUpSound` will be responsible for starting the next sound file on the channel and handling the manipulation of that sound file. queueUpSound is also responsible for starting a timeout once all sounds for the menu prompt have completed to handle reminding the user that they must choose a menu option. We'll cover that part shortly but first, we'll cover handling progerssing through the sounds that make up the menu prompt. We first initiate playback on the current sound in the sequence. We then register a callback to handle that playback finishing, which will trigger queueUpSound to be called again, moving on to the next sound in the sequence. Finally, we update the state object to reflect the next sound to be played in the menu prompt sequence.

```
function queueUpSound() {
  if (!state.done) {
    // have we played all sounds in the menu?
    if (!state.currentSound) {
      var timer = setTimeout(stillThere, 10 * 1000);
      timers[channel.id] = timer;
    } else {
      var playback = client.Playback();
      state.currentPlayback = playback;

      channel.play({media: state.currentSound}, playback, function(err) {
        // ignore errors
      });
      playback.once('PlaybackFinished', function(event, playback) {
        queueUpSound();
      });

      var nextSoundIndex = menu.sounds.indexOf(state.currentSound) + 1;
      state.currentSound = menu.sounds[nextSoundIndex];
    }
  }
}
```

Notice that when registering our PlaybackFinished callback handler, we use the once method on the resource instance instead of on. This ensures that the callback will be invoked once and then automatically be unregistered. Since a PlaybackFinished event will only be invoked once for a given Playback instance, it makes sense to use this method which will also enable the callback to be garbage collected once it has been invoked.

queueUpSound will play back the menu sounds, but it doesn't handle cancelling the menu, time-outs, or other conditions. To do that, we're going to need more information from Asterisk.

Cancelling the menu

When the user presses a DTMF key, we want to stop the current playback and end the menu. To do that, we'll need to subscribe for DTMF events from the channel. We'll define a new handler function, `cancelMenu`, and tell `ari-client` to call it when a DTMF key is received via the `ChannelDtmfReceived` event. We don't really care about the digit here - we just want to cancel the menu. In the handler function, we'll set `state.done` to `true`, then tell the `currentPlayback` to stop.

We should also stop the menu when the channel is hung up. To do this we'll subscribe to the `StasisEnd` event as well and register cancelMenu as its callback handler:

```
function cancelMenu() {
  state.done = true;
  if (state.currentPlayback) {
    state.currentPlayback.stop(function(err) {
      // ignore errors
    });
  }

  // remove listeners as future calls to playIntroMenu will create new ones
  channel.removeListener('ChannelDtmfReceived', cancelMenu);
  channel.removeListener('StasisEnd', cancelMenu);
}
```

Note that once the cancelMenu callback is invoked, we unregister both the ChannelDtmfReceived and StasisEnd events. This is performed so that once this particular menu instance stops, we do not leave registered callbacks behind that will never be garbage collected.

Timing out

Now we can cancel the menu, but we also need to restart it if the user doesn't do anything. We can use a JavaScript timeout to start a timer if we're finished playing sounds *and* we got to the end of the sound prompt sequence. We don't want to start the timer if the user pressed a DTMF key - in that case, we would have stopped the menu early and we should be off handling their DTMF key press. The timer will call `stillThere`, which will play back a "are you still there?" prompt, then restart the menu.

```
function stillThere() {
  console.log('Channel %s stopped paying attention...', channel.name);

  channel.play({media: 'sound:are-you-still-there'}, function(err) {
    if (err) {
      throw err;
    }

    playIntroMenu(channel);
  });
}
```

Now that we've introduced timers, we know we're going to need to stop them if the user does something. We'll store the timers in an object indexed by channel ID, so we can get them from various parts of the script:

```
var timers = {};
```

Handling the DTMF options

While we now have code that plays back the menu to the user, we actually have to implement the attendant menu still. Earlier in our example we registered a callback handler for a ChannelDtmfReceived event on a channel that enters into our application. In that callback, we need to do the following:

1. Cancel any timers associated with the channel. Note that we don't need to stop the playback of the menu, as the menu function `queueUpSound` already registers a handler for that event and cancels the menu when it gets any digit.
2. Actually handle the digit, if the digit is a `1` or a `2`.
3. If the digit isn't supported, play a prompt informing the user that their option was invalid, and re-play the menu.

The following implements these three items, deferring processing of the valid options to a separate function.

```
function dtmfReceived(event, channel) {
  cancelTimeout(channel);
  var digit = parseInt(event.digit);

  console.log('Channel %s entered %d', channel.name, digit);

  // will be non-zero if valid
  var valid = ~menu.options.indexOf(digit);
  if (valid) {
    handleDtmf(channel, digit);
  } else {
    console.log('Channel %s entered an invalid option!', channel.name);

    channel.play({media: 'sound:option-is-invalid'}, function(err, playback) {
      if (err) {
        throw err;
      }

      playIntroMenu(channel);
    });
  }
}
```

Cancelling the timer is done in a fashion similar to other examples. If the channel has a JavaScript timeout associated with it, we cancel the timer and remove it from the object.

```
function cancelTimeout(channel) {
  var timer = timers[channel.id];

  if (timer) {
    clearTimeout(timer);
    delete timers[channel.id];
  }
}
```

Finally, we need to actually do *something* when the user presses a `1` or a `2`. We could do anything here - but in our case, we're merely going to play back the number that they pressed and restart the menu.

```
function handleDtmf(channel, digit) {
  var parts = ['sound:you-entered', util.format('digits:%s', digit)];
  var done = 0;

  var playback = client.Playback();
  channel.play({media: 'sound:you-entered'}, playback, function(err) {
    // ignore errors
    channel.play({media: util.format('digits:%s', digit)}, function(err) {
      // ignore errors
      playIntroMenu(channel);
    });
  });
}
```

channel-aa.js

The full source for `channel-aa.js` is shown below:

channel-aa.js

```
/*jshint node:true*/
'use strict';

var ari = require('ari-client');
var util = require('util');

ari.connect('http://localhost:8088', 'asterisk', 'asterisk', clientLoaded);

var menu = {
  // valid menu options
  options: [1, 2],
  // note: this uses the 'extra' sounds package
  sounds: ['sound:press-1', 'sound:or', 'sound:press-2']
};

var timers = {};

// Handler for client being loaded
function clientLoaded (err, client) {
  if (err) {
    throw err;
  }

  client.on('StasisStart', stasisStart);
  client.on('StasisEnd', stasisEnd);
```

```
// Handler for StasisStart event
function stasisStart(event, channel) {
  console.log('Channel %s has entered the application', channel.name);

  channel.on('ChannelDtmfReceived', dtmfReceived);

  channel.answer(function(err) {
    if (err) {
      throw err;
    }

    playIntroMenu(channel);
  });
}

// Handler for StasisEnd event
function stasisEnd(event, channel) {
  console.log('Channel %s has left the application', channel.name);

  // clean up listeners
  channel.removeListener('ChannelDtmfReceived', dtmfReceived);
  cancelTimeout(channel);
}

// Main DTMF handler
function dtmfReceived(event, channel) {
  cancelTimeout(channel);
  var digit = parseInt(event.digit);

  console.log('Channel %s entered %d', channel.name, digit);

  // will be non-zero if valid
  var valid = ~menu.options.indexOf(digit);
  if (valid) {
    handleDtmf(channel, digit);
  } else {
    console.log('Channel %s entered an invalid option!', channel.name);

    channel.play({media: 'sound:option-is-invalid'}, function(err, playback) {
      if (err) {
        throw err;
      }

      playIntroMenu(channel);
    });
  }
}

/**
 * Play our intro menu to the specified channel
 *
 * Since we want to interrupt the playback of the menu when the user presses
 * a DTMF key, we maintain the state of the menu via the MenuState object.
 * A menu completes in one of two ways:
 * (1) The user hits a key
 * (2) The menu finishes to completion
 *
 * In the case of (2), a timer is started for the channel. If the timer pops,
 * a prompt is played back and the menu restarted.
 **/
function playIntroMenu(channel) {
  var state = {
    currentSound: menu.sounds[0],
    currentPlayback: undefined,
    done: false
  };

  channel.on('ChannelDtmfReceived', cancelMenu);
  channel.on('StasisEnd', cancelMenu);
  queueUpSound();

  // Cancel the menu, as the user did something
  function cancelMenu() {
    state.done = true;
    if (state.currentPlayback) {
      state.currentPlayback.stop(function(err) {
        // ignore errors
      });
    }

    // remove listeners as future calls to playIntroMenu will create new ones
    channel.removeListener('ChannelDtmfReceived', cancelMenu);
    channel.removeListener('StasisEnd', cancelMenu);
  }

  // Start up the next sound and handle whatever happens
  function queueUpSound() {
    if (!state.done) {
      // have we played all sounds in the menu?
      if (!state.currentSound) {
        var timer = setTimeout(stillThere, 10 * 1000);
        timers[channel.id] = timer;
      } else {
        var playback = client.Playback();
        state.currentPlayback = playback;

        channel.play({media: state.currentSound}, playback, function(err) {
          // ignore errors
        });
        playback.once('PlaybackFinished', function(event, playback) {
          queueUpSound();
        });

        var nextSoundIndex = menu.sounds.indexOf(state.currentSound) + 1;
        state.currentSound = menu.sounds[nextSoundIndex];
      }
    }
  }

  // plays are-you-still-there and restarts the menu
  function stillThere() {
    console.log('Channel %s stopped paying attention...', channel.name);

    channel.play({media: 'sound:are-you-still-there'}, function(err) {
      if (err) {
        throw err;
      }
      playIntroMenu(channel);
    });
  }
}

// Cancel the timeout for the channel
function cancelTimeout(channel) {
  var timer = timers[channel.id];

  if (timer) {
    clearTimeout(timer);
    delete timers[channel.id];
  }
}

// Handler for channel pressing valid option
function handleDtmf(channel, digit) {
  var parts = ['sound:you-entered', util.format('digits:%s', digit)];
  var done = 0;

  var playback = client.Playback();
  channel.play({media: 'sound:you-entered'}, playback, function(err) {
    // ignore errors
    channel.play({media: util.format('digits:%s', digit)}, function(err) {
      // ignore errors
      playIntroMenu(channel);
    });
  });
}
```

```
  client.start('channel-aa');
}
```

channel-aa.js in action

The following shows the output of `channel-aa.js` when a PJSIP channel presses `1`, `2`, `8`, then times out. Finally they hang up.

```
Channel PJSIP/alice-00000001 has entered the application
Channel PJSIP/alice-00000001 entered 1
Channel PJSIP/alice-00000001 entered 2
Channel PJSIP/alice-00000001 entered 8
Channel PJSIP/alice-00000001 entered an invalid option!
Channel PJSIP/alice-00000001 stopped paying attention...
PJSIP/alice-00000001 has left the application
```

Introduction to ARI and Bridges

Asterisk Bridges

In Asterisk, bridges can be thought of as a container for channels that form paths of communication between the channels contained within them. They can be used to pass media back and forth between the channels, as well as to play media to the various channels in a variety of ways.

More Information
For more information on bridges in Asterisk, see Bridges.

Bridges in a Stasis Application

Bridge Types

When a bridge is created through ARI, there are a number of attributes that can be specified that determine how the bridge mixes media between its participants. These include:

- `mixing` - specify that media should be passed between all channels in the bridge. This attribute cannot be used with `holding`.
- `dtmf_events` - specify that media should be decoded within Asterisk so that DTMF can be recognized. If this is not specified, then DTMF events may not be raised due to the media being passed directly between the channels in the bridge. This attribute only impacts how media is mixed when the `mixing` attribute is used.
- `proxy_media` - specify that media should always go through Asterisk, even if it could be redirected between clients. This attribute only impacts how media is mixed when the `mixing` attribute is used.
- `holding` - specify that the channels in the bridge should be entertained with some media. Channels in the bridge have two possible roles: a participant or an announcer. Media between participant channels is not shared; media from an announcer channel is played to all participant channels.

Depending on the combination of attributes selected when a bridge is created, different mixing technologies may be used for the participants in the bridge. Asterisk will attempt to use the most performant mixing technology that it can based on the channel types in the bridge, subject to the attributes specified when the bridge was created.

Subscription Model

Unlike channels, bridges in a Stasis application are not automatically subscribed to for events. In order to receive events concerning events for a given bridge, the applications resource must be used to subscribe to the bridge via the `POST - /applications/{app_name}/subscription` operation . Events related to channels entering and leaving bridges will be sent without the need to subscribe to them since they are related to a channel in a Stasis application.

Example: Interacting with Bridges

For this example, we're going to write an ARI application that will do the following:

1. When it connects, it will print out the names of all existing holding bridges. If there are no existing holding bridges, it will create one.
2. When a channel enters into its Stasis application, it will be added to the holding bridge and music on hold will be started on the bridge.

Dialplan

The dialplan for this will be very straight forward: a simple extension that drops a channel into Stasis.

extensions.conf

```
[default]

exten => 1000,1,NoOp()
 same =>      n,Answer()
 same =>      n,Stasis(bridge-hold)
 same =>      n,Hangup()
```

Python

For our Python examples, we will rely primarily on the ari-py library. Because the `ari` library will emit useful information using Python logging, we should go ahead and set that up as well - for now, a `basicConfig` with `ERROR` messages displayed should be sufficient. Finally, we'll need to get a client made by initiating a connection to Asterisk. This occurs using the `ari.connect` method, where we have to specify three things:

1. The HTTP base URI of the Asterisk server to connect to. Here, we assume that this is running on the same machine as the script, and that we're using the default port for Asterisk's HTTP server - `8088`.
2. The username of the ARI user account to connect as. In this case, we're specifying it as `asterisk`.
3. The password for the ARI user account. In this case, that's asterisk.

> Modify the connection credentials as appropriate for your server, although many examples will use these credentials.
>
> **Please don't use these credentials in production systems!**

```
#!/usr/bin/env python

import ari
import logging

logging.basicConfig(level=logging.ERROR)

client = ari.connect('http://localhost:8088', 'asterisk', 'asterisk')
```

Once we've made our connection, our first task is to look for an existing holding bridge - if there is no existing holding bridge - we need to create it. The bridges resource has an operation for listing existing bridges - `GET /bridges`. Using ari-py we need to use the operation nickname - `list`. We can then use another bridges resource operation to create a holding bridge if none was found - `POST /bridges`. Using ari-py, we need to use the operation nickname - `create`.

```
# find or create a holding bridge
bridges = [candidate for candidate in client.bridges.list() if
           candidate.json['bridge_type'] == 'holding']
if bridges:
    bridge = bridges[0]
    print "Using bridge %s" % bridge.id
else:
    bridge = client.bridges.create(type='holding')
    print "Created bridge %s" % bridge.id
```

The `GET /channels` operation returns back a list of `Bridge` resources. Those resources, however, are returned as JSON from the operation, and while the `ari-py` library converts the `uniqueid` of those into an attribute on the object, it leaves the rest of them in the JSON dictionary.

Our next step involves adding channels that enter our Stasis application to the bridge we either found or created and signaling when a channel leaves our Stasis application. To do that, we need to subscribe for the `StasisStart` and `StasisEnd` events:

```
client.on_channel_event('StasisStart', stasis_start_cb)
client.on_channel_event('StasisEnd', stasis_end_cb)
```

We need two handler functions - `stasis_start_cb` for the `StasisStart` event and `stasis_end_cb` for the `StasisEnd` event:

```
def stasis_start_cb(channel_obj, ev):
    """Handler for StasisStart event"""

    channel = channel_obj.get('channel')
    print "Channel %s just entered our application, adding it to bridge %s" % (
        channel.json.get('name'), bridge.id)
    channel.answer()
    bridge.addChannel(channel=channel.id)
    bridge.startMoh()

def stasis_end_cb(channel, ev):
    """Handler for StasisEnd event"""

    print "Channel %s just left our application" % channel.json.get('name')
```

Finally, we need to tell the `client` to run our application. Once we call `client.run`, the websocket connection will be made and our application will wait on events infinitely. We can use `Ctrl+C` to kill it and break the connection.

```
client.run(apps='bridge-hold')
```

bridge-hold.py

The full source code for `bridge-hold.py` is shown below:

```
#!/usr/bin/env python

import ari
import logging

logging.basicConfig(level=logging.ERROR)

client = ari.connect('http://localhost:8088', 'asterisk', 'asterisk')

# find or create a holding bridge
bridges = [candidate for candidate in client.bridges.list() if
           candidate.json['bridge_type'] == 'holding']
if bridges:
    bridge = bridges[0]
    print "Using bridge %s" % bridge.id
else:
    bridge = client.bridges.create(type='holding')
    print "Created bridge %s" % bridge.id

def stasis_start_cb(channel_obj, ev):
    """Handler for StasisStart event"""

    channel = channel_obj.get('channel')
    print "Channel %s just entered our application, adding it to bridge %s" % (
        channel.json.get('name'), bridge.id)

    channel.answer()
    bridge.addChannel(channel=channel.id)
    bridge.startMoh()

def stasis_end_cb(channel, ev):
    """Handler for StasisEnd event"""

    print "Channel %s just left our application" % channel.json.get('name')

client.on_channel_event('StasisStart', stasis_start_cb)
client.on_channel_event('StasisEnd', stasis_end_cb)

client.run(apps='bridge-hold')
```

bridge-hold.py in action

Here, we see the output from the `bridge-hold.py` script when a PJSIP channel for endpoint 'alice' enters into the application:

```
asterisk:~$ python bridge-hold.py
Created bridge 79f3ad78-b124-4d7b-a629-a53b7e7f50cd
Channel PJSIP/alice-00000001 just entered our application, adding it to bridge
79f3ad78-b124-4d7b-a629-a53b7e7f50cd
Channel PJSIP/alice-00000001 just left our application
```

JavaScript (Node.js)

For our JavaScript examples, we will rely primarily on the Node.js ari-client library. We'll need to get a client made by initiating a connection to Asterisk. This occurs using the `ari.connect` method, where we have to specify four things:

1. The HTTP base URI of the Asterisk server to connect to. Here, we assume that this is running on the same machine as the script,

and that we're using the default port for Asterisk's HTTP server - `8088`.

2. The username of the ARI user account to connect as. In this case, we're specifying it as `asterisk`.
3. The password for the ARI user account. In this case, that's asterisk.
4. A callback that will be called with an error if one occurred, followed by an instance of an ARI client.

> Modify the connection credentials as appropriate for your server, although many examples will use these credentials.
>
> **Please don't use these credentials in production systems!**

```
/*jshint node:true*/
'use strict';

var ari = require('ari-client');
var util = require('util');

ari.connect('http://localhost:8088', 'asterisk', 'asterisk', clientLoaded);

// handler for client being loaded
function clientLoaded (err, client) {
  if (err) {
    throw err;
  }
}
```

Once we've made our connection, our first task is to look for an existing holding bridge - if there is no existing holding bridge - we need to create it. The bridges resource has an operation for listing existing bridges - `GET /bridges`. Using ari-client we need to use the operation nickname - `list`. We can then use another bridges resource operation to create a holding bridge if none was found - `POST /bridges`. Using ari-client, we need to use the operation nickname - `create`.

```
// find or create a holding bridge
var bridge = null;
client.bridges.list(function(err, bridges) {
  if (err) {
    throw err;
  }
  bridge = bridges.filter(function(candidate) {
    return candidate.bridge_type === 'holding';
  })[0];
  if (bridge) {
    console.log(util.format('Using bridge %s', bridge.id));
  } else {
    client.bridges.create({type: 'holding'}, function(err, newBridge) {
      if (err) {
        throw err;
      }
      bridge = newBridge;
      console.log(util.format('Created bridge %s', bridge.id));
    });
  }
});
```

The `GET /channels` operation returns back a an error if it occurred and a list of `Bridge` resources. `ari-client` will return a JavaScript object for each `Bridge` resource. Properties such as `bridge_type` can be accessed on the object directly.

Our next step involves adding channels that enter our Stasis application to the bridge we either found or created and signaling when a channel leaves our Stasis application. To do that, we need to subscribe for the `StasisStart` and `StasisEnd` events:

```
client.on('StasisStart', stasisStart);
client.on('StasisEnd', stasisEnd);
```

We need two callback functions - `stasisStart` for the `StasisStart` event and `stasisEnd` for the `StasisEnd` event:

```
// handler for StasisStart event
function stasisStart(event, channel) {
  console.log(util.format(
      'Channel %s just entered our application, adding it to bridge %s',
      channel.name,
      bridge.id));

  channel.answer(function(err) {
    if (err) {
      throw err;
    }

    bridge.addChannel({channel: channel.id}, function(err) {
      if (err) {
        throw err;
      }

      bridge.startMoh(function(err) {
        if (err) {
          throw err;
        }
      });
    });
  });
}

// handler for StasisEnd event
function stasisEnd(event, channel) {
  console.log(util.format(
      'Channel %s just left our application', channel.name));
}
```

Finally, we need to tell the `client` to start our application. Once we call `client.start`, a websocket connection will be established and the client will emit Node.js events as events come in through the websocket. We can use `Ctrl+C` to kill it and break the connection.

```
client.start('bridge-hold');
```

bridge-hold.js

The full source code for `bridge-hold.js` is shown below:

```
/*jshint node:true*/
'use strict';

var ari = require('ari-client');
var util = require('util');

ari.connect('http://localhost:8088', 'asterisk', 'asterisk', clientLoaded);

// handler for client being loaded
function clientLoaded (err, client) {
  if (err) {
    throw err;
  }

  // find or create a holding bridge
  var bridge = null;
  client.bridges.list(function(err, bridges) {
    if (err) {
```

```
    throw err;
  }

  bridge = bridges.filter(function(candidate) {
    return candidate.bridge_type === 'holding';
  })[0];

  if (bridge) {
    console.log(util.format('Using bridge %s', bridge.id));
  } else {
    client.bridges.create({type: 'holding'}, function(err, newBridge) {
      if (err) {
        throw err;
      }

      bridge = newBridge;
      console.log(util.format('Created bridge %s', bridge.id));
    });
  }
});

// handler for StasisStart event
function stasisStart(event, channel) {
  console.log(util.format(
      'Channel %s just entered our application, adding it to bridge %s',
      channel.name,
      bridge.id));

  channel.answer(function(err) {
    if (err) {
      throw err;
    }

    bridge.addChannel({channel: channel.id}, function(err) {
      if (err) {
        throw err;
      }

      bridge.startMoh(function(err) {
        if (err) {
          throw err;
        }
      });
    });
  });
}

// handler for StasisEnd event
function stasisEnd(event, channel) {
  console.log(util.format(
      'Channel %s just left our application', channel.name));
}

client.on('StasisStart', stasisStart);
client.on('StasisEnd', stasisEnd);
```

```
  client.start('bridge-hold');
}
```

bridge-hold.js in action

Here, we see the output from the `bridge-hold.js` script when a PJSIP channel for endpoint 'alice' enters into the application:

```
asterisk:~$ node bridge-hold.js
Created bridge 41208316-174c-4e40-90bb-c45cca6579d4
Channel PJSIP/alice-00000001 just entered our application, adding it to bridge
41208316-174c-4e40-90bb-c45cca6579d4
Channel PJSIP/alice-00000001 just left our application
```

ARI and Bridges: Holding Bridges

Holding Bridges

Holding bridges are a special type of bridge in Asterisk. The purpose of a holding bridge is to provide a consistent way to place channels when you want the person on the other end of the channel to wait. Asterisk will mix the media to the channel depending on the type of **role** the channel has within the bridge. Two types of roles are supported:

- `participant` - the default role for channels in a holding bridge. Media from the bridge is played directly to the channels; however, media from the channels is not played to any other participant.
- `announcer` - if a channel joins a holding bridge as an announcer, the bridge will not play media to the channel. However, all media from the channel will be played to all `participant` channels in the bridge simultaneously.

Adding a channel as a participant

To add a channel as a participant to a holding bridge, you can either not specify a `role` (as the `participant` role is the default role for holding bridges), or you can specify the `participant` role directly:

```
POST /bridges/{bridge_id}/addChannel?channel=12345
POST /bridges/{bridge_id}/addChannel?channel=12345&role=participant
```

Adding a channel as an announcer

To add a channel as an announcer to a holding bridge, you must specify a role of `announcer`:

```
POST /bridges/{bridge_id}/addChannel?channel=56789&role=announcer
```

When is an Announcer channel useful?
If you want to simply play back a media file to all participants in a holding bridge, e.g., "your call is important to us, please keep waiting", you can simply initiate a `/play` operation on the holding bridge itself. That will perform a playback to all participants in the same fashion as an announcer channel.

An announcer channel is particularly useful when there is someone actually on the other end of the channel, as opposed to a pre-recorded message. For example, you may have a call queue supervisor who wants to let everyone who is waiting for an agent that response times are especially long, but to hold on for a bit longer. Jumping into the holding bridge as an announcer adds a small bit of humanity to the dreaded call queue experience!

Music on hold, media playback, recording, and other such things

When dealing with holding bridges, given the particular media rules and channel roles involves, there are some additional catches that you have to be aware when manipulating the bridge:

1. Playing music on hold to the bridge will play it for all participants, as well playing media to the bridge. However, you can only do **one** of those operations - you cannot play media to a holding bridge while you are simultaneously playing music on hold to the bridge. Initiating a `/play` operation on a holding bridge should only be done after stopping the music on hold; likewise, starting music on hold on a bridge with a `/play` operation currently in progress will fail.
2. Recording a holding bridge - while possible - is not terribly interesting. Participant media is dropped - so at best, you'll only record the entertainment that was played to the participants.

There can be only one!

You cannot have an announcer channel in a holding bridge at the same time that you perform a `play` operation or have music on hold playing to the bridge. Holding bridges do **not** mix the media between announcers. Since media from the `play` operation has to go to all participants, as does your announcer channel's media, the holding bridge will become quite confused about your application's intent.

Example: Infinite wait area

Now that we all know that holding bridges are perfect for building what many callers fear - the dreaded waiting area of doom - let's make one! This example ARI application will do the following:

1. When a channel enters into the Stasis application, it will be put into a existing holding bridge or a newly created one if none exist.
2. Music on hold will be played on the bridge.
3. Periodically, the `thnk-u-for-patience` sound will be played to the bridge thanking the users for their patience, which they will need since this holding bridge will never progress beyond this point!
4. When a channel leaves a holding bridge, if no other channels remain, the bridge will be destroyed.

This example will use a similar structure to the bridge-hold python example. Unlike that example, however, it will use some form of a timer to perform our periodic announcement to the holding bridge, and when all the channels have left the infinite wait area, we'll destroy the holding bridge (cleaning up resources is always good!)

Dialplan

For this example, we need to just drop the channel into Stasis, specifying our application:

extensions.conf

```
exten => 1000,1,NoOp()
 same =>      n,Stasis(bridge-infinite-wait)
 same =>      n,Hangup()
```

Python

When a channel enters our Stasis application, we first look for an existing holding bridge or create one if none is found. When we create a new bridge, we start music on hold in the bridge and create a timer that will call a callback after 30 seconds. That callback temporarily stops the music on hold, and starts a play operation on the bridge that thanks everyone for their patience. When the play operation finishes, it resumes music on hold.

```
# find or create a holding bridge
holding_bridge = None

# Announcer timer
announcer_timer = None

def find_or_create_bridge():
    """Find our infinite wait bridge, or create a new one

    Returns:
    The one and only holding bridge
    """

    global holding_bridge
    global announcer_timer

    if holding_bridge:
        return holding_bridge

    bridges = [candidate for candidate in client.bridges.list() if
               candidate.json['bridge_type'] == 'holding']
    if bridges:
        bridge = bridges[0]
        print "Using bridge %s" % bridge.id
    else:
        bridge = client.bridges.create(type='holding')
        bridge.startMoh()
        print "Created bridge %s" % bridge.id

    def play_announcement(bridge):
        """Play an announcement to the bridge"""

        def on_playback_finished(playback, ev):
            """Handler for the announcement's PlaybackFinished event"""
            global announcer_timer
            global holding_bridge

            holding_bridge.startMoh()

            announcer_timer = threading.Timer(30, play_announcement,
                                              [holding_bridge])
            announcer_timer.start()

        bridge.stopMoh()
        print "Letting the everyone know we care..."
        thanks_playback = bridge.play(media='sound:thnk-u-for-patience')
        thanks_playback.on_event('PlaybackFinished', on_playback_finished)

    holding_bridge = bridge
    holding_bridge.on_event('ChannelLeftBridge', on_channel_left_bridge)
    # After 30 seconds, let everyone in the bridge know that we care
    announcer_timer = threading.Timer(30, play_announcement, [holding_bridge])
    announcer_timer.start()
    return bridge
```

The function that does this work, `find_or_create_bridge`, is called from our `StasisStart` event handler. The bridge that it returns will have the new channel added to it.

```
def stasis_start_cb(channel_obj, ev):
    """Handler for StasisStart event"""

    bridge = find_or_create_bridge()

    channel = channel_obj.get('channel')
    print "Channel %s just entered our application, adding it to bridge %s" % (
        channel.json.get('name'), holding_bridge.id)

    channel.answer()
    bridge.addChannel(channel=channel.id)
```

In the `find_or_create_bridge` function, we also subscribed for the `ChannelLeftBridge` event. We'll add a callback handler for this in that function as well. When the channel leaves the bridge, we'll check to see if there are no more channels in the bridge and - if so - destroy the bridge.

```
def on_channel_left_bridge(bridge, ev):
        """Handler for ChannelLeftBridge event"""
        global holding_bridge
        global announcer_timer

        channel = ev.get('channel')
        channel_count = len(bridge.json.get('channels'))

        print "Channel %s left bridge %s" % (channel.get('name'), bridge.id)
        if holding_bridge.id == bridge.id and channel_count == 0:
            if announcer_timer:
                announcer_timer.cancel()
                announcer_timer = None

            print "Destroying bridge %s" % bridge.id
            holding_bridge.destroy()
            holding_bridge = None
```

bridge-infinite-wait.py

The full source code for `bridge-infinite-wait.py` is shown below:

bridge-infinite-wait.py

```
#!/usr/bin/env python

import ari
import logging
import threading

logging.basicConfig(level=logging.ERROR)

client = ari.connect('http://localhost:8088', 'asterisk', 'asterisk')

# find or create a holding bridge
holding_bridge = None

# Announcer timer
announcer_timer = None

def find_or_create_bridge():
```

```
"""Find our infinite wait bridge, or create a new one

Returns:
The one and only holding bridge
"""

global holding_bridge
global announcer_timer

if holding_bridge:
    return holding_bridge

bridges = [candidate for candidate in client.bridges.list() if
           candidate.json['bridge_type'] == 'holding']
if bridges:
    bridge = bridges[0]
    print "Using bridge %s" % bridge.id
else:
    bridge = client.bridges.create(type='holding')
    bridge.startMoh()
    print "Created bridge %s" % bridge.id

def play_announcement(bridge):
    """Play an announcement to the bridge"""

    def on_playback_finished(playback, ev):
        """Handler for the announcement's PlaybackFinished event"""
        global announcer_timer
        global holding_bridge

        holding_bridge.startMoh()

        announcer_timer = threading.Timer(30, play_announcement,
                                          [holding_bridge])
        announcer_timer.start()

    bridge.stopMoh()
    print "Letting the everyone know we care..."
    thanks_playback = bridge.play(media='sound:thnk-u-for-patience')
    thanks_playback.on_event('PlaybackFinished', on_playback_finished)

def on_channel_left_bridge(bridge, ev):
    """Handler for ChannelLeftBridge event"""
    global holding_bridge
    global announcer_timer

    channel = ev.get('channel')
    channel_count = len(bridge.json.get('channels'))

    print "Channel %s left bridge %s" % (channel.get('name'), bridge.id)
    if holding_bridge.id == bridge.id and channel_count == 0:
        if announcer_timer:
            announcer_timer.cancel()
            announcer_timer = None

        print "Destroying bridge %s" % bridge.id
        holding_bridge.destroy()
        holding_bridge = None
    holding_bridge = bridge
    holding_bridge.on_event('ChannelLeftBridge', on_channel_left_bridge)

    # After 30 seconds, let everyone in the bridge know that we care
    announcer_timer = threading.Timer(30, play_announcement, [holding_bridge])
    announcer_timer.start()

    return bridge


def stasis_start_cb(channel_obj, ev):
    """Handler for StasisStart event"""

    bridge = find_or_create_bridge()

    channel = channel_obj.get('channel')
    print "Channel %s just entered our application, adding it to bridge %s" % (
        channel.json.get('name'), holding_bridge.id)

    channel.answer()
    bridge.addChannel(channel=channel.id)

def stasis_end_cb(channel, ev):
    """Handler for StasisEnd event"""

    print "Channel %s just left our application" % channel.json.get('name')

client.on_channel_event('StasisStart', stasis_start_cb)
client.on_channel_event('StasisEnd', stasis_end_cb)
client.run(apps='bridge-infinite-wait')
```

bridge-infinite-wait.py in action

```
Created bridge 950c4805-c33c-4895-ad9a-2798055e4939
Channel PJSIP/alice-00000000 just entered our application, adding it to bridge
950c4805-c33c-4895-ad9a-2798055e4939
Letting the everyone know we care...
Channel PJSIP/alice-00000000 left bridge 950c4805-c33c-4895-ad9a-2798055e4939
Destroying bridge 950c4805-c33c-4895-ad9a-2798055e4939
Channel PJSIP/alice-00000000 just left our application
```

JavaScript (Node.js)

When a channel enters our Stasis application, we first look for an existing holding bridge or create one if none is found. When we create a new bridge, we start music on hold in the bridge and create a timer that will call a callback after 30 seconds. That callback temporarily stops the music on hold, and starts a play operation on the bridge that thanks everyone for their patience. When the play operation finishes, it resumes music on hold.

In all cases, we add the channel to the bridge via the `joinBridge` function.

```
console.log('Channel %s just entered our application', channel.name);

    // find or create a holding bridge
    client.bridges.list(function(err, bridges) {
      if (err) {
        throw err;
      }

      var bridge = bridges.filter(function(candidate) {
        return candidate.bridge_type === 'holding';
      })[0];

      if (bridge) {
        console.log('Using bridge %s', bridge.id);
        joinBridge(bridge);
      } else {
        client.bridges.create({type: 'holding'}, function(err, newBridge) {
          if (err) {
            throw err;
          }
          console.log('Created bridge %s', newBridge.id);
          newBridge.startMoh(function(err) {
            if (err) {
              throw err;
            }
          });
          joinBridge(newBridge);

          timer = setTimeout(play_announcement, 30000);

          // callback that will let our users know how much we care
          function play_announcement() {
            console.log('Letting everyone know we care...');
            newBridge.stopMoh(function(err) {
              if (err) {
                throw err;
              }

              var playback = client.Playback();
              newBridge.play({media: 'sound:thnk-u-for-patience'},
                             playback, function(err, playback) {
                if (err) {
                  throw err;
                }
              });
              playback.once('PlaybackFinished', function(event, playback) {
                newBridge.startMoh(function(err) {
                  if (err) {
                    throw err;
                  }
                });
                timer = setTimeout(play_announcement, 30000);
              });
            });
          }
        });
      }
```

The joinBridge function involves registered a callback for the ChannelLeftBridge event and adds the channel to the bridge.

```
function joinBridge(bridge) {
      channel.once('ChannelLeftBridge', function(event, instances) {
        channelLeftBridge(event, instances, bridge);
      });

      bridge.addChannel({channel: channel.id}, function(err) {
        if (err) {
          throw err;
        }
      });
      channel.answer(function(err) {
        if (err) {
          throw err;
        }
      });
    }
```

Notice that we use an anonymous function to pass the bridge as an extra parameter to the ChannelLeftBridge callback so we can keep the handler at the same level as joinBridge and avoid another indentation level of callbacks. Finally, we can handle destroying the bridge when the last channel contained in it has left:

```
// Handler for ChannelLeftBridge event
    function channelLeftBridge(event, instances, bridge) {
      var holdingBridge = instances.bridge;
      var channel = instances.channel;

      console.log('Channel %s left bridge %s', channel.name, bridge.id);

      if (holdingBridge.id === bridge.id &&
          holdingBridge.channels.length === 0) {

        if (timer) {
          clearTimeout(timer);
        }

        bridge.destroy(function(err) {
          if (err) {
            throw err;
          }
        });
      }
    }
```

bridge-infinite-wait.js

The full source code for `bridge-infinite-wait.js` is shown below:

bridge-infinite-wait.js

```
/*jshint node:true*/
'use strict';

var ari = require('ari-client');
var util = require('util');

var timer = null;
ari.connect('http://localhost:8088', 'asterisk', 'asterisk', clientLoaded);
```

```
// handler for client being loaded
function clientLoaded (err, client) {
  if (err) {
    throw err;
  }

  // handler for StasisStart event
  function stasisStart(event, channel) {
    console.log('Channel %s just entered our application', channel.name);

    // find or create a holding bridge
    client.bridges.list(function(err, bridges) {
      if (err) {
        throw err;
      }

      var bridge = bridges.filter(function(candidate) {
        return candidate.bridge_type === 'holding';
      })[0];

      if (bridge) {
        console.log('Using bridge %s', bridge.id);
        joinBridge(bridge);
      } else {
        client.bridges.create({type: 'holding'}, function(err, newBridge) {
          if (err) {
            throw err;
          }
          console.log('Created bridge %s', newBridge.id);
          newBridge.startMoh(function(err) {
            if (err) {
              throw err;
            }
          });
          joinBridge(newBridge);

          timer = setTimeout(play_announcement, 30000);

          // callback that will let our users know how much we care
          function play_announcement() {
            console.log('Letting everyone know we care...');
            newBridge.stopMoh(function(err) {
              if (err) {
                throw err;
              }

              var playback = client.Playback();
              newBridge.play({media: 'sound:thnk-u-for-patience'},
                             playback, function(err, playback) {
                if (err) {
                  throw err;
                }
              });
              playback.once('PlaybackFinished', function(event, playback) {
                newBridge.startMoh(function(err) {
                  if (err) {
                    throw err;
                  }
                });
                timer = setTimeout(play_announcement, 30000);
              });
            });
          }
        });
      }
    });

    function joinBridge(bridge) {
      channel.once('ChannelLeftBridge', function(event, instances) {
        channelLeftBridge(event, instances, bridge);
      });

      bridge.addChannel({channel: channel.id}, function(err) {
        if (err) {
          throw err;
        }
      });
      channel.answer(function(err) {
        if (err) {
          throw err;
        }
      });
    }

    // Handler for ChannelLeftBridge event
    function channelLeftBridge(event, instances, bridge) {
      var holdingBridge = instances.bridge;
      var channel = instances.channel;

      console.log('Channel %s left bridge %s', channel.name, bridge.id);

      if (holdingBridge.id === bridge.id &&
          holdingBridge.channels.length === 0) {

        if (timer) {
          clearTimeout(timer);
        }

        bridge.destroy(function(err) {
          if (err) {
            throw err;
          }
        });
      }
    }
  }

  // handler for StasisEnd event
  function stasisEnd(event, channel) {
    console.log('Channel %s just left our application', channel.name);
  }

  client.on('StasisStart', stasisStart);
  client.on('StasisEnd', stasisEnd);
```

```
  console.log('starting');
  client.start('bridge-infinite-wait');
}
```

bridge-infinite-wait.js in action

The following shows the output of the `bridge-infinite-wait.js` script when a `PJSIP` channel for `alice` enters the application:

```
Channel PJSIP/alice-00000001 just entered our application
Created bridge 31a4a193-36a7-412b-854b-cf2cf5f90bbd
Letting everyone know we care...
Channel PJSIP/alice-00000001 left bridge 31a4a193-36a7-412b-854b-cf2cf5f90bbd
Channel PJSIP/alice-00000001 just left our application
```

ARI and Bridges: Basic Mixing Bridges

Mixing Bridges

In a mixing bridge, Asterisk shares media between all the channels in the bridge. Depending on the attributes the bridge was created with and the types of channels in the bridge, a mixing bridge may attempt to share the media in a variety of ways. They are, in order of best performance to lowest performance:

- Direct packet sharing between devices - when there are two channels in a mixing bridge of similar types, it may be possible to have the media bypass Asterisk completely. In this type of bridge, the channels will pass media directly between each other, and Asterisk will simply monitor the state of the channels. However, because the media is not going through Asterisk, most features - such as recording, speech detection, DTMF, etc. - are not available. The `proxy_media` attribute or the `dtmf_events` attribute will prevent this mixing type from being used.
- Native packet sharing through Asterisk - when there are two channels in a mixing bridge of similar types, but the media cannot flow directly between the devices, Asterisk will attempt to mix the media between the channels by directly passing media from one channel to the other, and vice versa. The media itself is not decoded, and so - much like when the media is directly shared between the devices - Asterisk cannot use many features. The `proxy_media` attribute or the `dtmf_events` attribute will prevent this mixing type from being used.
- Two party mixing - when there are two channels in a mixing bridge, regardless of the channel type, Asterisk will decode the media from each channel and pass it to the other participant. This mixing technology allows for all the various features of Asterisk to be used on the channels while they are in the bridge, but does not necessarily incur any penalties from transcoding.
- Multi-party mixing - when there are more than two channels in a mixing bridge, Asterisk will transcode the media from each participant into signed linear, mix the media from all participants together into a new media frame, then write the media back out to all participants.

At all times, the bridge will attempt to mix the media in the most performant manner possible. As the situation in the bridge changes, Asterisk will switch the mixing technology to the best mixing technology available.

What Can Happen in a Mixing Bridge

Action	Bridge Response
A bridge is created using `POST /bridges`, and Alice's channel - which supports having media be directly sent to another device - is added to the bridge using `POST /bridges/{bridge_id}/addChannel`.	Asterisk picks the basic two-party mixing technology. We don't know yet what other channel is going to join the bridge - it could be anything! - and so Asterisk picks the best one based on the information it currently has.
Bob's channel - which also supports having media be directly sent to another device - also joins the bridge via `POST /bridges/{bridge_id}/addChannel`.	We have two channels in the bridge now, so Asterisk re-evaluates how the media is mixed. Since both channels support having their media be sent directly to each other, and mixing media that way is more performant than the current mixing technology, Asterisk picks the direct media mixing technology and instructs the channels to tell their devices to send the media to each other.
Carol's channel - which is a DAHDI channel (poor Carol, calling from the PSTN) - is also added to the bridge via `POST /bridges/{bridge_id}/addChannel`.	Since we now have three channels in the bridge, Asterisk switches the mixing technology to multi-mix. Alice and Bob's media is sent back to Asterisk, and Asterisk mixes the media from Alice, Bob, and Carol together and then sends the new media to each channel.
Eventually, Alice hangs up, leaving only Bob and Carol in the bridge.	Since Alice left, Asterisk switches back to the basic two-party mixing technology. We can't use a native mixing technology, as Bob and Carol's channels are incompatible, but we can use a mixing technology that is less expensive than the multi-mix technology.

Example: Implementing a basic dial

Dialing can be implemented by using the `POST - /channels` operation and putting both the resulting channel and the original Stasis channel in a mixing bridge to allow media to flow between them. An endpoint should be specified along with the originate operation as well as a Stasis application name. This will cause the dialed channel to enter Stasis, where it can be added to a mixing bridge. It's also a good idea to use Stasis application arguments to flag that the dialed channel was dialed using originate in order to handle it differently from the original channel once it enters into the Stasis application.

This example ARI application will do the following:

1. When a channel enters into the Stasis application, a call will be originated to the endpoint specified by the first command line argument to the script.
2. When that channel enters into the Stasis application, a mixing bridge will be created and the two channels will be put in it so that media can flow between them.
3. If either channel hangs up, the other channel will also be hung up.
4. Once the dialed channel exists the Stasis application, the mixing bridge will be destroyed.

Dialplan

For this example, we'll use a Stasis application that species not only the application - `bridge-dial` - but also:

- Whether or not the channel is `inbound` or a `dialed` channel.
- If the channel is `inbound`, the endpoint to dial.

As an example, here is a dialplan that dials the `PJSIP/bob` endpoint:

extensions.conf

```
exten => 1000,1,NoOp()
 same =>      n,Stasis(bridge-dial,inbound,PJSIP/bob)
 same =>      n,Hangup()
```

Python

As is typical for an ARI application, we'll start off by implementing a callback handler for the `StasisStart` event. In this particular case, our callback handler will be called in two conditions:

1. When an inbound channel enters into the `Stasis` dialplan application. In that case, we'll want to initiate the outbound dial.
2. When an outbound channel answers. In that case, we'll want to defer processing to a different callback handler and - in that handler - initiate the bridging of the two channels.

In our `StasisStart` callback handler, we can expect to have two pieces of information passed to the application:

1. The "type" of channel entering the callback handler. In this case, we expect the type to be either `inbound` or `dialed`.
2. If the "type" of channel is `inbound`, we expect the second argument to be the endpoint to dial.

The following code shows the `StasisStart` callback handler for the `inbound` channel. Note that if the "type" is not `inbound`, we defer processing to another callback handler. We also tell the inbound channel to start ringing via the `ring` operation, and initiate an outbound dial by creating a new channel to the endpoint specified. Finally, we subscribe to the `StasisEnd` event for both channels, and instruct them to call a `safe_hangup` function on the opposing channel. This ensures that if either party hangs up, we hang up the person they were talking to. We'll show the implementation of that function shortly.

> **Be careful of errors!**
> Note that we wrap the origination with a `try / except` block, in case the endpoint provided by the dialplan doesn't exist. When taking in input from a user or from the Asterisk dialplan, it is always good to be mindful of possible errors.

```
def stasis_start_cb(channel_obj, ev):
    """Handler for StasisStart"""

    channel = channel_obj.get('channel')
    channel_name = channel.json.get('name')
    args = ev.get('args')

    if not args:
        print "Error: {} didn't provide any arguments!".format(channel_name)
        return

    if args and args[0] != 'inbound':
        # Only handle inbound channels here
        return

    if len(args) != 2:
        print "Error: {} didn't tell us who to dial".format(channel_name)
        channel.hangup()
        return

    print "{} entered our application".format(channel_name)
    channel.ring()

    try:
        print "Dialing {}".format(args[1])
        outgoing = client.channels.originate(endpoint=args[1],
                                             app='bridge-dial',
                                             appArgs='dialed')
    except requests.HTTPError:
        print "Whoops, pretty sure %s wasn't valid" % args[1]
        channel.hangup()
        return

    channel.on_event('StasisEnd', lambda *args: safe_hangup(outgoing))
    outgoing.on_event('StasisEnd', lambda *args: safe_hangup(channel))
```

The `safe_hangup` function referenced above simply does a "safe" hangup on the channel provided. This is because it is entirely possible for both parties to hang up nearly simultaneously. Since our Python code is running in a separate process from Asterisk, we may be processing the hang up of the first party and instruct Asterisk to hang up the second party when they are already technically hung up! Again, it is always a good idea to view the processing of a communications application in an asynchronous fashion: we live in an asynchronous world, and a user can take an action at any moment in time.

```
def safe_hangup(channel):
    """Safely hang up the specified channel"""
    try:
        channel.hangup()
        print "Hung up {}".format(channel.json.get('name'))
    except requests.HTTPError as e:
        if e.response.status_code != requests.codes.not_found:
            raise e
```

We now have to handle the outbound channel when it answers. Currently, when it answers it will be immediately placed in our Stasis application, which will call the `stasis_start_cb` we previously defined. While we could have some additional `if / else` blocks in that handler, we can also just apply a `StasisStart` callback to the outbound channel after we create it, and handle its entrance separately.

When the outbound channel is answered, we need to do the following:

1. Answer the inbound channel.
2. Create a new mixing bridge.
3. Put both channels into the mixing bridge.
4. Ensure we destroy the mixing bridge when either channel leaves our application.

This is shown in the following code:

```
def outgoing_start_cb(channel_obj, ev):
        """StasisStart handler for our dialed channel"""

        print "{} answered; bridging with {}".format(outgoing.json.get('name'),
                                                     channel.json.get('name'))
        channel.answer()

        bridge = client.bridges.create(type='mixing')
        bridge.addChannel(channel=[channel.id, outgoing.id])

        # Clean up the bridge when done
        channel.on_event('StasisEnd', lambda *args:
                         safe_bridge_destroy(bridge))
        outgoing.on_event('StasisEnd', lambda *args:
                          safe_bridge_destroy(bridge))

    outgoing.on_event('StasisStart', outgoing_start_cb)
```

Note that the `safe_bridge_destroy` function is similar to the `safe_hangup` function, except that it attempts to safely destroy the mixing bridge, as opposed to hanging up the other party.

bridge-dial.py

The full source code for `bridge-dial.py` is shown below:

basic-dial.py

```
#!/usr/bin/env python

import logging
import requests
import ari

logging.basicConfig(level=logging.ERROR)

client = ari.connect('http://localhost:8088', 'asterisk', 'asterisk')


def safe_hangup(channel):
    """Safely hang up the specified channel"""
    try:
        channel.hangup()
        print "Hung up {}".format(channel.json.get('name'))
    except requests.HTTPError as e:
        if e.response.status_code != requests.codes.not_found:
            raise e


def safe_bridge_destroy(bridge):
    """Safely destroy the specified bridge"""
    try:
        bridge.destroy()
    except requests.HTTPError as e:
        if e.response.status_code != requests.codes.not_found:
            raise e
```

```
def stasis_start_cb(channel_obj, ev):
    """Handler for StasisStart"""

    channel = channel_obj.get('channel')
    channel_name = channel.json.get('name')
    args = ev.get('args')

    if not args:
        print "Error: {} didn't provide any arguments!".format(channel_name)
        return

    if args and args[0] != 'inbound':
        # Only handle inbound channels here
        return

    if len(args) != 2:
        print "Error: {} didn't tell us who to dial".format(channel_name)
        channel.hangup()
        return

    print "{} entered our application".format(channel_name)
    channel.ring()

    try:
        print "Dialing {}".format(args[1])
        outgoing = client.channels.originate(endpoint=args[1],
                                             app='bridge-dial',
                                             appArgs='dialed')
    except requests.HTTPError:
        print "Whoops, pretty sure %s wasn't valid" % args[1]
        channel.hangup()
        return

    channel.on_event('StasisEnd', lambda *args: safe_hangup(outgoing))
    outgoing.on_event('StasisEnd', lambda *args: safe_hangup(channel))

    def outgoing_start_cb(channel_obj, ev):
        """StasisStart handler for our dialed channel"""

        print "{} answered; bridging with {}".format(outgoing.json.get('name'),
                                                     channel.json.get('name'))
        channel.answer()

        bridge = client.bridges.create(type='mixing')
        bridge.addChannel(channel=[channel.id, outgoing.id])

        # Clean up the bridge when done
        channel.on_event('StasisEnd', lambda *args:
                         safe_bridge_destroy(bridge))
        outgoing.on_event('StasisEnd', lambda *args:
                          safe_bridge_destroy(bridge))

    outgoing.on_event('StasisStart', outgoing_start_cb)


client.on_channel_event('StasisStart', stasis_start_cb)
client.run(apps='bridge-dial')
```

bridge-dial.py in action

The following shows the output of the `bridge-dial.py` script when a `PJSIP` channel for `alice` enters the application and dials a `PJSIP` channel for `bob`:

```
PJSIP/Alice-00000001 entered our application
Dialing PJSIP/Bob
PJSIP/Bob-00000002 answered; bridging with PJSIP/Alice-00000001
Hung up PJSIP/Bob-00000002
```

JavaScript (Node.js)

This example shows how to use anonymous functions to call functions with extra parameters that would otherwise require a closer. This can be done to reduce the number of nested callbacks required to implement the flow of an application. First, we look for an application argument in our `StasisStart` event callback to ensure that we will only originate a call if the channel entering Stasis is a channel that dialed our application extension we defined in the extensions.conf file above. We then play a sound on the channel asking the caller to wait while they are being connected and call the originate() function to process down the application flow:

```
function stasisStart(event, channel) {
  // ensure the channel is not a dialed channel
  var dialed = event.args[0] === 'dialed';

  if (!dialed) {
    channel.answer(function(err) {
      if (err) {
        throw err;
      }

      console.log('Channel %s has entered our application', channel.name);

      var playback = client.Playback();
      channel.play({media: 'sound:pls-wait-connect-call'},
        playback, function(err, playback) {
          if (err) {
            throw err;
          }
      });

      originate(channel);
    });
  }
}
```

We then prepare an object with a locally generate Id for the dialed channel and register event callbacks either channels hanging up and the dialed channel entering into the Stasis application. We then originate a call to the endpoint specified by the first command line argument to the script passing in a Stasis application argument of dialed so we can skip the dialed channel when the original StasisStart event callback fires for it:

```
function originate(channel) {
  var dialed = client.Channel();

  channel.on('StasisEnd', function(event, channel) {
    hangupDialed(channel, dialed);
  });

  dialed.on('ChannelDestroyed', function(event, dialed) {
    hangupOriginal(channel, dialed);
  });

  dialed.on('StasisStart', function(event, dialed) {
    joinMixingBridge(channel, dialed);
  });

  dialed.originate(
    {endpoint: process.argv[2], app: 'bridge-dial', appArgs: 'dialed'},
    function(err, dialed) {
      if (err) {
        throw err;
      }
  });
}
```

We then handle either channel hanging up by hanging up the other channel. Note that we skip any errors that occur on hangup since it is possible that the channel we are attempting to hang up is the one that has already left and would result in an HTTP error as it is no longer a Statis channel:

```
function hangupDialed(channel, dialed) {
  console.log(
    'Channel %s left our application, hanging up dialed channel %s',
    channel.name, dialed.name);

  // hangup the other end
  dialed.hangup(function(err) {
    // ignore error since dialed channel could have hung up, causing the
    // original channel to exit Stasis
  });
}

// handler for the dialed channel hanging up so we can gracefully hangup the
// other end
function hangupOriginal(channel, dialed) {
  console.log('Dialed channel %s has been hung up, hanging up channel %s',
    dialed.name, channel.name);

  // hangup the other end
  channel.hangup(function(err) {
    // ignore error since original channel could have hung up, causing the
    // dialed channel to exit Stasis
  });
}
```

We then handle the StasisStart event for the dialed channel by registered an event callback for the StasisEnd event on the dialed channel, answer that answer, creating a new mixing bridge, and finally calling a function to add the two channels to the new bridge:

```
function joinMixingBridge(channel, dialed) {
  var bridge = client.Bridge();

  dialed.on('StasisEnd', function(event, dialed) {
    dialedExit(dialed, bridge);
  });

  dialed.answer(function(err) {
    if (err) {
      throw err;
    }
  });

  bridge.create({type: 'mixing'}, function(err, bridge) {
    if (err) {
      throw err;
    }

    console.log('Created bridge %s', bridge.id);

    addChannelsToBridge(channel, dialed, bridge);
  });
}
```

We then handle the dialed channel exiting the Stasis application by destroying the mixing bridge:

```
function dialedExit(dialed, bridge) {
  console.log(
      'Dialed channel %s has left our application, destroying bridge %s',
      dialed.name, bridge.id);

  bridge.destroy(function(err) {
    if (err) {
      throw err;
    }
  });
}
```

Finally, the function that was called earlier by the callback handling the StasisStart event for the dialed channel adds the two channels to the mixing bridge which allows media to flow between the two channels:

```
function addChannelsToBridge(channel, dialed, bridge) {
  console.log('Adding channel %s and dialed channel %s to bridge %s',
      channel.name, dialed.name, bridge.id);

  bridge.addChannel({channel: [channel.id, dialed.id]}, function(err) {
    if (err) {
      throw err;
    }
  });
}
```

bridge-dial.js

The full source code for `bridge-dial.js` is shown below:

bridge-dial.js

```
/*jshint node:true*/
```

```
'use strict';

var ari = require('ari-client');
var util = require('util');

// ensure endpoint was passed in to script
if (!process.argv[2]) {
  console.error('usage: node bridge-dial.js endpoint');
  process.exit(1);
}

ari.connect('http://localhost:8088', 'asterisk', 'asterisk', clientLoaded);

// handler for client being loaded
function clientLoaded (err, client) {
  if (err) {
    throw err;
  }

  // handler for StasisStart event
  function stasisStart(event, channel) {
    // ensure the channel is not a dialed channel
    var dialed = event.args[0] === 'dialed';

    if (!dialed) {
      channel.answer(function(err) {
        if (err) {
          throw err;
        }

        console.log('Channel %s has entered our application', channel.name);

        var playback = client.Playback();
        channel.play({media: 'sound:pls-wait-connect-call'},
          playback, function(err, playback) {
            if (err) {
              throw err;
            }
        });

        originate(channel);
      });
    }
  }

  function originate(channel) {
    var dialed = client.Channel();

    channel.on('StasisEnd', function(event, channel) {
      hangupDialed(channel, dialed);
    });

    dialed.on('ChannelDestroyed', function(event, dialed) {
      hangupOriginal(channel, dialed);
    });

    dialed.on('StasisStart', function(event, dialed) {
      joinMixingBridge(channel, dialed);
    });
    dialed.originate(
      {endpoint: process.argv[2], app: 'bridge-dial', appArgs: 'dialed'},
      function(err, dialed) {
        if (err) {
          throw err;
        }
    });
  }

  // handler for original channel hanging up so we can gracefully hangup the
  // other end
  function hangupDialed(channel, dialed) {
    console.log(
      'Channel %s left our application, hanging up dialed channel %s',
      channel.name, dialed.name);

    // hangup the other end
    dialed.hangup(function(err) {
      // ignore error since dialed channel could have hung up, causing the
      // original channel to exit Stasis
    });
  }

  // handler for the dialed channel hanging up so we can gracefully hangup the
  // other end
  function hangupOriginal(channel, dialed) {
    console.log('Dialed channel %s has been hung up, hanging up channel %s',
      dialed.name, channel.name);

    // hangup the other end
    channel.hangup(function(err) {
      // ignore error since original channel could have hung up, causing the
      // dialed channel to exit Stasis
    });
  }

  // handler for dialed channel entering Stasis
  function joinMixingBridge(channel, dialed) {
    var bridge = client.Bridge();

    dialed.on('StasisEnd', function(event, dialed) {
      dialedExit(dialed, bridge);
    });

    dialed.answer(function(err) {
      if (err) {
        throw err;
      }
    });

    bridge.create({type: 'mixing'}, function(err, bridge) {
      if (err) {
        throw err;
      }

      console.log('Created bridge %s', bridge.id);

      addChannelsToBridge(channel, dialed, bridge);
    });
  }

  // handler for the dialed channel leaving Stasis
  function dialedExit(dialed, bridge) {
    console.log(
        'Dialed channel %s has left our application, destroying bridge %s',
        dialed.name, bridge.id);

    bridge.destroy(function(err) {
      if (err) {
        throw err;
      }
    });
  }

  // handler for new mixing bridge ready for channels to be added to it
  function addChannelsToBridge(channel, dialed, bridge) {
    console.log('Adding channel %s and dialed channel %s to bridge %s',
        channel.name, dialed.name, bridge.id);

    bridge.addChannel({channel: [channel.id, dialed.id]}, function(err) {
      if (err) {
        throw err;
      }
    });
  }

  client.on('StasisStart', stasisStart);

  client.start('bridge-dial');
}
```

bridge-dial.js in action

The following shows the output of the `bridge-dial.js` script when a `PJSIP` channel for `alice` enters the application and dials a PJSIP channel for bob:

```
Channel PJSIP/alice-00000001 has entered our application
Created bridge 30430e82-83ed-4242-9f37-1bc040f70724
Adding channel PJSIP/alice-00000001 and dialed channel PJSIP/bob-00000002 to bridge
30430e82-83ed-4242-9f37-1bc040f70724
Dialed channel PJSIP/bob-00000002 has left our application, destroying bridge
30430e82-83ed-4242-9f37-1bc040f70724
Dialed channel PJSIP/bob-00000002 has been hung up, hanging up channel
PJSIP/alice-00000001
Channel PJSIP/alice-00000001 left our application, hanging up dialed channel undefined
```

ARI and Bridges: Bridge Operations

Moving Between Bridges

Channels can be both added and removed from bridges via the `POST - /bridges/{bridgeId}/addChannel` and `POST - /bridges/{bridgeId}/removeChannel` operations. This allows channels to be put in a holding bridge while waiting for an application to continue to its next step for example. One example of this would be to put an incoming channel into a holding bridge playing music on hold while dialing another endpoint. Once that endpoint answers, the incoming channel can be moved from the holding bridge to a mixing bridge to establish an audio call between the two channels.

Example: Dialing with Entertainment

This example ARI application will do the following:

1. When a channel enters into the Stasis application, it will be put in a holding bridge and a call will be originated to the endpoint specified by the first command line argument to the script.
2. When that channel enters into the Stasis application, the original channel will be removed from the holding bridge, a mixing bridge will be created, and the two channels will be put in it.
3. If either channel hangs up, the other channel will also be hung up.
4. Once the dialed channel exists the Stasis application, the mixing bridge will be destroyed.

Dialplan

For this example, we need to just drop the channel into Stasis, specifying our application:

extensions.conf

```
exten => 1000,1,NoOp()
 same =>      n,Stasis(bridge-move,inbound,PJSIP/bob)
 same =>      n,Hangup()
```

Python

A large part of the implementation of this particular example is similar to the `bridge-dial.py` example. However, instead of ringing the inbound channel, we'll instead create a holding bridge and place the channel in said holding bridge. Since a holding bridge can hold a number of channels, we'll reuse the same holding bridge for all of the channels that use the application. The method to obtain the holding bridge is `find_or_create_holding_bridge`, shown below:

```
# Our one and only holding bridge
holding_bridge = None


def find_or_create_holding_bridge():
    """Find our infinite wait bridge, or create a new one

    Returns:
    The one and only holding bridge
    """
    global holding_bridge

    if holding_bridge:
        return holding_bridge

    bridges = [candidate for candidate in client.bridges.list() if
               candidate.json['bridge_type'] == 'holding']
    if bridges:
        bridge = bridges[0]
        print "Using bridge {}".format(bridge.id)
    else:
        bridge = client.bridges.create(type='holding')
        bridge.startMoh()
        print "Created bridge {}".format(bridge.id)

    holding_bridge = bridge
    return holding_bridge
```

When the inbound channel enters the application, we'll place it into our waiting bridge:

```
wait_bridge = find_or_create_holding_bridge()
    wait_bridge.addChannel(channel=channel.id)
```

When the dialed channel answers, we can remove the inbound channel from the waiting bridge - since there is only one waiting bridge being used, we can use `find_or_create_holding_bridge` to obtain it. We then place it into a newly created mixing bridge along with the dialed channel, in the same fashion as the `bridge-dial.py` example.

```
print "{} answered; bridging with {}".format(outgoing.json.get('name'),
                                                     channel.json.get('name'))

        wait_bridge = find_or_create_holding_bridge()
        wait_bridge.removeChannel(channel=channel.id)

        bridge = client.bridges.create(type='mixing')
        bridge.addChannel(channel=[channel.id, outgoing.id])
```

bridge-move.py

The full source code for `bridge-move.py` is shown below:

```
#!/usr/bin/env python

import logging
import requests
import ari

logging.basicConfig(level=logging.ERROR)
```

```
client = ari.connect('http://localhost:8088', 'asterisk', 'asterisk')

# Our one and only holding bridge
holding_bridge = None


def find_or_create_holding_bridge():
    """Find our infinite wait bridge, or create a new one

    Returns:
    The one and only holding bridge
    """
    global holding_bridge

    if holding_bridge:
        return holding_bridge

    bridges = [candidate for candidate in client.bridges.list() if
               candidate.json['bridge_type'] == 'holding']
    if bridges:
        bridge = bridges[0]
        print "Using bridge {}".format(bridge.id)
    else:
        bridge = client.bridges.create(type='holding')
        bridge.startMoh()
        print "Created bridge {}".format(bridge.id)

    holding_bridge = bridge
    return holding_bridge


def safe_hangup(channel):
    """Safely hang up the specified channel"""
    try:
        channel.hangup()
        print "Hung up {}".format(channel.json.get('name'))
    except requests.HTTPError as e:
        if e.response.status_code != requests.codes.not_found:
            raise e


def safe_bridge_destroy(bridge):
    """Safely destroy the specified bridge"""
    try:
        bridge.destroy()
    except requests.HTTPError as e:
        if e.response.status_code != requests.codes.not_found:
            raise e


def stasis_start_cb(channel_obj, ev):
    """Handler for StasisStart"""

    channel = channel_obj.get('channel')
    channel_name = channel.json.get('name')
    args = ev.get('args')

    if not args:
        print "Error: {} didn't provide any arguments!".format(channel_name)
        return

    if args and args[0] != 'inbound':
        # Only handle inbound channels here
        return

    if len(args) != 2:
        print "Error: {} didn't tell us who to dial".format(channel_name)
        channel.hangup()
        return

    wait_bridge = find_or_create_holding_bridge()
    wait_bridge.addChannel(channel=channel.id)

    try:
        outgoing = client.channels.originate(endpoint=args[1],
                                             app='bridge-move',
                                             appArgs='dialed')
    except requests.HTTPError:
        print "Whoops, pretty sure %s wasn't valid" % args[1]
        channel.hangup()
        return

    channel.on_event('StasisEnd', lambda *args: safe_hangup(outgoing))
    outgoing.on_event('StasisEnd', lambda *args: safe_hangup(channel))

    def outgoing_start_cb(channel_obj, ev):
        """StasisStart handler for our dialed channel"""

        print "{} answered; bridging with {}".format(outgoing.json.get('name'),
                                                     channel.json.get('name'))

        wait_bridge = find_or_create_holding_bridge()
        wait_bridge.removeChannel(channel=channel.id)

        bridge = client.bridges.create(type='mixing')
        bridge.addChannel(channel=[channel.id, outgoing.id])

        # Clean up the bridge when done
        channel.on_event('StasisEnd', lambda *args:
                         safe_bridge_destroy(bridge))
        outgoing.on_event('StasisEnd', lambda *args:
                          safe_bridge_destroy(bridge))

    outgoing.on_event('StasisStart', outgoing_start_cb)


client.on_channel_event('StasisStart', stasis_start_cb)
client.run(apps='bridge-move')
```

bridge-move.py in action

The following shows the output of the `bridge-move.py` script when a `PJSIP` channel for `alice` enters the application and dials a PJSIP channel for bob:

```
PJSIP/Alice-00000001 entered our application
Dialing PJSIP/Bob
PJSIP/Bob-00000002 answered; bridging with PJSIP/Alice-00000001
Hung up PJSIP/Bob-00000002
```

JavaScript (Node.js)

This example is very similar to bridge-dial.js with one main difference: the original Stasis channel is put in a holding bridge while the an originate operation is used to dial another channel. Once the dialed channel enters into the Stasis application, the original channel will be removed from the holding bridge, and both channels will finally be put into a mixing bridge. Once a channel enters into our Stasis application, we either find an existing holding bridge or create one:

```
function findOrCreateHoldingBridge(channel) {
  client.bridges.list(function(err, bridges) {
    var holdingBridge = bridges.filter(function(candidate) {
      return candidate.bridge_type === 'holding';
    })[0];

    if (holdingBridge) {
      console.log('Using existing holding bridge %s', holdingBridge.id);

      originate(channel, holdingBridge);
    } else {
      client.bridges.create({type: 'holding'}, function(err, holdingBridge) {
        if (err) {
          throw err;
        }

        console.log('Created new holding bridge %s', holdingBridge.id);

        originate(channel, holdingBridge);
      });
    }
  });
}
```

We then add the channel to the holding bridge and start music on hold before continuing with dialing we we did in the bridge-dial.js example:

```
holdingBridge.addChannel({channel: channel.id}, function(err) {
  if (err) {
    throw err;
  }
   holdingBridge.startMoh(function(err) {
    // ignore error
  });
});
```

Once the endpoint has answered and a mixing bridge has been created, we proceed by first removing the original channel from the holding bridge and then adding both channels to the mixing bridge as before:

```
function moveToMixingBridge(channel, dialed, mixingBridge, holdingBridge) {
  console.log('Adding channel %s and dialed channel %s to bridge %s',
      channel.name, dialed.name, mixingBridge.id);

  holdingBridge.removeChannel({channel: channel.id}, function(err) {
    if (err) {
      throw err;
    }

    mixingBridge.addChannel(
        {channel: [channel.id, dialed.id]}, function(err) {
      if (err) {
        throw err;
      }
    });
  });
}
```

Note that we need to keep track of one more variable as we go down the application flow to ensure we have a reference to both the holding and mixing bridge. Again we use anonymous functions to pass extra arguments to callback handlers to keep the nested callbacks to a minimum.

bridge-move.js

The full source code for `bridge-move.js` is shown below:

bridge-move.js

```
 /*jshint node:true*/
'use strict';

var ari = require('ari-client');
var util = require('util');

// ensure endpoint was passed in to script
if (!process.argv[2]) {
  console.error('usage: node bridge-move.js endpoint');
  process.exit(1);
}

ari.connect('http://localhost:8088', 'asterisk', 'asterisk', clientLoaded);

// handler for client being loaded
function clientLoaded (err, client) {
  if (err) {
    throw err;
  }

  // handler for StasisStart event
  function stasisStart(event, channel) {
    // ensure the channel is not a dialed channel
    var dialed = event.args[0] === 'dialed';

    if (!dialed) {
      channel.answer(function(err) {
        if (err) {
          throw err;
        }

        console.log('Channel %s has entered our application', channel.name);
        findOrCreateHoldingBridge(channel);
      });
    }
  }

  function findOrCreateHoldingBridge(channel) {
    client.bridges.list(function(err, bridges) {
      var holdingBridge = bridges.filter(function(candidate) {
        return candidate.bridge_type === 'holding';
      })[0];

      if (holdingBridge) {
        console.log('Using existing holding bridge %s', holdingBridge.id);

        originate(channel, holdingBridge);
      } else {
        client.bridges.create({type: 'holding'}, function(err, holdingBridge) {
          if (err) {
            throw err;
          }

          console.log('Created new holding bridge %s', holdingBridge.id);

          originate(channel, holdingBridge);
        });
      }
    });
  }

  function originate(channel, holdingBridge) {
    holdingBridge.addChannel({channel: channel.id}, function(err) {
      if (err) {
        throw err;
      }

      holdingBridge.startMoh(function(err) {
        // ignore error
      });
    });

    var dialed = client.Channel();

    channel.on('StasisEnd', function(event, channel) {
      safeHangup(dialed);
    });

    dialed.on('ChannelDestroyed', function(event, dialed) {
      safeHangup(channel);
    });

    dialed.on('StasisStart', function(event, dialed) {
      joinMixingBridge(channel, dialed, holdingBridge);
    });

    dialed.originate(
      {endpoint: process.argv[2], app: 'bridge-move', appArgs: 'dialed'},
      function(err, dialed) {
        if (err) {
          throw err;
        }
    });
  }

  // safely hangs the given channel
  function safeHangup(channel) {
    console.log('Hanging up channel %s', channel.name);

    channel.hangup(function(err) {
      // ignore error
    });
  }

  // handler for dialed channel entering Stasis
  function joinMixingBridge(channel, dialed, holdingBridge) {
    var mixingBridge = client.Bridge();

    dialed.on('StasisEnd', function(event, dialed) {
      dialedExit(dialed, mixingBridge);
    });

    dialed.answer(function(err) {
      if (err) {
        throw err;
      }
    });

    mixingBridge.create({type: 'mixing'}, function(err, mixingBridge) {
      if (err) {
        throw err;
      }

      console.log('Created mixing bridge %s', mixingBridge.id);

      moveToMixingBridge(channel, dialed, mixingBridge, holdingBridge);
    });
  }

  // handler for the dialed channel leaving Stasis
  function dialedExit(dialed, mixingBridge) {
    console.log(
      'Dialed channel %s has left our application, destroying mixing bridge %s',
      dialed.name, mixingBridge.id);

    mixingBridge.destroy(function(err) {
      if (err) {
        throw err;
      }
    });
  }

  // handler for new mixing bridge ready for channels to be added to it
  function moveToMixingBridge(channel, dialed, mixingBridge, holdingBridge) {
    console.log('Adding channel %s and dialed channel %s to bridge %s',
        channel.name, dialed.name, mixingBridge.id);

    holdingBridge.removeChannel({channel: channel.id}, function(err) {
      if (err) {
        throw err;
      }

      mixingBridge.addChannel(
          {channel: [channel.id, dialed.id]}, function(err) {
        if (err) {
          throw err;
        }
      });
    });
  }

  client.on('StasisStart', stasisStart);
```

```
  client.start('bridge-move');
}
```

bridge-move.js in action

The following shows the output of the `bridge-move.js` script when a `PJSIP` channel for `alice` enters the application and dials a PJSIP channel for bob:

```
Channel PJSIP/alice-00000001 has entered our application
Created new holding bridge e58641af-2006-4c3d-bf9e-8817baa27381
Created mixing bridge 5ae49fee-e353-4ad9-bfa7-f8306d9dfd1e
Adding channel PJSIP/alice-00000001 and dialed channel PJSIP/bob-00000002 to bridge
5ae49fee-e353-4ad9-bfa7-f8306d9dfd1e
Dialed channel PJSIP/bob-00000002 has left our application, destroying mixing bridge
5ae49fee-e353-4ad9-bfa7-f8306d9dfd1e
Hanging up channel PJSIP/alice-00000001
Hanging up channel undefined
```

Introduction to ARI and Media Manipulation

Media Control

ARI contains tools for manipulating media, such as playing sound files, playing tones, playing numbers and digits, recording media, deleting stored recordings, manipulating playbacks (e.g. rewind and fast-forward), and intercepting DTMF tones. Some channel-specific information and examples for playing media and intercepting DTMF have been covered on previous pages. While some information will be repeated here, the intention of this section is to delve deeper into media manipulation.

To frame the discussion, we will be creating a set of applications that mimic a minimal voice mail system.

About the Code Samples

The following ARI client libraries are used in the code samples on these pages

- Python code samples use ari-py
- Node.js samples use node-ari-client

On This Page

Media In Depth

- ARI and Media: Part 1 - Recording
- ARI and Media: Part 2 - Playbacks

All of the code presented here has been tested with Asterisk 13 and works as intended. That being said, the code samples given are intended more to demonstrate the capabilities of ARI than to be a best practices guide for writing an application or to illustrate watertight code. Error-handling is virtually non-existent in the code samples. For a real application, Python calls to ARI should likely be in `try-catch` blocks in case there is an error, and Node.js calls to ARI should provide callbacks that detect if there was an error.

The asynchronous nature of Node.js and the node-ari-client library is not always used in the safest ways in the code samples provided. For instance, there are code samples where DTMF presses cause media operations to take place, and the code does not await confirmation that the media operation has actually completed before accepting more DTMF presses. This could potentially result in the desired media operations happening out of order if many DTMF presses occur in rapid succession.

State Machine

Voice mail, at its heart, is an IVR. IVRs are most easily represented using a finite state machine. The way a state machine works is that a program switches between pre-defined states based on events that occur. Certain events will cause program code within the state to take certain actions, and other events will result in a change to a different program state. Each state can almost be seen as a self-contained program.

To start our state machine, we will define what events might cause state transitions. If you think about a typical IVR, the events that can occur are DTMF key presses, and changes in the state of a call, such as hanging up. As such, we'll define a base set of events.

event.py

Expand source

```
class Event(object):
    # DTMF digits
    DTMF_1 = "1"
    DTMF_2 = "2"
    DTMF_3 = "3"
    DTMF_4 = "4"
    DTMF_5 = "5"
    DTMF_6 = "6"
    DTMF_7 = "7"
    DTMF_8 = "8"
    DTMF_9 = "9"
    DTMF_0 = "0"
    # Use "octothorpe" so there is no confusion about "pound" or "hash"
    # terminology.
    DTMF_OCTOTHORPE = "#"
    DTMF_STAR = "*"
    # Call has hung up
    HANGUP = "hangup"
    # Playback of a file has completed
    PLAYBACK_COMPLETE = "playback_complete"
    # Mailbox has been emptied
    MAILBOX_EMPTY = "empty"
```

event.js

Expand source

```
var Event = {
    // DTMF digits
    DTMF_1: "1",
    DTMF_2: "2",
    DTMF_3: "3",
    DTMF_4: "4",
    DTMF_5: "5",
    DTMF_6: "6",
    DTMF_7: "7",
    DTMF_8: "8",
    DTMF_9: "9",
    DTMF_0: "0",
    // Use "octothorpe" so there is no confusion about "pound" or "hash"
    // terminology.
    DTMF_OCTOTHORPE: "#",
    DTMF_STAR: "*",
    // Call has hung up
    HANGUP: "hangup",
    // Playback of a file has completed
    PLAYBACK_COMPLETE: "playback_complete",
    // Mailbox has been emptied
    MAILBOX_EMPTY: "empty"
}
module.exports = Event;
```

There is no hard requirement for our application that we define events as named constants, but doing so makes it easier for tools like pylint and jslint to find potential mistakes.

After we have defined our events, we need to create a state machine itself. The state machine keeps track of what the current state is, and which events cause state changes. Here is a simple implementation of a state machine

state_machine.py Expand source

```
class StateMachine(object):
    def __init__(self):
        self.transitions = {}
        self.current_state = None

    def add_transition(self, src_state, event, dst_state):
        if not self.transitions.get(src_state.state_name):
            self.transitions[src_state.state_name] = {}

        self.transitions[src_state.state_name][event] = dst_state

    def change_state(self, event):
        self.current_state = self.transitions[self.current_state.state_name][event]
        self.current_state.enter()

    def start(self, initial_state):
        self.current_state = initial_state
        self.current_state.enter()
```

state_machine.js Expand source

```
function StateMachine() {
    var transitions = {};
    var current_state = null;

    this.add_transition = function(src_state, event, dst_state) {
        if (!transitions.hasOwnProperty(src_state.state_name)) {
            transitions[src_state.state_name] = {};
        }
        transitions[src_state.state_name][event] = dst_state;
    }

    this.change_state = function(event) {
        current_state = transitions[current_state.state_name][event];
        current_state.enter();
    }

    this.start = function(initial_state) {
        current_state = initial_state;
        current_state.enter();
    }
}

module.exports = StateMachine;
```

The state machine code is pretty straightforward. The state machine has transitions added to it with the `add_transition()` method and can be started with the `start()` method. Our use of the state machine will always be to define all transitions, and then to start the state machine.

States within a state machine have certain duties that they must fulfill if they want to work well in the state machine we have devised

- A state must know what events should cause it to change states, though it does not need to know what state it will be transitioning to.
- A state must set up ARI event listeners each time the state is entered, and it must remove these before changing states.
- The state must define the following attributes:
    - `state_name`, a string that represents the name of the state.
    - `enter()`, a method that is called whenever the state is entered.

It should be noted that this state machine implementation is not necessarily ideal, since it requires the states to know what events cause it to change states. However, it will become clear later that for a simple voice mail system, this is not that big a deal. To see how we use this state machine, continue on to the sub-pages.

ARI and Media: Part 1 - Recording

The Recording API

Recordings in ARI are divided into two main categories: live and stored. Live recordings are those that are currently being recorded on a channel or bridge, and stored recordings are recordings that have been completed and saved to the file system. The API for the `/recordings` resource can be found here.

Live recordings can be manipulated as they are being made, with options to manipulate the flow of audio such as muting, pausing, stopping, or canceling the recording. Stored recordings are simply files on the file system on which Asterisk is installed. The location of stored recordings is in the `/recording` subdirectory of the configured `astspooldir` in `asterisk.conf`. By default, this places recordings in `/var/spool/asterisk/recording`.

Channels can have their audio recorded using the `/channels/{channelId}/record` resource, and Bridges can have their audio recorded using the `/bridges/{bridgeId}/record` resource.

Voice Mail Application Skeleton

Our application to record voice mails will be based on the following skeleton. As we add new features, we will create new states for our state machine, and add a few extra lines to our application skeleton in order to link the states together appropriately.

On this Page

vm-record.py

Expand source

```
#!/usr/bin/env python

import ari
import logging
import time
import os
import sys

from event import Event
from state_machine import StateMachine
# As we add more states to our state machine, we'll import the necessary
# states here.

logging.basicConfig(level=logging.ERROR)
LOGGER = logging.getLogger(__name__)

client = ari.connect('http://localhost:8088', 'asterisk', 'asterisk')


class VoiceMailCall(object):
    def __init__(self, ari_client, channel, mailbox):
        self.client = ari_client
        self.channel = channel
        self.vm_path = os.path.join('voicemail', mailbox, str(time.time()))
        self.setup_state_machine()

    def setup_state_machine(self):
        # This is where we will initialize states, create a state machine, add
        # state transitions to the state machine, and start the state machine.


def stasis_start_cb(channel_obj, event):
    channel = channel_obj['channel']
    channel_name = channel.json.get('name')
    mailbox = event.get('args')[0]

    print("Channel {0} recording voicemail for {1}".format(
        channel_name, mailbox))

    channel.answer()
    VoiceMailCall(client, channel, mailbox)


client.on_channel_event('StasisStart', stasis_start_cb)
client.run(apps=sys.argv[1])
```

vm-record.js

Expand source

```
/*jshint node:true*/
'use strict';

var ari = require('ari-client');
var util = require('util');
var path = require('path');

var Event = require('./event');
var StateMachine = require('./state_machine');
// As we add new states to our state machine, this is where we will
// add the new required states.

ari.connect('http://localhost:8088', 'asterisk', 'asterisk', clientLoaded);

var VoiceMailCall = function(ari_client, channel, mailbox) {
    this.client = ari_client;
    this.channel = channel;
    this.vm_path = path.join('voicemail', mailbox, Date.now().toString());

 this.setup_state_machine = function() {
    // This is where we will initialize states, create a state machine, add
    // state transitions to the state machine, and start the state machine.
    }

    this.setup_state_machine();
}


function clientLoaded(err, client) {
    if (err) {
        throw err;
    }

    client.on('StasisStart', stasisStart);

    function stasisStart(event, channel) {
        var mailbox = event.args[0]
        channel.answer(function(err) {
            if (err) {
                throw err;
            }
            new VoiceMailCall(client, channel, mailbox);
        });
    }

    client.start(process.argv[2]);
}
```

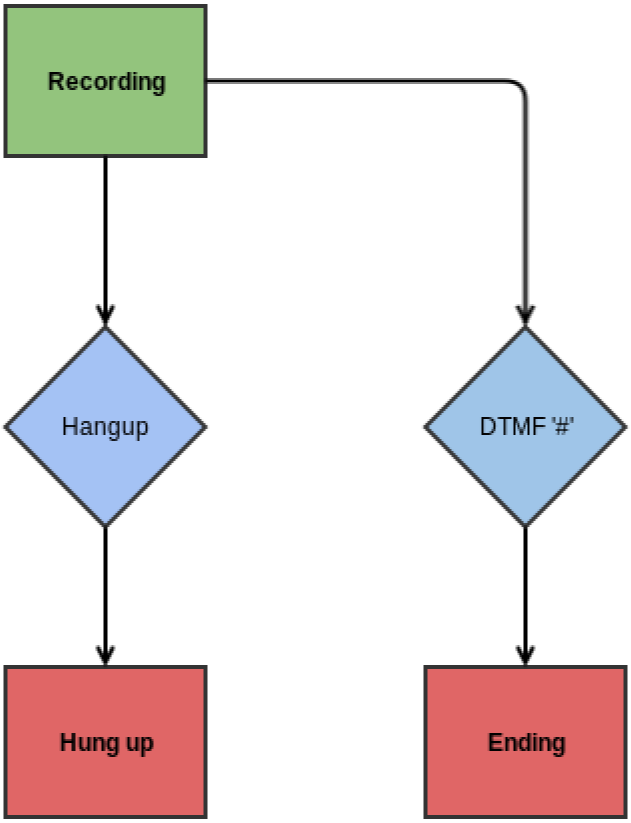
With a few modifications, this same application skeleton can be adapted for use with non-voice mail applications. The biggest voice mail-specific thing being done here is the calculation of the path for voice mail recordings based on the application argument. The intended use of this application is something like the following:

```
[default]
exten => _3XX,1,NoOp()
        same => n,Stasis(vm-record, ${EXTEN})
        same => n,Hangup()
```

This way, when calling any three-digit extension that begins with the number '3', the user will call into the application with the mailbox dialled (e.g. dialling "305" will allow the user to leave a message for mailbox "305").

Basic Voice Mail Recording

We've seen a lot of the underlying concepts for our application, so let's actually make something useful now. We'll start with a very simple application that allows callers to record a message upon entering the application. When the caller has completed recording the message, the caller may press the '#' key or may hang up to accept the recording. Here is a state machine diagram for the application:



Notice that even though DTMF '#' and a caller hangup result in the same end result that there are two separate states that are transitioned into. This is because our code needs to behave differently at the end of a voice mail call based on whether the channel has been hung up or not. In short, the "Ending" state in the diagram will forcibly hang up the channel, whereas the "Hung up" state does not need to do so.

For this, we will be defining three states: recording, hungup, and ending. The following is the code for the three states:

recording_state.py

Expand source

```
from event import Event

class RecordingState(object):
    state_name = "recording"

    def __init__(self, call):
        self.call = call
        self.hangup_event = None
        self.dtmf_event = None
        self.recording = None

    def enter(self):
        print "Entering recording state"
        self.hangup_event = self.call.channel.on_event('ChannelHangupRequest',
                self.on_hangup)
        self.dtmf_event = self.call.channel.on_event('ChannelDtmfReceived',
                                                     self.on_dtmf)
        self.recording = self.call.channel.record(name=self.call.vm_path,
                                                  format='wav',
                                                  beep=True,
                                                  ifExists='overwrite')
        print "Recording voicemail at {0}".format(self.call.vm_path)

    def cleanup(self):
        print "Cleaning up event handlers"
        self.dtmf_event.close()
        self.hangup_event.close()

    def on_hangup(self, channel, event):
        print "Accepted recording {0} on hangup".format(self.call.vm_path)
        self.cleanup()
        self.call.state_machine.change_state(Event.HANGUP)

    def on_dtmf(self, channel, event):
        digit = event.get('digit')
        if digit == '#':
            rec_name = self.recording.json.get('name')
            print "Accepted recording {0} on DTMF #".format(rec_name)
            self.cleanup()
            self.recording.stop()
            self.call.state_machine.change_state(Event.DTMF_OCTOTHORPE)
```

recording_state.js

Expand source

```
var Event = require('./event')

function RecordingState(call) {
    this.state_name = "recording";

    this.enter = function() {
        var recording = call.client.LiveRecording(call.client, {name: call.vm_path});
        console.log("Entering recording state");
        call.channel.on("ChannelHangupRequest", on_hangup);
        call.channel.on("ChannelDtmfReceived", on_dtmf);
        call.channel.record({name: recording.name, format: 'wav', beep: true, ifExists:
'overwrite'}, recording);

        function cleanup() {
            call.channel.removeListener('ChannelHangupRequest', on_hangup);
            call.channel.removeListener('ChannelDtmfReceived', on_dtmf);
        }

        function on_hangup(event, channel) {
            console.log("Accepted recording %s on hangup", recording.name);
            cleanup();
            call.state_machine.change_state(Event.HANGUP);
        }

        function on_dtmf(event, channel) {
            switch (event.digit) {
            case '#':
                console.log("Accepted recording", call.vm_path);
                cleanup();
                recording.stop(function(err) {
                    call.state_machine.change_state(Event.DTMF_OCTOTHORPE);
                });
                break;
            }
        }
    }
}

module.exports = RecordingState;
```

When entered, the state sets up listeners for hangup and DTMF events on the channel, since those are the events that will cause the state to change. In all cases, before a state change occurs, the `cleanup()` function is invoked to remove event listeners. This way, the event listeners set by the recording state will not accidentally still be set up when the next state is entered. This same `cleanup()` method will be used for all states we create that set up ARI event listeners.

The `stop` method causes a live recording to finish and be saved to the file system. Notice that the `on_hangup()` method does not attempt to stop the live recording. This is because when a channel hangs up, any live recordings on that channel are automatically stopped and stored.

The other two states in the state machine are much simpler, since they are terminal states and do not need to watch for any events.

ending_state.py Expand source

```
class EndingState(object):
    state_name = "ending"

    def __init__(self, call):
        self.call = call

    def enter(self):
        channel_name = self.call.channel.json.get('name')
        print "Ending voice mail call from {0}".format(channel_name)
        self.call.channel.hangup()
```

ending_state.js Expand source

```
function EndingState(call) {
    this.state_name = "ending";

    this.enter = function() {
        channel_name = call.channel.name;
        console.log("Ending voice mail call from", channel_name);
        call.channel.hangup();
    }
}

module.exports = EndingState;
```

hangup_state.py Expand source

```
class HungUpState(object):
    state_name = "hungup"

    def __init__(self, call):
        self.call = call

    def enter(self):
        channel_name = self.call.channel.json.get('name')
        print "Channel {0} hung up".format(channel_name)
```

hungup_state.js Expand source

```
function HungUpState(call) {
    this.state_name = "hungup";

    this.enter = function() {
        channel_name = call.channel.name;
        console.log("Channel %s hung up", channel_name);
    }
}

module.exports = HungUpState;
```

These two states are two sides to the same coin. The `EndingState` is used to end the call by hanging up the channel, and the `HungUpState` is used to terminate the state machine when the caller has hung up. You may find yourself wondering why a `HungUpState` is needed at all. For our application, it does not do much, but it's a great place to perform post-call logic if your application demands it. See the second reader exercise on this page to see an example of that.

Using the application skeleton we set up earlier, we can make the following modifications to accommodate our state machine:

vm-call.py Expand source

```
# At the top of the file
from recording_state import RecordingState
from ending_state import EndingState
from hungup_state import HungUpState

# Inside our VoiceMailCall class
    def setup_state_machine(self):
        hungup_state = HungUpState(self)
        recording_state = RecordingState(self)
        ending_state = EndingState(self)

        self.state_machine = StateMachine()
        self.state_machine.add_transition(recording_state, Event.DTMF_OCTOTHORPE,
                                          ending_state)
        self.state_machine.add_transition(recording_state, Event.HANGUP,
                                          hungup_state)
        self.state_machine.start(recording_state)
```

vm-call.js Expand source

```
// At the top of the file
var RecordingState = require('./recording_state');
var EndingState = require('./ending_state');
var HungUpState = require('./hungup_state');


 // Inside our VoiceMailCall function
 this.setup_state_machine = function() {
        var hungup_state = new HungUpState(self)
        var recording_state = new RecordingState(self)
        var ending_state = new EndingState(self)

        this.state_machine = new StateMachine()
        this.state_machine.add_transition(recording_state, Event.DTMF_OCTOTHORPE,
ending_state)
        this.state_machine.add_transition(recording_state, Event.HANGUP, hungup_state)
        this.state_machine.start(recording_state)
 }
```

The following is a sample output of a user calling the application and pressing the '#' key when finished recording

```
Channel PJSIP/200-00000003 recording voicemail for 305
Entering recording state
Recording voicemail at voicemail/305/1411497846.53
Accepted recording voicemail/305/1411497846.53
Cleaning up event handlers
Ending voice mail call from PJSIP/200-00000003
```

Reader Exercise 1

Currently, the voicemails being recorded are all kept in a single "folder" for a specific mailbox. See if you can change the code to record messages in an "INBOX" folder on the mailbox instead.

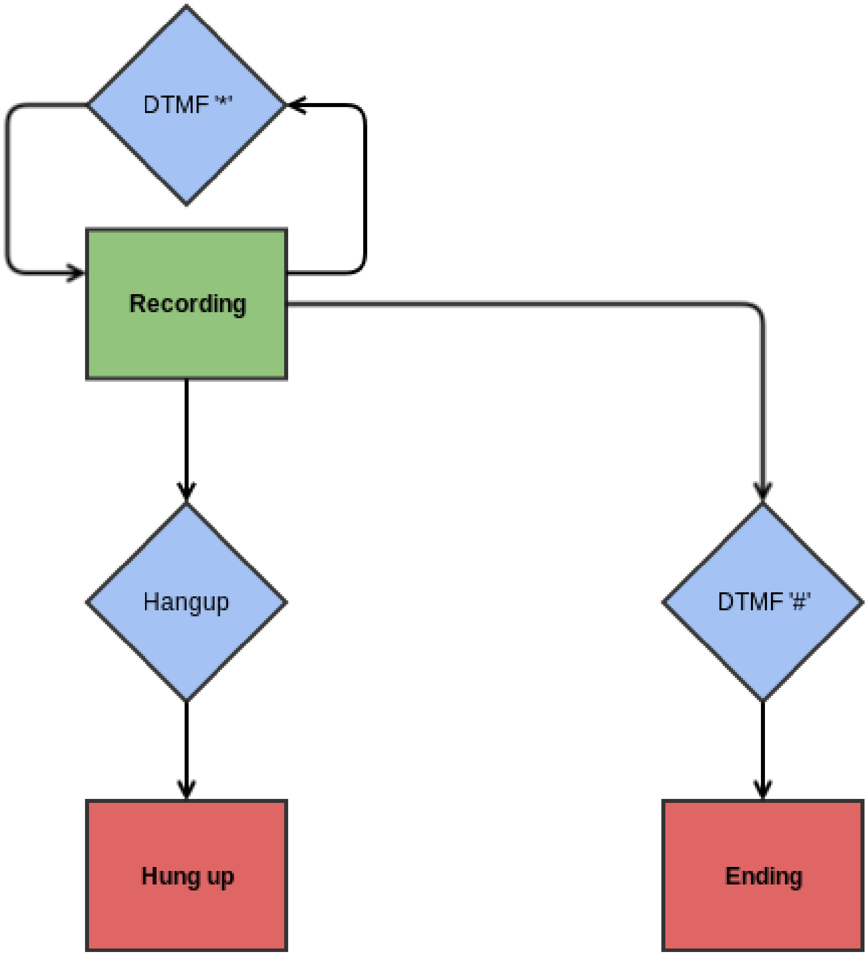
Reader Exercise 2

`EndingState` and `HungUpState` don't do much of anything at the moment. States like these can be great in voice mail applications for updating the message-waiting state of mailboxes on a system. If you're feeling industrious, read the API for the `/mailboxes` resource in ARI. Try to change `HungUpState` and `EndingState` to update the message-waiting status of a mailbox when a new message is left. To keep the exercise simple, for now you can assume the following:

- The number of "old" messages in a mailbox is always 0
- Since you are more concerned with alerting the mailbox owner that there exist new messages, do not try to count the messages in the mailbox. Instead, just state that there is 1 new message.

Cancelling a Recording

Now we have a simple application set up to record a message, but it's pretty bare at the moment. Let's start expanding some. One feature we can add is the ability to press a DTMF key while recording a voice mail to cancel the current recording and re-start the recording process. We'll use the DTMF '*' key to accomplish this. The updated state machine diagram looks like the following:



All that has changed is that there is a new transition, which means a minimal change to our current code to facilitate the change. In our `recording_state` file, we will rewrite the `on_dtmf` method as follows:

recording_state.py Expand source

```
def on_dtmf(self, channel, event):
        digit = event.get('digit')
        if digit == '#':
            rec_name = self.recording.json.get('name')
            print "Accepted recording {0} on DTMF #".format(rec_name)
            self.cleanup()
            self.recording.stop()
            self.call.state_machine.change_state(Event.DTMF_OCTOTHORPE)
        # NEW CONTENT
        elif digit == '*':
            rec_name = self.recording.json.get('name')
            print "Canceling recording {0} on DTMF *".format(rec_name)
            self.cleanup()
            self.recording.cancel()
            self.call.state_machine.change_state(Event.DTMF_STAR)
```

recording_state.js Expand source

```
function on_dtmf(event, channel) {
        switch (event.digit) {
        case '#':
            console.log("Accepted recording", call.vm_path);
            cleanup();
            recording.stop(function(err) {
                call.state_machine.change_state(Event.DTMF_OCTOTHORPE);
            });
            break;
        // NEW CONTENT
        case '*':
            console.log("Canceling recording", call.vm_path);
            cleanup();
            recording.cancel(function(err) {
                call.state_machine.change_state(Event.DTMF_STAR);
            });
            break;
        }
    }
```

The first part of the method is the same as it was before, but we have added extra handling for when the user presses the '*' key. The `cancel()` method for live recordings causes the live recording to be stopped and for it not to be stored on the file system.

We also need to add our new transition while setting up our state machine. Our `VoiceMailCall::setup_state_machine()` method now looks like:

vm-call.py

Expand source

```
def setup_state_machine(self):
        hungup_state = HungUpState(self)
        recording_state = RecordingState(self)
        ending_state = EndingState(self)
        self.state_machine = StateMachine()
        self.state_machine.add_transition(recording_state, Event.DTMF_OCTOTHORPE,
                                          ending_state)
        self.state_machine.add_transition(recording_state, Event.HANGUP,
                                          hungup_state)
        self.state_machine.add_transition(recording_state, Event.DTMF_STAR,
                                          recording_state)
        self.state_machine.start(recording_state)
```

vm-call.js

Expand source

```
this.setup_state_machine = function() {
    var hungup_state = new HungUpState(this);
    var recording_state = new RecordingState(this);
    var ending_state = new EndingState(this);

 this.state_machine = new StateMachine();
    this.state_machine.add_transition(recording_state, Event.DTMF_OCTOTHORPE,
ending_state);
    this.state_machine.add_transition(recording_state, Event.HANGUP, hungup_state);
    this.state_machine.add_transition(recording_state, Event.DTMF_STAR, recording_state);
    this.state_machine.start(recording_state);
}
```

This is exactly the same as it was, except for the penultimate line adding the `Event.DTMF_STAR` transition. Here is sample output for when a user calls in, presses '*' twice, and then presses '#' to complete the call

```
Channel PJSIP/200-00000007 recording voicemail for 305
Entering recording state
Recording voicemail at voicemail/305/1411498790.65
Canceling recording voicemail/305/1411498790.65 on DTMF *
Cleaning up event handlers
Entering recording state
Recording voicemail at voicemail/305/1411498790.65
Canceling recording voicemail/305/1411498790.65 on DTMF *
Cleaning up event handlers
Entering recording state
Recording voicemail at voicemail/305/1411498790.65
Accepted recording voicemail/305/1411498790.65 on DTMF #
Cleaning up event handlers
Ending voice mail call from PJSIP/200-00000007
```

Reader Exercise 3

We have covered the `stop()` and `cancel()` methods, but live recordings provide other methods as well. In particular, there are `pause()`, which causes the live recording to temporarily stop recording audio, and `unpause()`, which causes the live recording to resume recording audio.

Modify the `RecordingState` to allow a DTMF digit of your choice to toggle pausing and unpausing the live recording.

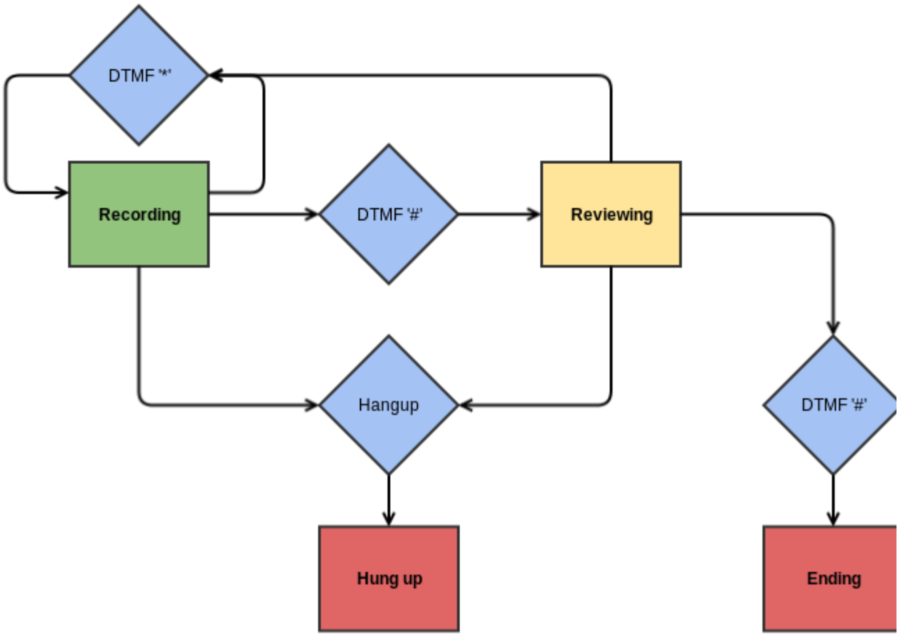
Reader Exercise 4

Our application provides the ability to cancel recordings and re-record them, but it gives no ability to cancel the recording and end the call.

Modify the `RecordingState` to allow for a DTMF digit of your choice to cancel the recording and end the call.

Operating on Stored Recordings

So far, we've recorded a channel, stopped a live recording, and cancelled a live recording. Now let's turn our attention to operations that can be performed on stored recordings. An obvious operation to start with is to play back the stored recording. We're going to make another modification to our voice mail recording application that adds a "reviewing" state after a voicemail is recorded. In this state, a user that has recorded a voice mail will hear the recorded message played back to him/her. The user may press the '#' key or hang up in order to accept the recorded message, or the user may press '*' to erase the stored recording and record a new message in its place. Below is the updated state diagram with the new "reviewing" state added.



To realize this, here is the code for our new "reviewing" state:

reviewing_state.py Expand source

```
import uuid

class ReviewingState(object):
    state_name = "reviewing"

    def __init__(self, call):
        self.call = call
        self.playback_id = None
        self.hangup_event = None
        self.playback_finished = None
        self.dtmf_event = None
        self.playback = None

    def enter(self):
        self.playback_id = str(uuid.uuid4())
        print "Entering reviewing state"
        self.hangup_event = self.call.channel.on_event("ChannelHangupRequest",
                self.on_hangup)
        self.playback_finished = self.call.client.on_event(
                'PlaybackFinished', self.on_playback_finished)
        self.dtmf_event = self.call.channel.on_event('ChannelDtmfReceived',
                                                     self.on_dtmf)
        self.playback = self.call.channel.playWithId(
                playbackId=self.playback_id, media="recording:{0}".format(
                    self.call.vm_path))

    def cleanup(self):
        self.playback_finished.close()
        if self.playback:
            self.playback.stop()
        self.dtmf_event.close()
        self.hangup_event.close()

    def on_hangup(self, channel, event):
        print "Accepted recording {0} on hangup".format(self.call.vm_path)
        self.cleanup()
        self.call.state_machine.change_state(Event.HANGUP)

    def on_playback_finished(self, event):
        if self.playback_id == event.get('playback').get('id'):
            self.playback = None

    def on_dtmf(self, channel, event):
        digit = event.get('digit')
        if digit == '#':
            print "Accepted recording {0} on DTMF #".format(self.call.vm_path)
            self.cleanup()
            self.call.state_machine.change_state(Event.DTMF_OCTOTHORPE)
        elif digit == '*':
            print "Discarding stored recording {0} on DTMF *".format(self.call.vm_path)
            self.cleanup()
            self.call.client.recordings.deleteStored(
                    recordingName=self.call.vm_path)
            self.call.state_machine.change_state(Event.DTMF_STAR)
```

reviewing_state.js

Expand source

```
var Event = require('./event');

function ReviewingState(call) {
    this.state_name = "reviewing";

    this.enter = function() {
        var playback = call.client.Playback();
        var url = "recording:" + call.vm_path;
        console.log("Entering reviewing state");
        call.channel.on("ChannelHangupRequest", on_hangup);
        call.channel.on("ChannelDtmfReceived", on_dtmf);
        call.client.on("PlaybackFinished", on_playback_finished);
        call.channel.play({media: url}, playback);

        function cleanup() {
            call.channel.removeListener('ChannelHangupRequest', on_hangup);
            call.channel.removeListener('ChannelDtmfReceived', on_dtmf);
            call.client.removeListener('PlaybackFinished', on_playback_finished);
            if (playback) {
                playback.stop();
            }
        }

        function on_hangup(event, channel) {
            console.log("Accepted recording %s on hangup", call.vm_path);
            playback = null;
            cleanup();
            call.state_machine.change_state(Event.HANGUP);
        }

        function on_playback_finished(event) {
            if (playback && (playback.id === event.playback.id)) {
                playback = null;
            }
        }

        function on_dtmf(event, channel) {
            switch (event.digit) {
            case '#':
                console.log("Accepted recording", call.vm_path);
                cleanup();
                call.state_machine.change_state(Event.DTMF_OCTOTHORPE);
                break;
            case '*':
                console.log("Canceling recording", call.vm_path);
                cleanup();
                call.client.recordings.deleteStored({recordingName: call.vm_path});
                call.state_machine.change_state(Event.DTMF_STAR);
                break;
            }
        }
    }
}

module.exports = ReviewingState;
```

The code for this state is similar to the code from `RecordingState`. The big difference is that instead of recording a message, it is playing back a stored recording. Stored recordings can be played using the channel's `play()` method (or as we have used in the python code, `playWithId()`). If the URI of the media to be played is prefixed with the "recording:" scheme, then Asterisk knows to search for the specified file where recordings are stored. More information on playing back files on channels, as well as a detailed list of media URI schemes can be found here. Note the method that is called when a DTMF '*' is received. The `deleteStored()` method can be used on the `/recordings` resource of the ARI client to delete a stored recording from the file system on which Asterisk is running.

One more thing to point out is the code that runs in `on_playback_finished()`. When reviewing a voicemail recording, the message may finish playing back before the user decides what to do with it. If this happens, we detect that the playback has finished so that we do not attempt to stop an already-finished playback once the user decides how to proceed.

We need to get this new state added into our state machine, so we make the following modifications to our code to allow for the new state to be added:

vm-call.py Expand source

```
#At the top of the file
from reviewing_state import ReviewingState

#In VoiceMailCall::setup_state_machine
    def setup_state_machine(self):
        hungup_state = HungUpState(self)
        recording_state = RecordingState(self)
        ending_state = EndingState(self)
        reviewing_state = ReviewingState(self)
        self.state_machine = StateMachine()
        self.state_machine.add_transition(recording_state, Event.DTMF_OCTOTHORPE,
                                          reviewing_state)
        self.state_machine.add_transition(recording_state, Event.HANGUP,
                                          hungup_state)
        self.state_machine.add_transition(recording_state, Event.DTMF_STAR,
                                          recording_state)
        self.state_machine.add_transition(reviewing_state, Event.HANGUP,
                                          hungup_state)
        self.state_machine.add_transition(reviewing_state, Event.DTMF_OCTOTHORPE,
                                          ending_state)
        self.state_machine.add_transition(reviewing_state, Event.DTMF_STAR,
                                          recording_state)
        self.state_machine.start(recording_state)
```

vm-call.js Expand source

```
// At the top of the file
var ReviewingState = require('./reviewing_state');

// In VoicemailCall::setup_state_machine
this.setup_state_machine = function() {
    var hungup_state = new HungUpState(this);
    var recording_state = new RecordingState(this);
    var ending_state = new EndingState(this);
    var reviewing_state = new ReviewingState(this);

 this.state_machine = new StateMachine();
    this.state_machine.add_transition(recording_state, Event.DTMF_OCTOTHORPE,
reviewing_state);
    this.state_machine.add_transition(recording_state, Event.HANGUP, hungup_state);
    this.state_machine.add_transition(recording_state, Event.DTMF_STAR, recording_state);
    this.state_machine.add_transition(reviewing_state, Event.DTMF_OCTOTHORPE,
ending_state);
    this.state_machine.add_transition(reviewing_state, Event.HANGUP, hungup_state);
    this.state_machine.add_transition(reviewing_state, Event.DTMF_STAR, recording_state);
    this.state_machine.start(recording_state);
}
```

The following is the output from a sample call. The user records audio, then presses '#'. Upon hearing the recording, the user decides to record again, so the user presses '*'. After re-recording, the user presses '#'. The user hears the new version of the recording played back and is satisfied with it, so the user presses '#' to accept the recording.

```
Channel PJSIP/200-00000009 recording voicemail for 305
Entering recording state
Recording voicemail at voicemail/305/1411501058.42
Accepted recording voicemail/305/1411501058.42 on DTMF #
Cleaning up event handlers
Entering reviewing state
Discarding stored recording voicemail/305/1411501058.42 on DTMF *
Entering recording state
Recording voicemail at voicemail/305/1411501058.42
Accepted recording voicemail/305/1411501058.42 on DTMF #
Cleaning up event handlers
Entering reviewing state
Accepted recording voicemail/305/1411501058.42 on DTMF #
Ending voice mail call from PJSIP/200-00000009
```

Reader Exercise 5

In the previous section we introduced the ability to delete a stored recording. Stored recordings have a second operation available to them: copying. The `copy()` method of a stored recording can be used to copy the stored recording from one location to another.

For this exercise modify `ReviewingState` to let a DTMF key of your choice copy the message to a different mailbox on the system. When a user presses this DTMF key, the state machine should transition into a new state called "copying." The "copying" state should gather DTMF from the user to determine which mailbox the message should be copied to. If '#' is entered, then the message is sent to the mailbox the user has typed in. If '*' is entered, then the copying operation is cancelled. Both a '#' and a '*' should cause the state machine to transition back into `ReviewingState`.

As an example, let's say that you have set DTMF '0' to be the key that the user presses in `ReviewingState` to copy the message. The user presses '0'. The user then presses '3' '2' '0' '#'. The message should be copied to mailbox "320", and the user should start hearing the message played back again. Now let's say the user presses '0' to copy the message again. The user then presses '3' '2' '1' '0' '*'. The message should not be copied to any mailbox, and the user should start hearing the message played back again.

Recording Bridges

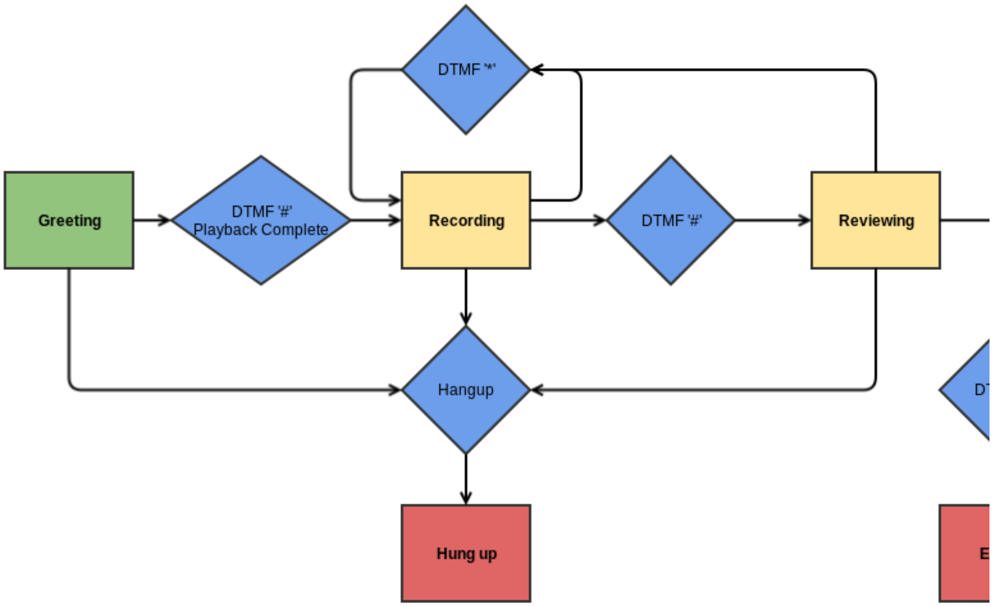
This discussion of recordings has focused on recording channel audio. It's important to note that bridges also have an option to be recorded. What's the difference? Recording a channel's audio records only the audio coming **from** a channel. Recording a bridge records the mixed audio coming from all channels into the bridge. This means that if you are attempting to do something like record a conversation between participants in a phone call, you would want to record the audio in the bridge rather than on either of the channels involved.

Once recording is started on a bridge, the operations available for the live recording and the resulting stored recording are exactly the same as for live recordings and stored recordings on a channel. Since the API for recording a bridge and recording a channel are so similar, this page will not provide any examples of recording bridge audio.

ARI and Media: Part 2 - Playbacks

Querying for sounds

In our voice mail application we have been creating, we have learned the ins and outs of creating and manipulating live and stored recordings. Let's make the voice mail application more user-friendly now by adding some playbacks of installed sounds. The voice mail application has some nice capabilities, but it is not very user-friendly yet. Let's modify the current application to play a greeting to the user when they call into the application. This is the updated state machine:



On this Page

To make the new "greeting" state more interesting, we are going to add some safety to this state by ensuring that the sound we want to play is installed on the system. The `/sounds` resource in ARI provides methods to list the sounds installed on the system, as well as the ability to get specific sound files.

Asterisk searches for sounds in the `/sounds/` subdirectory of the configured `astdatadir` option in `asterisk.conf`. By default, Asterisk will search for sounds in `/var/lib/asterisk/sounds`. When Asterisk starts up, it indexes the installed sounds and keeps an in-data representation of those sound files. When an ARI application asks Asterisk for details about a specific sound or for a list of sounds on the system, Asterisk consults its in-memory index instead of searching the file system directly. This has some trade-offs. When querying for sound information, this in-memory indexing makes the operations much faster. On the other hand, it also means that Asterisk has to be "poked" to re-index the sounds if new sounds are added to the file system after Asterisk is running. The Asterisk CLI command "module reload sounds" provides a means of having Asterisk re-index the sounds on the system so that they are available to ARI.

For our greeting, we will play the built-in sound "vm-intro". Here is the code for our new state:

greeting_state.py

Expand source

```
from event import Event

def sounds_installed(client):
    try:
        client.sounds.get(soundId='vm-intro')
    except:
        print "Required sound 'vm-intro' not installed. Aborting"
        raise


class GreetingState(object):
    state_name = "greeting"

    def __init__(self, call):
        self.call = call
        self.hangup_event = None
        self.playback_finished = None
        self.dtmf_event = None
        self.playback = None
        sounds_installed(call.client)

    def enter(self):
        print "Entering greeting state"
        self.hangup_event = self.call.channel.on_event('ChannelHangupRequest',
                self.on_hangup)
        self.playback_finished = self.call.client.on_event(
                'PlaybackFinished', self.on_playback_finished)
        self.dtmf_event = self.call.channel.on_event('ChannelDtmfReceived',
                                                     self.on_dtmf)
        self.playback = self.call.channel.play(media="sound:vm-intro")

    def cleanup(self):
        self.playback_finished.close()
        self.dtmf_event.close()
        self.hangup_event.close()

    def on_hangup(self, channel, event):
        print "Abandoning voicemail recording on hangup"
        self.cleanup()
        self.call.state_machine.change_state(Event.HANGUP)

    def on_playback_finished(self, playback):
        self.cleanup()
        self.call.state_machine.change_state(Event.PLAYBACK_COMPLETE)

    def on_dtmf(self, channel, event):
        digit = event.get('digit')
        if digit == '#':
            print "Cutting off greeting on DTMF #"
            # Let on_playback_finished take care of state change
            self.playback.stop()
```

greeting_state.js

Expand source

```
var Event = require('./event');

function sounds_installed(client) {
    client.sounds.get({soundId: 'vm-intro'}, function(err) {
        if (err) {
            console.log("Required sound 'vm-intro' not installed. Aborting");
            throw err;
        }
    });
}

function GreetingState(call) {
    this.state_name = "greeting";
    sounds_installed(call.client);

    this.enter = function() {
        var playback = call.client.Playback();
        console.log("Entering greeting state");
        call.channel.on("ChannelHangupRequest", on_hangup);
        call.channel.on("ChannelDtmfReceived", on_dtmf);
        call.client.on("PlaybackFinished", on_playback_finished);
        call.channel.play({media: 'sound:vm-intro'}, playback);

        function cleanup() {
            call.channel.removeListener('ChannelHangupRequest', on_hangup);
            call.channel.removeListener('ChannelDtmfReceived', on_dtmf);
            call.client.removeListener('PlaybackFinished', on_playback_finished);
            if (playback) {
                playback.stop();
            }
        }

        function on_hangup(event, channel) {
            console.log("Abandoning voicemail recording on hangup");
            cleanup();
            call.state_machine.change_state(Event.HANGUP);
        }

        function on_playback_finished(event) {
            if (playback && playback.id === event.playback.id) {
                cleanup();
                call.state_machine.change_state(Event.PLAYBACK_COMPLETE);
            }
        }

        function on_dtmf(event, channel) {
            switch (event.digit) {
            case '#':
                console.log("Skipping greeting");
                cleanup();
                call.state_machine.change_state(Event.DTMF_OCTOTHORPE);
            }
        }
    }
}
 module.exports = GreetingState;
```

The `sounds.get()` method employed here allows for a single sound to be retrieved based on input parameters. Here, we simply specify the name of the recording we want to ensure that it exists in some form on the system. By checking for the sound's existence in the initialization of `GreetingState`, we can abort the call early if the sound is not installed.

And here is our updated state machine:

vm-call.py

Expand source

```
#At the top of the file
from greeting_state import GreetingState

#In VoiceMailCall::setup_state_machine()
    def setup_state_machine(self):
        hungup_state = HungUpState(self)
        recording_state = RecordingState(self)
        ending_state = EndingState(self)
        reviewing_state = ReviewingState(self)
        greeting_state = GreetingState(self)
        self.state_machine = StateMachine()
        self.state_machine.add_transition(recording_state, Event.DTMF_OCTOTHORPE,
                                          reviewing_state)
        self.state_machine.add_transition(recording_state, Event.HANGUP,
                                          hungup_state)
        self.state_machine.add_transition(recording_state, Event.DTMF_STAR,
                                          recording_state)
        self.state_machine.add_transition(reviewing_state, Event.HANGUP,
                                          hungup_state)
        self.state_machine.add_transition(reviewing_state, Event.DTMF_OCTOTHORPE,
                                          ending_state)
        self.state_machine.add_transition(reviewing_state, Event.DTMF_STAR,
                                          recording_state)
        self.state_machine.add_transition(greeting_state, Event.HANGUP,
                                          hungup_state)
        self.state_machine.add_transition(greeting_state,
                                          Event.PLAYBACK_COMPLETE,
                                          recording_state)
        self.state_machine.start(greeting_state)
```

vm-call.js Expand source

```
//At the top of the file
var GreetingState = require('./greeting_state');

//In VoicemailCall::setup_state_machine()
this.setup_state_machine = function() {
    var hungup_state = new HungUpState(this);
    var recording_state = new RecordingState(this);
    var ending_state = new EndingState(this);
    var reviewing_state = new ReviewingState(this);
    var greeting_state = new GreetingState(this);
    this.state_machine = new StateMachine();
    this.state_machine.add_transition(recording_state, Event.DTMF_OCTOTHORPE,
reviewing_state);
    this.state_machine.add_transition(recording_state, Event.HANGUP, hungup_state);
    this.state_machine.add_transition(recording_state, Event.DTMF_STAR, recording_state);
    this.state_machine.add_transition(reviewing_state, Event.DTMF_OCTOTHORPE,
ending_state);
    this.state_machine.add_transition(reviewing_state, Event.HANGUP, hungup_state);
    this.state_machine.add_transition(reviewing_state, Event.DTMF_STAR, recording_state);
    this.state_machine.add_transition(greeting_state, Event.HANGUP, hungup_state);
    this.state_machine.add_transition(greeting_state, Event.PLAYBACK_COMPLETE,
recording_state);
    this.state_machine.start(greeting_state);
}
```

Here is a sample run where the user cuts off the greeting by pressing the '#' key, records a greeting and presses the '#' key, and after listening to the recording presses the '#' key once more.

```
Channel PJSIP/200-0000000b recording voicemail for 305
Entering greeting state
Cutting off greeting on DTMF #
Entering recording state
Recording voicemail at voicemail/305/1411503204.75
Accepted recording voicemail/305/1411503204.75 on DTMF #
Cleaning up event handlers
Entering reviewing state
Accepted recording voicemail/305/1411503204.75 on DTMF #
Ending voice mail call from PJSIP/200-0000000b
```

Reader Exercise 1

Our current implementation of `GreetingState` does not take language into consideration. The `sounds_installed` method checks for the existence of the sound file, but it does not ensure that we have the sound file in the language of the channel that is in our application.

For this exercise, modify `sounds_installed()` to also check if the retrieved sound exists in the language of the calling channel. The channel's language can be retrieved using the `getChannelVar()` method on a channel to retrieve the value of variable "CHANNEL(language)". The sound returned by `sounds.get()` contains an array of `FormatLang` objects that are a pair of format and language strings. If the sound exists, but not in the channel's language, then throw an exception.

Controlling playbacks

So far in our voice mail application, we have stopped playbacks, but there are a lot more interesting operations that can be done on them, such as reversing and fast-forwarding them. Within the context of recording a voicemail, these operations are pretty useless, so we will shift our focus now to the other side of voicemail: listening to recorded voicemails.

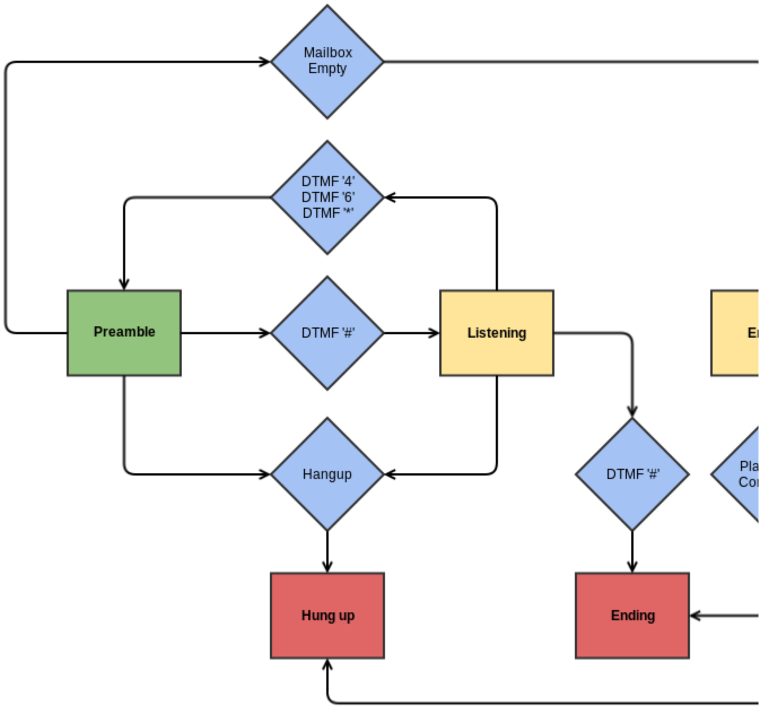
For this, we will write a new application. This new application will allow a caller to listen to the voicemails that are stored in a specific mailbox. When the caller enters the application, a prompt is played to the caller saying which message number the caller is hearing. When the message number finishes playing (or if the caller interrupts the playback with '#'), then the caller hears the specified message in the voicemail box. While listening to the voicemail, the caller can do several things:

- Press the '1' key to go back 3 seconds in the current message playback.
- Press the '2' key to pause or unpause the current message playback.
- Press the '3' key to go forward 3 seconds in the current message playback.
- Press the '4' key to play to the previous message.

- Press the '5' key to restart the current message playback.
- Press the '6' key to play to the next message.
- Press the '*' key to delete the current message and play the next message.
- Press the '#' key to end the call.

If all messages in a mailbox are deleted or if the mailbox contained no messages to begin with, then "no more messages" is played back to the user, and the call is completed.

This means defining a brand new state machine. To start with, we'll define three new states. The "preamble" state is the initial state of the state machine, where the current message number is played back to the listener. The "listening" state is where the voice mail message is played back to the listener. The "empty" state is where no more messages remain in the mailbox. Here is the state machine we will be using:



Notice that while in the listening state, DMTF '4', '6', and '*' all change the state to the preamble state. This is so that the new message number can be played back to the caller before the next message is heard. Also notice that the preamble state is responsible for determining if the state should change to empty. This keeps the logic in the listening state more straight-forward since it is already having to deal with a lot of DTMF events. It also gracefully handles the case where a caller calls into the application when the caller has no voicemail messages.

Here is the implementation of the application.

vm-playback.py

Expand source

```
#!/usr/bin/env python

import ari
import logging
import sys

from state_machine import StateMachine
from ending_state import EndingState
from hungup_state import HungUpState
from listening_state import ListeningState
from preamble_state import PreambleState
from empty_state import EmptyState
from event import Event

logging.basicConfig(level=logging.ERROR)
LOGGER = logging.getLogger(__name__)

client = ari.connect('http://localhost:8088', 'asterisk', 'asterisk')
```

```
class VoiceMailCall(object):
    def __init__(self, ari_client, channel, mailbox):
        self.client = ari_client
        self.channel = channel

        self.voicemails = []
        recordings = ari_client.recordings.listStored()
        vm_number = 1
        for rec in recordings:
            if rec.json['name'].startswith('voicemail/{0}'.format(mailbox)):
                self.voicemails.append((vm_number, rec.json['name']))
                vm_number += 1

        self.current_voicemail = 0
        self.setup_state_machine()

    def setup_state_machine(self):
        hungup_state = HungUpState(self)
        ending_state = EndingState(self)
        listening_state = ListeningState(self)
        preamble_state = PreambleState(self)
        empty_state = EmptyState(self)
        self.state_machine = StateMachine()
        self.state_machine.add_transition(listening_state, Event.DTMF_4,
                                          preamble_state)
        self.state_machine.add_transition(listening_state, Event.DTMF_6,
                                          preamble_state)
        self.state_machine.add_transition(listening_state, Event.HANGUP,
                                          hungup_state)
        self.state_machine.add_transition(listening_state, Event.DTMF_OCTOTHORPE,
                                          ending_state)
        self.state_machine.add_transition(listening_state, Event.DTMF_STAR,
                                          preamble_state)
        self.state_machine.add_transition(preamble_state, Event.DTMF_OCTOTHORPE,
                                          listening_state)
        self.state_machine.add_transition(preamble_state,
                                          Event.PLAYBACK_COMPLETE,
                                          listening_state)
        self.state_machine.add_transition(preamble_state, Event.MAILBOX_EMPTY,
                                          empty_state)
        self.state_machine.add_transition(preamble_state, Event.HANGUP,
                                          hungup_state)
        self.state_machine.add_transition(empty_state, Event.HANGUP,
                                          hungup_state)
        self.state_machine.add_transition(empty_state,
                                          Event.PLAYBACK_COMPLETE,
                                          ending_state)
        self.state_machine.start(preamble_state)

    def next_message(self):
        self.current_voicemail += 1
        if self.current_voicemail == len(self.voicemails):
            self.current_voicemail = 0

    def previous_message(self):
        self.current_voicemail -= 1
        if self.current_voicemail < 0:
            self.current_voicemail = len(self.voicemails) - 1

    def delete_message(self):
        del self.voicemails[self.current_voicemail]
        if self.current_voicemail == len(self.voicemails):
            self.current_voicemail = 0

    def get_current_voicemail_number(self):
        return self.voicemails[self.current_voicemail][0]

    def get_current_voicemail_file(self):
        return self.voicemails[self.current_voicemail][1]

    def mailbox_empty(self):
        return len(self.voicemails) == 0


def stasis_start_cb(channel_obj, event):
    channel = channel_obj['channel']
    channel_name = channel.json.get('name')
    mailbox = event.get('args')[0]
    print("Channel {0} recording voicemail for {1}".format(
        channel_name, mailbox))
    channel.answer()
    VoiceMailCall(client, channel, mailbox)
```

```
client.on_channel_event('StasisStart', stasis_start_cb)
client.run(apps=sys.argv[1])
```

vm-playback.js

Expand source

```
/*jshint node:true*/
'use strict';

var ari = require('ari-client');
var util = require('util');
var path = require('path');

var Event = require('./event');
var StateMachine = require('./state_machine');
var EndingState = require('./ending_state');
var HungUpState = require('./hungup_state');
var PreambleState = require('./preamble_state');
var EmptyState = require('./empty_state');
var ListeningState = require('./listening_state');

ari.connect('http://localhost:8088', 'asterisk', 'asterisk', clientLoaded);

var VoiceMailCall = function(ari_client, channel, mailbox) {
    this.client = ari_client;
    this.channel = channel;
    var current_voicemail = 0;
    var voicemails = [];

    this.setup_state_machine = function() {
        var hungup_state = new HungUpState(this);
        var ending_state = new EndingState(this);
        var listening_state = new ListeningState(this);
        var preamble_state = new PreambleState(this);
        var empty_state = new EmptyState(this);
        this.state_machine = new StateMachine(this);
        this.state_machine.add_transition(listening_state, Event.DTMF_4, preamble_state);
        this.state_machine.add_transition(listening_state, Event.DTMF_6, preamble_state);
        this.state_machine.add_transition(listening_state, Event.HANGUP, hungup_state);
        this.state_machine.add_transition(listening_state, Event.DTMF_OCTOTHORPE,
ending_state);
        this.state_machine.add_transition(listening_state, Event.DTMF_STAR,
preamble_state);
        this.state_machine.add_transition(preamble_state, Event.DTMF_OCTOTHORPE,
listening_state);
        this.state_machine.add_transition(preamble_state, Event.PLAYBACK_COMPLETE,
listening_state);
        this.state_machine.add_transition(preamble_state, Event.MAILBOX_EMPTY,
empty_state);
        this.state_machine.add_transition(preamble_state, Event.HANGUP, hungup_state);
        this.state_machine.add_transition(empty_state, Event.HANGUP, hungup_state);
        this.state_machine.add_transition(empty_state, Event.PLAYBACK_COMPLETE,
ending_state);
        this.state_machine.start(preamble_state);
    }

    this.next_message = function() {
        current_voicemail++;
        if (current_voicemail === voicemails.length) {
            current_voicemail = 0;
        }
    }

    this.previous_message = function() {
        current_voicemail--;
        if (current_voicemail < 0) {
            current_voicemail = voicemails.length - 1;
        }
    }

    this.delete_message = function() {
        voicemails.splice(current_voicemail, 1);
        if (current_voicemail === voicemails.length) {
            current_voicemail = 0;
        }
    }

    this.get_current_voicemail_file = function() {
        return voicemails[current_voicemail]['file'];
    }

    this.mailbox_empty = function() {
        return voicemails.length === 0;
    }

    var self = this;
    ari_client.recordings.listStored(function (err, recordings) {
        var vm_number = 1;
        var regex = new RegExp('^voicemail/' + mailbox);
        for (var i = 0; i < recordings.length; i++) {
            var rec_name = recordings[i].name;
            if (rec_name.search(regex) != -1) {
                console.log("Found voicemail", rec_name);
                voicemails.push({'number': vm_number, 'file': rec_name});
                vm_number++;
            }
        }
        self.setup_state_machine();
    });
}

function clientLoaded(err, client) {
    if (err) {
        throw err;
    }
    client.on('StasisStart', stasisStart);
    function stasisStart(event, channel) {
        var mailbox = event.args[0]
        channel.answer(function(err) {
            if (err) {
                throw err;
            }
            new VoiceMailCall(client, channel, mailbox);
        });
    }
    client.start(process.argv[2]);
}
```

Quite a bit of this is similar to what we were using for our voice mail recording application. The biggest difference here is that the call has many more methods defined since playing back voice mails is more complicated than recording a single one.

Now that we have the state machine defined and the application written, let's actually write the required new states. First of the new states is the "preamble" state.

preamble_state.py

Expand source

```
from event import Event
import uuid

def sounds_installed(client):
    try:
        client.sounds.get(soundId='vm-message')
    except:
        print "Required sound 'vm-message' not installed. Aborting"
        raise


class PreambleState(object):
    state_name = "preamble"

    def __init__(self, call):
        self.call = call
        self.hangup_event = None
        self.playback_finished = None
        self.dtmf_event = None
        self.playback = None
        sounds_installed(call.client)

    def enter(self):
        print "Entering preamble state"
        if self.call.mailbox_empty():
            self.call.state_machine.change_state(Event.MAILBOX_EMPTY)
            return
        self.hangup_event = self.call.channel.on_event("ChannelHangupRequest",
                                                       self.on_hangup)
        self.playback_finished = self.call.client.on_event(
                'PlaybackFinished', self.on_playback_finished)
        self.dtmf_event = self.call.channel.on_event('ChannelDtmfReceived',
                                                     self.on_dtmf)
        self.initialize_playbacks()

    def initialize_playbacks(self):
        self.current_playback = 0
        current_voicemail = self.call.get_current_voicemail_number()
        self.sounds_to_play = [
            {
                'id': str(uuid.uuid4()),
                'media': 'sound:vm-message'
            },
            {
                'id': str(uuid.uuid4()),
                'media': 'number:{0}'.format(current_voicemail)
            }
        ]
        self.start_playback()

    def start_playback(self):
        current_sound = self.sounds_to_play[self.current_playback]
        self.playback = self.call.channel.playWithId(
                playbackId=current_sound['id'],
                media=current_sound['media']
        )

    def cleanup(self):
        self.playback_finished.close()
        if self.playback:
            self.playback.stop()
        self.dtmf_event.close()
        self.hangup_event.close()

    def on_hangup(self, channel, event):
        self.playback = None
        self.cleanup()
        self.call.state_machine.change_state(Event.HANGUP)

    def on_playback_finished(self, event):
        current_sound = self.sounds_to_play[self.current_playback]
        if current_sound['id'] == event.get('playback').get('id'):
            self.playback = None
            self.current_playback += 1
            if self.current_playback == len(self.sounds_to_play):
                self.cleanup()
                self.call.state_machine.change_state(Event.PLAYBACK_COMPLETE)
            else:
                self.start_playback()

    def on_dtmf(self, channel, event):
        digit = event.get('digit')
        if digit == '#':
            self.cleanup()
            self.call.state_machine.change_state(Event.DTMF_OCTOTHORPE)
```

preamble_state.js

Expand source

```
var Event = require('./event');

function sounds_installed(client) {
    client.sounds.get({soundId: 'vm-message'}, function(err) {
        if (err) {
            console.log("Required sound 'vm-message' not installed. Aborting");
            throw err;
        }
    });
}

function PreambleState(call) {
    this.state_name = "preamble";
    this.enter = function() {
        var current_playback;
        var sounds_to_play;
        var playback;
        console.log("Entering preamble state");
        if (call.mailbox_empty()) {
            call.state_machine.change_state(Event.MAILBOX_EMPTY);
            return;
        }
        call.channel.on("ChannelHangupRequest", on_hangup);
        call.client.on("PlaybackFinished", on_playback_finished);
        call.channel.on("ChannelDtmfReceived", on_dtmf);
        initialize_playbacks();

        function initialize_playbacks() {
            current_playback = 0;
            sounds_to_play = [
                {
                    'playback': call.client.Playback(),
                    'media': 'sound:vm-message'
                },
                {
                    'playback': call.client.Playback(),
                    'media': 'number:' + call.get_current_voicemail_number()
                }
            ];
            start_playback();
        }

        function start_playback() {
            current_sound = sounds_to_play[current_playback];
            playback = current_sound['playback'];
            call.channel.play({media: current_sound['media']}, playback);
        }

        function cleanup() {
            call.channel.removeListener('ChannelHangupRequest', on_hangup);
            call.channel.removeListener('ChannelDtmfReceived', on_dtmf);
            call.client.removeListener('PlaybackFinished', on_playback_finished);
            if (playback) {
                playback.stop();
            }
        }

        function on_hangup(event, channel) {
            playback = null;
            cleanup();
            call.state_machine.change_state(Event.HANGUP);
        }

        function on_playback_finished(event) {
            var current_sound = sounds_to_play[current_playback];
            if (playback && (playback.id === event.playback.id)) {
                playback = null;
                current_playback++;
                if (current_playback === sounds_to_play.length) {
                    cleanup();
                    call.state_machine.change_state(Event.PLAYBACK_COMPLETE);
                } else {
                    start_playback();
                }
            }
        }

        function on_dtmf(event, channel) {
            switch(event.digit) {
            case '#':
                cleanup();
                call.state_machine.change_state(Event.DTMF_OCTOTHORPE);
            }
        }
    }
}
```

```
module.exports = PreambleState;
```

`PreambleState` should look similar to the `GreetingState` introduced previously on this page. The biggest difference is that the code is structured to play multiple sound files instead of just a single one. Note that it is acceptable to call `channel.play()` while a playback is playing on a channel in order to queue a second playback. For our application though, we have elected to play the second sound only after the first has completed. The reason for this is that if there is only a single active playback at any given time, then it becomes easier to clean up the current state when an event occurs that causes a state change.

Next, here is the "empty" state code:

empty_state.py

Expand source

```
from event import Event
import uuid

def sounds_installed(client):
    try:
        client.sounds.get(soundId='vm-nomore')
    except:
        print "Required sound 'vm-nomore' not installed. Aborting"
        raise


class EmptyState(object):
    state_name = "empty"

    def __init__(self, call):
        self.call = call
        self.playback_id = None
        self.hangup_event = None
        self.playback_finished = None
        self.playback = None
        sounds_installed(call.client)

    def enter(self):
        self.playback_id = str(uuid.uuid4())
        print "Entering empty state"
        self.hangup_event = self.call.channel.on_event("ChannelHangupRequest",
                self.on_hangup)
        self.playback_finished = self.call.client.on_event(
                'PlaybackFinished', self.on_playback_finished)
        self.playback = self.call.channel.playWithId(
                playbackId=self.playback_id, media="sound:vm-nomore")

    def cleanup(self):
        self.playback_finished.close()
        if self.playback:
            self.playback.stop()
        self.hangup_event.close()

    def on_hangup(self, channel, event):
        # Setting playback to None stops cleanup() from trying to stop the
        # playback.
        self.playback = None
        self.cleanup()
        self.call.state_machine.change_state(Event.HANGUP)

    def on_playback_finished(self, event):
        if self.playback_id == event.get('playback').get('id'):
            self.playback = None
            self.cleanup()
            self.call.state_machine.change_state(Event.PLAYBACK_COMPLETE)
```

empty_state.js

Expand source

```
var Event = require('./event');

function sounds_installed(client) {
    client.sounds.get({soundId: 'vm-nomore'}, function(err) {
        if (err) {
            console.log("Required sound 'vm-nomore' not installed. Aborting");
            throw err;
        }
    });
}

function EmptyState(call) {
    this.state_name = "empty";

    this.enter = function() {
        console.log("Entering empty state");
        playback = call.client.Playback();
        call.channel.on("ChannelHangup", on_hangup);
        call.client.on("PlaybackFinished", on_playback_finished);
        call.channel.play({media: 'sound:vm-nomore'}, playback);

        function cleanup() {
            call.channel.removeListener('ChannelHangupRequest', on_hangup);
            call.channel.removeListener('PlaybackFinished', on_playback_finished);
            if (playback) {
                playback.stop();
            }
        }

        function on_hangup(event, channel) {
            playback = null;
            cleanup();
            call.state_machine.change_state(Event.HANGUP);
        }

        function on_playback_finished(event) {
            if (playback && playback.id === event.playback.id) {
                playback = null;
                cleanup();
                call.state_machine.change_state(Event.PLAYBACK_COMPLETE);
            }
        }
    }
}

module.exports = EmptyState;
```

This state does not introduce anything we haven't seen already, so let's move on to the "listening" state code:

listening_state.py

Expand source

```
from event import Event
import uuid

class ListeningState(object):
    state_name = "listening"
```

```
def __init__(self, call):
    self.call = call
    self.playback_id = None
    self.hangup_event = None
    self.playback_finished = None
    self.dtmf_event = None
    self.playback = None

def enter(self):
    self.paused = False
    self.playback_id = str(uuid.uuid4())
    print "Entering listening state"
    self.hangup_event = self.call.channel.on_event("ChannelHangupRequest",
            self.on_hangup)
    self.playback_finished = self.call.client.on_event(
            'PlaybackFinished', self.on_playback_finished)
    self.dtmf_event = self.call.channel.on_event('ChannelDtmfReceived',
                                                 self.on_dtmf)
    self.playback = self.call.channel.playWithId(
            playbackId=self.playback_id, media="recording:{0}".format(
                self.call.get_current_voicemail_file()))

def cleanup(self):
    self.playback_finished.close()
    if self.playback:
        self.playback.stop()
    self.dtmf_event.close()
    self.hangup_event.close()

def on_hangup(self, channel, event):
    self.cleanup()
    self.call.state_machine.change_state(Event.HANGUP)

def on_playback_finished(self, event):
    if self.playback_id == event.get('playback').get('id'):
        self.playback = None

def on_dtmf(self, channel, event):
    digit = event.get('digit')
    if digit == '1':
        if self.playback:
            self.playback.control(operation='reverse')
    elif digit == '2':
        if not self.playback:
            return
        if self.paused:
            self.playback.control(operation='unpause')
            self.paused = False
        else:
            self.playback.control(operation='pause')
            self.paused = True
    elif digit == '3':
        if self.playback:
            self.playback.control(operation='forward')
    elif digit == '4':
        self.cleanup()
        self.call.previous_message()
        self.call.state_machine.change_state(Event.DTMF_4)
    elif digit == '5':
        if self.playback:
            self.playback.control(operation='restart')
    elif digit == '6':
        self.cleanup()
        self.call.next_message()
        self.call.state_machine.change_state(Event.DTMF_6)
    elif digit == '#':
        self.cleanup()
        self.call.state_machine.change_state(Event.DTMF_OCTOTHORPE)
    elif digit == '*':
        print ("Deleting stored recording {0}".format(
            self.call.get_current_voicemail_file()))
        self.cleanup()
        self.call.client.recordings.deleteStored(
                recordingName=self.call.get_current_voicemail_file())
        self.call.delete_message()
        self.call.state_machine.change_state(Event.DTMF_STAR)
```

listening_state.js

Expand source

```
var Event = require('./event');

var ListeningState = function(call) {
   this.state_name = "listening";
   this.call = call;
}

function ListeningState(call) {
    this.state_name = "listening";
    this.enter = function() {
        var playback = call.client.Playback();
        var url = "recording:" + call.get_current_voicemail_file();

        console.log("Entering Listening state");
        call.channel.on("ChannelHangupRequest", on_hangup);
        call.channel.on("ChannelDtmfReceived", on_dtmf);
        call.client.on("PlaybackFinished", on_playback_finished);
        console.log("Playing file %s", url);
        call.channel.play({media: url}, playback, function(err) {
            if (err) {
                console.error(err);
            }
        });

        function cleanup() {
            call.channel.removeListener('ChannelHangupRequest', on_hangup);
            call.channel.removeListener('ChannelDtmfReceived', on_dtmf);
            call.client.removeListener('PlaybackFinished', on_playback_finished);
            if (playback) {
                playback.stop();
            }
        }

        function on_hangup(event, channel) {
            playback = null;
            cleanup();
            call.state_machine.change_state(Event.HANGUP);
        }

        function on_playback_finished(event) {
            if (playback && (playback.id === event.playback.id)) {
                playback = null;
            }
        }

        function on_dtmf(event, channel) {
            switch (event.digit) {
            case '1':
                if (playback) {
                    playback.control({operation: 'reverse'});
                }
                break;
            case '2':
                if (!playback) {
                    break;
                }
                if (paused) {
                    playback.control({operation: 'unpause'});
                    paused = false;
                } else {
                    playback.control({operation: 'pause'});
                    paused = true;
                }
                break;
            case '3':
                if (playback) {
                    playback.control({operation: 'forward'});
                }
                break;
            case '4':
                cleanup();
                call.previous_message();
                call.state_machine.change_state(Event.DTMF_4);
                break;
            case '5':
                if (playback) {
                    playback.control({operation: 'restart'});
                }
                break;
            case '6':
                cleanup();
                call.next_message();
                call.state_machine.change_state(Event.DTMF_6);
                break;
            case '#':
                cleanup();
                call.state_machine.change_state(Event.DTMF_OCTOTHORPE);
                break;
            case '*':
                console.log("Deleting stored recording",
call.get_current_voicemail_file());
                cleanup();

call.client.recordings.deleteStored({recordingName:call.get_current_voicemail_file()});
                call.delete_message();
                call.state_machine.change_state(Event.DTMF_STAR);
            }
        }
    }
```

```
}

module.exports = ListeningState;
```

`ListeningState` is where we introduce new playback control concepts. Playbacks have their controlling operations wrapped in a single method, `control()`, rather than having lots of separate operations. All control operations (reverse, pause, unpause, forward, and restart) are demonstrated by this state.

Reader Exercise 2

You may have noticed while exploring the playbacks API that the `control()` method takes no parameters other than the operation. This means that certain properties of operations are determined when the playback is started on the channel.

For this exercise, modify the `ListeningState` so that DTMF '1' and '3' reverse or forward the playback by 5000 milliseconds instead of the default 3000 milliseconds.

Playbacks on bridges

Just as channels allow for playbacks to be performed on them, bridges also have the capacity to have sounds, recordings, tones, numbers, etc. played back on them. The difference is is that all participants in the bridge will hear the playback instead of just a single channel. In bridging situations, it can be useful to play certain sounds to an entire bridge (e.g. Telling participants the call is being recorded), but it can also be useful to play sounds to specific participants (e.g. Telling a caller he has joined a conference bridge). A playback on a bridge can be stopped or controlled exactly the same as a playback on a channel.

Reader Exercise 3

If you've read through the Recording and Playbacks pages, then you should have a good grasp on the operations available, as well as a decent state machine implementation. For our final exercise, instead of adding onto the existing voice mail applications, create a new application that uses some of the recording and playback operations that you have learned about.

You will be creating a rudimentary call queue application. The queue application will have two types of participants: agents and callers. Agents and callers call into the same Stasis application and are distinguished based on arguments to the Stasis application (e.g. A caller might call Stasis(queue,caller) and an agent might call Stasis(queue,agent) from the dialplan).

- When an agent calls into the Stasis application, the agent is placed into an agent queue.
  - While in the agent queue, the agent should hear music on hold. Information about how to play music on hold to a channel can be found here.
- When a caller calls into the Stasis application, the caller is placed into a caller queue.
  - While in the caller queue, the caller should hear music on hold.
  - Every 30 seconds, the caller should hear an announcement. If the caller is at the front of the queue, the "queue-youarenext" sound should play. If the caller is not at the front of the queue, then the caller should hear the "queue-thereare" sound, then the number of callers ahead in the queue, then the "queue-callswaiting" sound.
- If there is a caller in the caller queue and an agent in the agent queue, then the caller and agent at the front of their respective queues are removed and placed into a bridge with each other. For more information on bridging channels, see this page. If this happens while a caller has a sound file playing, then this should cause the sound file to immediately stop playing in order to bridge the call.
- Once bridged, the agent can perform the following:
  - Pressing '0' should start recording the bridge. Upon recording the bridge, the caller should hear a beep sound.
  - Pressing '5' while the call is being recorded should pause the recording. Pressing '5' a second time should unpause the recording.
  - Pressing '7' while the call is being recorded should stop the recording.
  - Pressing '9' while the call is being recorded should discard the recording.
  - Pressing '#' at any time (whether the call is being recorded or not) should end the call, causing the agent to be placed into the back of the agent queue and the caller to be hung up.
  - If the agent hangs up, then the caller should also be hung up.
- Once bridged, if the caller hangs up, then the agent should be placed into the back of the agent queue.

Best of luck!

The Asterisk Resource

The Asterisk Resource

While the primary purpose of ARI is to allow developers to build their own communications applications using Asterisk as a media engine, there are other resources in the API that are useful outside of this use case. One of these is the `asterisk` resource. This resource not only provides information about the running Asterisk instance, but also exposes resources and operations that allow an external system to manipulate the overall Asterisk system.

On This Page

More Detail

- ARI Push Configuration

Retrieving System Information

The `asterisk` resource provides the ability to retrieve basic information about the running Asterisk process. This includes:

- Information about how Asterisk was compiled
- Information about Asterisk's configuration
- Current status of the Asterisk process
- Information about the system the Asterisk process is running on

An example of this is shown below:

```
$ curl -X GET -u asterisk:SECRET https://localhost:8088/ari/asterisk/info

{
  "status":
  {
    "startup_time": "2015-07-16T21:01:37.273-0500",
    "last_reload_time": "2015-07-16T21:01:37.273-0500"
  },
  "build":
  {
    "user": "mjordan",
    "options": "AST_DEVMODE, LOADABLE_MODULES, OPTIONAL_API, TEST_FRAMEWORK",
    "machine": "x86_64",
    "os": "Linux",
    "kernel": "3.13.0-24-generic",
    "date": "2015-07-11 15:51:57 UTC"
  },
  "system":
  {
    "version": "GIT-master-0b2cbeaM",
    "entity_id": "ec:f4:bb:67:a6:d0"
  },
  "config":
  {
    "default_language": "en",
    "name":"mjordan-laptop",
    "setid":
    {
      "user": "asterisk",
      "group": "asterisk"
    }
  }
}
```

ARI Push Configuration

Overview

Asterisk typically retrieves its configuration information by *pulling* it from some configuration source - whether that be a static configuration file or a relational database. This page describes an alternative way to provide configuration information to Asterisk using a *push* model through ARI. Note that only modules whose configuration is managed by the Sorcery data abstraction framework in Asterisk can make use of this mechanism. Predominately, this implies configuration of the PJSIP stack.

On This Page

Version Information

This feature was introduced in ARI version 1.8.0, or Asterisk 13.5.0 or later.

Push Configuration Workflow

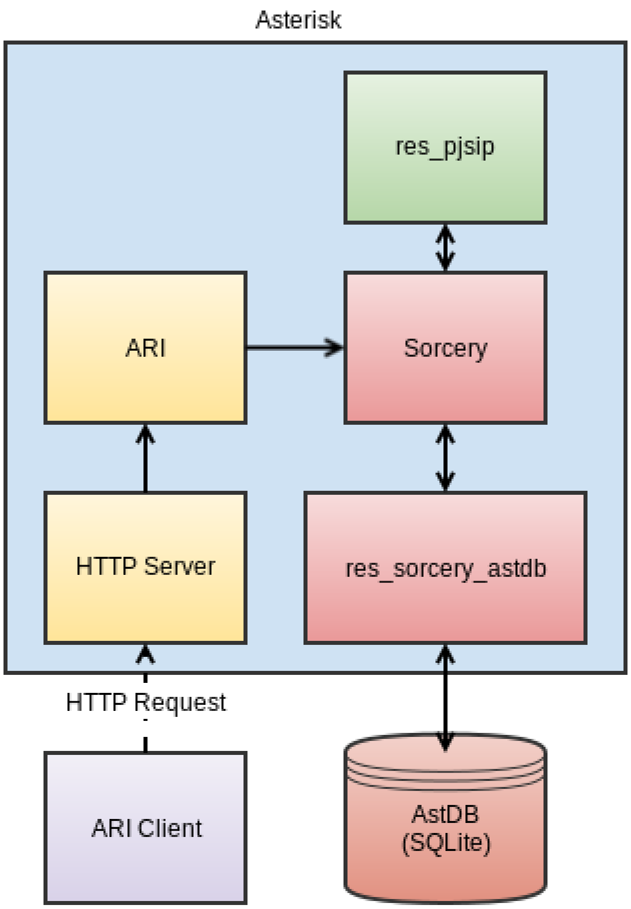
With push configuration, an external process uses ARI to perform a configuration operation. The configuration operation could be any one of the four classic operations for persistent storage - Create, Retrieve, Update, or Delete. For the purposes of this workflow, we'll assume that the operation is to create a configuration change in Asterisk.

The ARI client makes a PUT request, where the body contains the configuration object to create, encoded in JSON. This is first handled by ARI, which performs basic validation on the inbound request and its contents. Once the request is validated, ARI asks the Sorcery framework to create the actual object.

Sorcery requires three pieces of information, at a minimum, to create an object:

- The overall class of configuration types. This is usually a module or namespace that provides multiple types of objects to be created, e.g., 'res_pjsip'.
- The type of configuration object to create, e.g., 'endpoint'.
- A unique identifier (amongst objects of the same type) for the object, e.g., 'alice'.

Once Sorcery has determined that it knows how to create the type of object, it creates it using the provided data in the JSON body. If some piece of data in the body can't be converted to an attribute on the object, the inbound request is rejected.



If the object is created successfully, Sorcery then has to determine what to do with it. While we've just had a piece of configuration pushed to Asterisk, Sorcery is responsible for storing it in some permanent - or semi-permanent - storage. For this, it looks to its configuration in `sorcery.conf`. We'll assume that our object should be created in the AstDB, a SQLite database. In that case, Asterisk pushes the newly created object to `res_sorcery_astdb`, which is the Sorcery driver for the AstDB. This module then writes the information to the SQLite database.

When the PJSIP stack next needs the object - such as when an `INVITE` request is received that maps to Alice's endpoint - it queries Sorcery for the object. At this point, from Sorcery's perspective, the retrieval of the configuration information is exactly the same as if a static configuration file or a relational database was providing the information, and it returns the pushed configuration information.

Asterisk Configuration

To make use of push configuration, you **must** configure Sorcery to persist the pushed configuration somewhere. If you don't want the information to persist beyond reloads, you can use the in-memory Sorcery driver, `res_sorcery_memory`. The example below assumes that we will push configuration to the PJSIP stack, and that we want information to persist even if Asterisk is restarted. For that reason, we'll use the AstDB.

sorcery.conf

```
[res_pjsip]

endpoint=astdb,ps_endpoints
auth=astdb,ps_auths
aor=astdb,ps_aors
domain_alias=astdb,ps_domain_aliases
contact=astdb,ps_contacts
system=astdb,ps_systems

[res_pjsip_endpoint_identifier_ip]
identify=astdb,ps_endpoint_id_ips

[res_pjsip_outbound_registration]
registration=astdb,ps_registrations
```

You **must** configure `sorcery.conf` for this feature to work. The standard data provider Sorcery uses for PJSIP objects is the static configuration file driver. This driver does not support creation, updating, or deletion of objects - which means only the `GET` request will succeed. Any of the following drivers will work to support push configuration:

- `res_sorcery_memory`
- `res_sorcery_astdb`

- res_sorcery_realtime

Pushing PJSIP Configuration

This walk-through will show how we can use the asterisk resource in ARI to push a PJSIP endpoint into the AstDB, and then later remove the endpoint.

Original PJSIP Configuration

Assume we have the following static PJSIP configuration file that defines an endpoint for Alice:

pjsip.conf

```
[transport-udp]
type=transport
protocol=udp
bind=0.0.0.0:5060

[transport-tcp]
type=transport
protocol=tcp
bind=0.0.0.0:5060

[alice]
type=aor
support_path=yes
remove_existing=yes
max_contacts=1

[alice]
type=auth
auth_type=userpass
username=alice
password=secret

[alice]
type=endpoint
from_user=alice
allow=!all,g722,ulaw,alaw
ice_support=yes
force_rport=yes
rewrite_contact=yes
rtp_symmetric=yes
context=default
auth=alice
aors=alice
```

If we then ask Asterisk what endpoints we have, it will show us something like the following:

Asterisk CLI

```
*CLI> pjsip show endpoints
 Endpoint:  <Endpoint/CID.....................................>  <State.....>
<Channels.>
    I/OAuth:
<AuthId/UserName...........................................................>
        Aor:  <Aor............................................>  <MaxContact>
      Contact:  <Aor/ContactUri...............................>  <Status....>
<RTT(ms)..>
  Transport:  <TransportId........>  <Type>  <cos>  <tos>
<BindAddress..................>
   Identify:
<Identify/Endpoint.........................................................>
        Match:  <ip/cidr.........................>
    Channel:  <ChannelId......................................>  <State.....>
<Time(sec)>
        Exten: <DialedExten...........>  CLCID: <ConnectedLineCID.......>
 ==========================================================================
====
 Endpoint:  alice                                                Unavailable   0 of
inf
     InAuth:  alice/alice
        Aor:  alice                                              1
*CLI>
```

Our goal is to recreate alice, using ARI.

New Configuration

PJSIP

Remove Alice from `pjsip.conf`:

pjsip.conf

```
[transport-udp]
type=transport
protocol=udp
bind=0.0.0.0:5060

[transport-tcp]
type=transport
protocol=tcp
bind=0.0.0.0:5060
```

Sorcery

Tell the Sorcery data abstraction framework to pull *endpoint*, *aor*, and *auth* objects from the Asterisk Database:

sorcery.conf

```
[res_pjsip]
endpoint=astdb,ps_endpoints
auth=astdb,ps_auths
aor=astdb,ps_aors
```

Asterisk CLI

Now, if we ask Asterisk for the PJSIP endpoints, it will tell us none are defined:

Asterisk CLI

```
*CLI> pjsip show endpoints
No objects found.
```

Pushing Configuration

First, let's push in Alice's authentication:

```
$ curl -X PUT -H "Content-Type: application/json" -u asterisk:secret -d '{"fields": [
{ "attribute": "auth_type", "value": "userpass"}, {"attribute": "username", "value":
"alice"}, {"attribute": "password", "value": "secret" } ] }'
https://localhost:8088/ari/asterisk/config/dynamic/res_pjsip/auth/alice

[{"attribute":"md5_cred","value":""},{"attribute":"realm","value":""},{"attribute":"au
th_type","value":"userpass"},{"attribute":"password","value":"secret"},{"attribute":"n
once_lifetime","value":"32"},{"attribute":"username","value":"alice"}]
```

We can note a few things from this:

1. We supply the non-default values that make up Alice's authentication in the JSON body of the request. The body specifies the "fields" to update, which is a list of attributes to modify on the object we're creating.
2. We don't have to provide default values for the object. This includes the "type" attribute - ARI is smart enough to figure out what that is from the request URI, where we specify that we are creating an "auth" object.
3. When we've created the object successfully, ARI returns back all the attributes that make up that object as a list of attribute/value pairs - even the attributes we didn't specify.

Next, we can push in Alice's AoRs:

```
$ curl -X PUT -H "Content-Type: application/json" -u asterisk:secret -d '{"fields": [
{ "attribute": "support_path", "value": "yes"}, {"attribute": "remove_existing",
"value": "yes"}, {"attribute": "max_contacts", "value": "1"} ] }'
https://localhost:8088/ari/asterisk/config/dynamic/res_pjsip/aor/alice

[{"attribute":"support_path","value":"true"},{"attribute":"default_expiration","value"
:"3600"},{"attribute":"qualify_timeout","value":"3.000000"},{"attribute":"mailboxes","
value":""},{"attribute":"minimum_expiration","value":"60"},{"attribute":"outbound_prox
y","value":""},{"attribute":"maximum_expiration","value":"7200"},{"attribute":"qualify
_frequency","value":"0"},{"attribute":"authenticate_qualify","value":"false"},{"attrib
ute":"contact","value":""},{"attribute":"max_contacts","value":"1"},{"attribute":"remo
ve_existing","value":"true"}]
```

Finally, we can push in Alice's endpoint:

```
$ curl -X PUT -H "Content-Type: application/json" -u asterisk:secret -d '{"fields": [
{ "attribute": "from_user", "value": "alice" }, { "attribute": "allow", "value":
"!all,g722,ulaw,alaw"}, {"attribute": "ice_support", "value": "yes"}, {"attribute":
"force_rport", "value": "yes"}, {"attribute": "rewrite_contact", "value": "yes"},
{"attribute": "rtp_symmetric", "value": "yes"}, {"attribute": "context", "value":
"default" }, {"attribute": "auth", "value": "alice" }, {"attribute": "aors", "value":
"alice"} ] }'
https://localhost:8088/ari/asterisk/config/dynamic/res_pjsip/endpoint/alice

[{"attribute":"timers_sess_expires","value":"1800"},{"attribute":"device_state_busy_at
","value":"0"},{"attribute":"dtls_cipher","value":""},{"attribute":"from_domain","valu
e":""},{"attribute":"dtls_rekey","value":"0"},{"attribute":"dtls_fingerprint","value":
"SHA-256"},{"attribute":"direct_media_method","value":"invite"},{"attribute":"send_rpi
d","value":"false"},{"attribute":"pickup_group","value":""},{"attribute":"sdp_session"
,"value":"Asterisk"},{"attribute":"dtls_verify","value":"No"},{"attribute":"message_co
ntext","value":""},{"attribute":"mailboxes","value":""},{"attribute":"named_pickup_gro
up","value":""},{"attribute":"record_on_feature","value":"automixmon"},{"attribute":"d
tls_private_key","value":""},{"attribute":"named_call_group","value":""},{"attribute":
"t38_udptl_maxdatagram","value":"0"},{"attribute":"media_encryption_optimistic","value
":"false"},{"attribute":"aors","value":"alice"},{"attribute":"rpid_immediate","value":
"false"},{"attribute":"outbound_proxy","value":""},{"attribute":"identify_by","value":
"username"},{"attribute":"inband_progress","value":"false"},{"attribute":"rtp_symmetri
c","value":"true"},{"attribute":"transport","value":""},{"attribute":"t38_udptl_ec","v
alue":"none"},{"attribute":"fax_detect","value":"false"},{"attribute":"t38_udptl_nat",
"value":"false"},{"attribute":"allow_transfer","value":"true"},{"attribute":"tos_video
","value":"0"},{"attribute":"srtp_tag_32","value":"false"},{"attribute":"timers_min_se
","value":"90"},{"attribute":"call_group","value":""},{"attribute":"sub_min_expiry","v
alue":"0"},{"attribute":"100rel","value":"yes"},{"attribute":"direct_media","value":"t
rue"},{"attribute":"g726_non_standard","value":"false"},{"attribute":"dtmf_mode","valu
e":"rfc4733"},{"attribute":"dtls_cert_file","value":""},{"attribute":"media_encryption
","value":"no"},{"attribute":"media_use_received_transport","value":"false"},{"attribu
te":"direct_media_glare_mitigation","value":"none"},{"attribute":"trust_id_inbound","v
alue":"false"},{"attribute":"force_avp","value":"false"},{"attribute":"record_off_feat
ure","value":"automixmon"},{"attribute":"send_diversion","value":"true"},{"attribute":
"language","value":""},{"attribute":"mwi_from_user","value":""},{"attribute":"rtp_ipv6
","value":"false"},{"attribute":"ice_support","value":"true"},{"attribute":"callerid",
"value":"<unknown>"},{"attribute":"aggregate_mwi","value":"true"},{"attribute":"one_to
uch_recording","value":"false"},{"attribute":"moh_passthrough","value":"false"},{"attr
ibute":"cos_video","value":"0"},{"attribute":"accountcode","value":""},{"attribute":"a
llow","value":"(g722|ulaw|alaw)"},{"attribute":"rewrite_contact","value":"true"},{"att
ribute":"t38_udptl_ipv6","value":"false"},{"attribute":"tone_zone","value":""},{"attri
bute":"user_eq_phone","value":"false"},{"attribute":"allow_subscribe","value":"true"},
{"attribute":"rtp_engine","value":"asterisk"},{"attribute":"auth","value":"alice"},{"a
ttribute":"from_user","value":"alice"},{"attribute":"disable_direct_media_on_nat","val
ue":"false"},{"attribute":"set_var","value":""},{"attribute":"use_ptime","value":"fals
e"},{"attribute":"outbound_auth","value":""},{"attribute":"media_address","value":""},
{"attribute":"tos_audio","value":"0"},{"attribute":"dtls_ca_path","value":""},{"attrib
ute":"dtls_setup","value":"active"},{"attribute":"force_rport","value":"true"},{"attri
bute":"connected_line_method","value":"invite"},{"attribute":"callerid_tag","value":""
},{"attribute":"timers","value":"yes"},{"attribute":"sdp_owner","value":"-"},{"attribu
te":"trust_id_outbound","value":"false"},{"attribute":"use_avpf","value":"false"},{"at
tribute":"context","value":"default"},{"attribute":"moh_suggest","value":"default"},{"
attribute":"send_pai","value":"false"},{"attribute":"t38_udptl","value":"false"},{"att
ribute":"dtls_ca_file","value":""},{"attribute":"callerid_privacy","value":"allowed_no
t_screened"},{"attribute":"cos_audio","value":"0"}]
```

We can now verify that Alice's endpoint exists:

Asterisk CLI

```
*CLI> pjsip show endpoints
 Endpoint:  <Endpoint/CID.....................................>  <State.....>
<Channels.>
    I/OAuth:
<AuthId/UserName...........................................................>
        Aor:  <Aor............................................>  <MaxContact>
      Contact:  <Aor/ContactUri...............................>  <Status....>
<RTT(ms)..>
  Transport:  <TransportId........>  <Type>  <cos>  <tos>
<BindAddress..................>
   Identify:
<Identify/Endpoint.........................................................>
        Match:  <ip/cidr.........................>
    Channel:  <ChannelId......................................>  <State.....>
<Time(sec)>
        Exten: <DialedExten...........>  CLCID: <ConnectedLineCID.......>
 =====================================================================================
====
 Endpoint:  alice/unknown                                        Invalid       0 of
inf
     InAuth:  alice/alice
        Aor:  alice                                              1
```

Order Matters!
While ARI itself won't care about the order you create objects in, PJSIP will. A PJSIP endpoint is used to look-up the endpoint's authentication and AoRs. Asterisk and ARI will let you create the endpoint first, referencing an authentication and AoR object that don't yet exist. If an inbound request arrives for that endpoint, the request will be rejected because the endpoint won't be able to find the authentication or store the Contact on a REGISTER request! By creating things in the order that we did, we can guarantee that everything will be in place when the endpoint is instantiated.

We can also verify that Alice exists in the AstDB:

Asterisk CLI

```
*CLI> database show
/ps_aors/aor/alice                                :
{"qualify_frequency":"0","maximum_expiration":"7200","minimum_expiration":"60","qualif
y_timeout":"3.000000","support_path":"true","default_expiration":"3600","mailboxes":""
,"authenticate_qualify":"false","outbound_proxy":"","max_contacts":"1","remove_existin
g":"true"}
/ps_auths/auth/alice                              :
{"realm":"","md5_cred":"","nonce_lifetime":"32","auth_type":"userpass","password":"sec
ret","username":"alice"}
/ps_endpoints/endpoint/alice                      :
{"send_diversion":"true","device_state_busy_at":"0","direct_media_method":"invite","sd
p_owner":"-","pickup_group":"","timers_sess_expires":"1800","message_context":"","acco
untcode":"","dtls_fingerprint":"SHA-256","rpid_immediate":"false","force_avp":"false",
"aors":"alice","trust_id_inbound":"false","ice_support":"true","fax_detect":"false","o
utbound_proxy":"","t38_udptl_maxdatagram":"0","direct_media_glare_mitigation":"none","
dtls_rekey":"0","context":"default","media_encryption_optimistic":"false","named_picku
p_group":"","from_domain":"","mailboxes":"","sdp_session":"Asterisk","cos_video":"0","
identify_by":"username","t38_udptl":"false","send_rpid":"false","rtp_engine":"asterisk
","t38_udptl_ec":"none","dtls_verify":"No","aggregate_mwi":"true","moh_suggest":"defau
lt","media_encryption":"no","callerid":"<unknown>","named_call_group":"","record_on_fe
ature":"automixmon","dtls_setup":"active","inband_progress":"false","timers_min_se":"9
0","tos_video":"0","rtp_symmetric":"true","rtp_ipv6":"false","transport":"","t38_udptl
_nat":"false","connected_line_method":"invite","allow_transfer":"true","allow_subscrib
e":"true","srtp_tag_32":"false","g726_non_standard":"false","100rel":"yes","use_avpf":
"false","call_group":"","moh_passthrough":"false","user_eq_phone":"false","allow":"(g7
22|ulaw|alaw)","sub_min_expiry":"0","force_rport":"true","direct_media":"true","dtmf_m
ode":"rfc4733","media_use_received_transport":"false","record_off_feature":"automixmon
","language":"","mwi_from_user":"","one_touch_recording":"false","rewrite_contact":"tr
ue","cos_audio":"0","t38_udptl_ipv6":"false","trust_id_outbound":"false","tone_zone":"
","auth":"alice","from_user":"alice","disable_direct_media_on_nat":"false","tos_audio"
:"0","use_ptime":"false","media_address":"","timers":"yes","send_pai":"false","calleri
d_privacy":"allowed_not_screened"}
3 results found.
*CLI>
```

Deleting Configuration

If we no longer want Alice to have an endpoint, we can remove it and its related objects using the `DELETE` operation:

```
$ curl -X DELETE -u asterisk:secret
https://localhost:8088/ari/asterisk/config/dynamic/res_pjsip/endpoint/alice
$ curl -X DELETE -u asterisk:secret
https://localhost:8088/ari/asterisk/config/dynamic/res_pjsip/aor/alice
$ curl -X DELETE -u asterisk:secret
https://localhost:8088/ari/asterisk/config/dynamic/res_pjsip/auth/alice
```

And we can confirm that Alice no longer exists:

Asterisk CLI

```
*CLI> pjsip show endpoints
No objects found.
*CLI> database show
0 results found.
*CLI>
```

⚠ **Order Matters!**
Note that we remove Alice in reverse order of how her endpoint was created - we first remove the endpoint, then its related objects. Once the endpoint is removed, further requests will no longer be serviced, which allows us to safely remove the auth and aor objects.

ARI Versioning

ARI version numbers are formatted as MAJOR.BREAKING.NON-BREAKING:

- MAJOR – changes when a new major version of Asterisk is released
- BREAKING – changes when an incompatible API modification is made
- NON-BREAKING – changes when backwards compatible updates are made (new additions or bug fixes)

Any time a new major version of Asterisk is released the ARI version number is modified to reflect that. This is done by taking the major ARI number of the last major release of Asterisk, increasing it by one, and then resetting the other numbers to zero. From there the breaking and non-breaking versions are increased by one when a change (incompatible and compatible respectively) has been made to ARI.

This has the effect of making the ARI version number relative in relation to an associated Asterisk release, and contextual applicable only to that major release of Asterisk.

ARI Libraries

ARI Libraries

Listing of community Asterisk Rest Interface libraries and frameworks.

Library	Language	Website
`ari-py`	Python	https://github.com/asterisk/ari-py
`AsterNET.ARI`	C# / .NET	https://github.com/skrusty/AsterNET.ARI
`ari4java`	Java	https://github.com/l3nz/ari4java
`node-ari-client`	JavaScript (node)	https://github.com/asterisk/node-ari-client
phpari	PHP	http://www.phpari.org/
asterisk-ari-client	Ruby	https://github.com/svoboda-jan/asterisk-ari
Phparia	PHP	http://phparia.org/
aricpp	C++	http://github.com/daniele77/aricpp

Back-end Database and Realtime Connectivity

Under Construction

Top-level page for configuration pages about ODBC modules, native database connectors and modules used for realtime.

cURL

⚠ This page is under construction and may be incomplete or missing information in some areas. If you have questions, please wait until this notice is removed before asking, since it is possible your question will be answered by the time this page is completed.

Asterisk's ability to retrieve and store data to realtime backends is most commonly associated with relational databases. One of the lesser-known realtime backends available in Asterisk is cURL. Using this realtime backend makes Asterisk use HTTP GET and POST requests in order to retrieve data from and store data to an HTTP server.

Justification

If Asterisk is capable of using a relational database as a store for realtime data, then what is the need for using HTTP? There are several potential reasons:

- Your setup hinges on a web service such as Django or something else, and you would prefer that Asterisk go through this service instead of skirting it to get directly at the data.
- You are forced to use a database that Asterisk does not have a native backend for and whose ODBC support is subpar.
- A relational database carries too much overhead

Dependencies and Installation

Asterisk's realtime cURL backend is provided by the module `res_config_curl.so`. In order to build this module, you will first need to have the libcurl development library installed on your machine. If you wish to install the library from source, you can find it here. If you would rather use your Linux distribution's package management, then you should be able to download the development libraries that way instead.

If you use a distribution with aptitude-based packaging (Debian, Ubuntu, Mint, et al), then use this command to install:

```
apt-get install libcurl4-openssl-dev
```

If you use a distribution with yum-based packaging (CentOS, RHEL, Fedora, et al), then use this command to install:

```
yum -y install libcurl-devel
```

Both of the above commands assume that you have permission to install the packages. You may need to prepend the command with "sudo" in order to be able to install the packages.

Once you have the libcurl development libraries installed, you need to run Asterisk's configure script in order for Asterisk to detect the installed library:

```
$ ./configure
```

In addition to the libcurl development library, `res_config_curl.so` relies on two other modules within Asterisk: `res_curl.so` and `func_curl.so`. `res_curl.so` initializes the cURL library within Asterisk. `func_curl.so` provides dialplan functions (`CURL` and `CURLOPT`) that are used directly by `res_config_curl.so`.

After running the configure script, run

```
$ make menuselect
```

to select which modules to build. Ensure that you can select `res_curl` and `res_config_curl` from the "Resource Modules" menu and that you can select `func_curl` from the "Dialplan Functions" menu. Once you have ensured that these have been selected, save your changes ('x' key if using curses-based menuselect or select the "Save & Exit" option if using newt-based or gtk-based menuselect). After, you just need to run

```
$ make && make install
```

in order to build Asterisk and install it on the system. You may need to prepend "sudo" to the "make install" command if there are permission problems when attempting to install. Once you have installed Asterisk, you can test that `res_config_curl.so` has been installed properly by starting Asterisk:

```
$ asterisk -c
```

Once Asterisk has started, type the following on the CLI:

```
*CLI> module show like res_config_curl
Module                         Description                              Use Count  Status
res_config_curl.so             Realtime Curl configuration              0          Running
1 modules loaded
```

The output when you run the command should look like what is shown above. If it does, then Asterisk is capable of using cURL for realtime.

Troubleshooting

If you encounter problems along the way, here are some tips to help you get back on track.

- If the required modules in Asterisk are unselectable when you run `make menuselect`, then Asterisk did not detect the libcurl development library on your machine. If you installed the libcurl development library in a nonstandard place, then when running Asterisk's configure script, specify `--with-curl=/path/to/library` so that Asterisk can know where to look.
- If you built the required Asterisk modules but the `res_config_curl.so` module is not properly loaded, then check your `modules.conf` file to ensure that the necessary modules are being loaded. If you are noloading any of the required modules, then `res_config_curl.so` will not be able to load. If you are loading modules individually, be sure to list `res_curl.so` and `func_curl.so` before `res_config_curl.so` in your configuration.

Configuration

Unlike other realtime backends, Asterisk does not have a specific configuration file for the realtime cURL backend. Instead, Asterisk gets the information it needs by reading the `extconfig.conf` file that it typically uses for general static and dynamic realtime configuration. The name of the realtime engine that Asterisk uses for cURL is called "curl" in `extconfig.conf`. Here is a sample:

```
[settings]
voicemail = curl,http://myserver.com:8000/voicemail
sippeers = curl,http://myserver.com:8000/sippeers
queues = curl,http://myserver.com:8000/my_queues
```

The basic syntax when using cURL is:

```
realtime_data = curl,<HTTP URL>
```

There are no hard-and-fast rules on what URL you place here. In the above sample, each of the various realtime stores correspond to resources on the same HTTP server. However, it would be perfectly valid to specify completely different servers for different realtime stores. Notice also that there is no requirement for the name of the realtime store to appear in the HTTP URL. In the above example the "queues" realtime store maps to the resource "my_queues" on the HTTP server.

Operations

The way Asterisk performs operations on your data is to send HTTP requests to different resources on your HTTP server. For instance, let's say that, based on your `extconfig.conf` file, you have mapped the "queues" realtime store to http://myserver.com:8000/queues. Asterisk will append whatever realtime operation it wishes to perform as a resource onto the end of the URL that you have provided. If Asterisk wanted to perform the "single" realtime operation, then Asterisk would send an HTTP request to http://myserver.com:8000/queues/single.

If your server is able to provide a response, then your server should return that response as the body of a 200-class HTTP response. If the request is unservable, then an appropriate HTTP error code should be sent.

The operations, as well as what is expected in response, are defined below.

For the first five examples, we will be using external MWI as the sample realtime store that Asterisk will be interacting with. The realtime MWI store stores the following data for each object

- id: The name of the mailbox for which MWI is being provided
- msgs_new: The number of new messages the mailbox currently has
- msgs_old: The number of old messages the mailbox currently has

We will operate with the assumption that the following two objects exist in the realtime store:

- -
    - id: "Dazed"
    - msgs_new: 5
    - msgs_old: 4
- -

- id: "Confused"
- msgs_new: 6
- msgs_old: 8

Our `extconfig.conf` file looks like this:

```
[settings]
mailboxes => curl,http://myserver.com:8000/mwi
```

single

The "single" resource is used for Asterisk to retrieve a single object from realtime.

Asterisk sends an HTTP POST request, using the body to indicate what data it wants. Here is an example of such a request:

```
POST /mwi/single HTTP/1.1
User-Agent: asterisk-libcurl-agent/1.0
Host: localhost:8000
Accept: */*
Content-Length: 8
Content-Type: application/x-www-form-urlencoded

id=Dazed
```

In this case, the request from Asterisk wants a single object whose id is "Dazed". Given the data we have stored, we would respond like so:

```
HTTP/1.1 200 OK
Date: Sat, 15 Mar 2014 18:23:21 GMT
Content-Length: 30
Content-Type: text/html

msgs_new=5&msgs_old=4&id=Dazed
```

The parameters describing the requested mailbox are returned on a single line in the HTTP response body. The order that the parameters are listed in is irrelevant.

If a "single" query from Asterisk matches more than one entity, you may choose to either respond with an HTTP error or simply return one of the matching records.

multi

The "multi" resource is used to retrieve multiple objects from the realtime store.

Asterisk sends an HTTP POST request, using the body to indicate what data it wants. Here is an example of such a request:

```
POST /mwi/multi HTTP/1.1
User-Agent: asterisk-libcurl-agent/1.0
Host: localhost:8000
Accept: */*
Content-Length: 13
Content-Type: application/x-www-form-urlencoded

id%20LIKE=%25
```

The "multi" resource is one where Asterisk shows a weakness when not dealing with a relational database as its realtime backend. In this case, Asterisk has requested multiple rows with "id LIKE=%". What this means is that Asterisk wants to retrieve every object from the particular realtime store with an id equal to anything. Other queries Asterisk may send may be more like "foo LIKE=%bar%". In this case, Asterisk would be requesting all objects with a foo parameter that has "bar" as part of its value (so something with foo=barbara would match the query).

For this particular request, we would respond with the following:

```
HTTP/1.1 200 OK
Date: Sat, 15 Mar 2014 18:40:58 GMT
Content-Length: 65
Content-Type: text/html

msgs_new=5&msgs_old=4&id=Dazed
msgs_new=6&msgs_old=8&id=Confused
```

Each returned object is on its own line of the response.

store

The "store" resource is used to save an object into the realtime store.

Asterisk sends an HTTP POST request, using the body to indicate what new object to store. Here is an example of such a request:

```
POST /mwi/store HTTP/1.1
User-Agent: asterisk-libcurl-agent/1.0
Host: localhost:8000
Accept: */*
Content-Length: 30
Content-Type: application/x-www-form-urlencoded

id=Shocked&msgs_old=5&msgs_new=7
```

In this case, Asterisk is attempting to store a new object with id "Shocked", 5 old messages and 7 new messages. Our realtime backend should reply with the number of objects stored.

```
HTTP/1.1 200 OK
Date: Sat, 15 Mar 2014 18:46:54 GMT
Content-Length: 1
Content-Type: text/html

1
```

Since we have stored one new object, we return "1" as our response.

If attempting to store an item that already exists in the database, you may either return an HTTP error or overwrite the old object with the new, depending on your policy.

update

The "update" resource is used to change the values of parameters of objects in the realtime store.

Asterisk sends an HTTP POST request, using URL parameters to indicate what objects to update and using the body to indicate what values within those objects to update. Here is an example of such a request:

```
POST /mwi/update?id=Dazed HTTP/1.1
User-Agent: asterisk-libcurl-agent/1.0
Host: localhost:8000
Accept: */*
Content-Length: 24
Content-Type: application/x-www-form-urlencoded

msgs_old=25&msgs_new=300
```

In this case, the URL parameter "id=Dazed" tells us that Asterisk wants us to update all objects whose id is "Dazed". For any objects that match the criteria, we should update the number of old messages to 25 and the number of new messages to 300.

Our response indicates how many objects we updated. In this case, since we have updated one object, we respond with "1".

```
HTTP/1.1 200 OK
Date: Sat, 15 Mar 2014 18:52:26 GMT
Content-Length: 1
Content-Type: text/html

1
```

If there are no items that match the criteria, you may either respond with a "0" response body or return an HTTP error.

destroy

The "destroy" resource is used to delete objects in the realtime store.

Asterisk sends an HTTP POST request, using the body to indicate what object to delete. Here is an example of such a request:

```
POST /mwi/destroy HTTP/1.1
User-Agent: asterisk-libcurl-agent/1.0
Host: localhost:8000
Accept: */*
Content-Length: 9
Content-Type: application/x-www-form-urlencoded

id=Dazed
```

In this case, Asterisk has requested that we delete the object with the id of "Dazed".

The body of our response indicates the number of items we deleted. Since we have deleted one object, we put "1" in our response body:

```
HTTP/1.1 200 OK
Date: Sat, 15 Mar 2014 18:57:23 GMT
Content-Length: 1
Content-Type: text/html

1
```

If asked to delete an object that does not exist, you may either respond with a "0" body or with an HTTP error.

static

The "static" resource is used for static realtime requests.

Static realtime is a different realm from the more common dynamic realtime. Whereas dynamic realtime is restricted to certain configuration types that are designed to be used this way, static realtime uses a generic construct that can be substituted for any configuration file in Asterisk. The downside to static realtime is that Asterisk only ever interacts with the static realtime backend when the module that uses the configuration is reloaded. Internally, the Asterisk module thinks that it is reading its configuration from a configuration file, but under the hood, the configuration is actually retrieved from a realtime backend.

Static realtime "objects" are all the same, no matter what configuration file the static realtime store is standing in for. Object has been placed in quotation marks in that previous sentence because each static realtime object does not represent an entire configuration object, but rather represents a line in a configuration file. Here are the parameters for each static realtime object:

- id: A unique numerical id for the static realtime object
- filename: The configuration file that this static realtime object belongs to
- cat_metric: Numerical id for a configuration category. Used by Asterisk to order categories for evaluation.
- category: Name of the configuration category
- var_metric: Numerical id for a variable within a category. Used by Asterisk to order variables for evaluation.
- var_name: Parameter name
- var_val: Parameter value
- commented: If non-zero, indicates this object should be ignored

For our example, we will have the following objects stored in our static realtime store:

- -
    - id: 0
    - cat_metric: 0
    - var_metric: 0
    - filename: pjsip.conf
    - category: alice
    - var_name: type
    - var_val: endpoint
    - commented: 0
- -
    - id: 1
    - cat_metric: 0
    - var_metric: 1
    - filename: pjsip.conf
    - category: alice
    - var_name: allow
    - var_val: ulaw
    - commented: 0
- -
    - id: 2
    - cat_metric: 0
    - var_metric: 2
    - filename: pjsip.conf
    - category: alice
    - var_name: context
    - var_val: fabulous
    - commented: 0

This schema is identical to the `pjsip.conf` configuration file:

```
[alice]
type=endpoint
allow=ulaw
context=fabulous
```

Asterisk uses an HTTP GET to request static realtime data, using a URL parameter to indicate which filename it cares about. Here is an example of such a request:

```
GET /astconfig/static?file=pjsip.conf HTTP/1.1
User-Agent: asterisk-libcurl-agent/1.0
Host: localhost:8000
Accept: */*
```

In this case, Asterisk wants all static realtime objects whose filename is "pjsip.conf". Note that the HTTP request calls the parameter "file", whereas the actual name of the parameter returned from the realtime store is called "filename".

Our response contains all matching static realtime objects:

```
HTTP/1.1 200 OK
Date: Sat, 15 Mar 2014 19:13:41 GMT
Content-Length: 328
Content-Type: text/html

category=alice&commented=0&var_metric=0&var_name=type&var_val=endpoint&id=0&filename=pjsip.conf&cat_metric=0
category=alice&commented=0&var_metric=1&var_name=allow&var_val=ulaw&id=1&filename=pjsip.conf&cat_metric=0
category=alice&commented=0&var_metric=2&var_name=context&var_val=fabulous&id=2&filename=pjsip.conf&cat_metric=0
```

Unlike other realtime responses, the static realtime response needs to present the data in a particular order:

- First order: by descending cat_metric
- Second order: by ascending var_metric
- Third: lexicographically by category name
- Fourth: lexicographically by variable name

Note that Asterisk only pays attention to the "cat_metric", "var_metric", "category", "var_name", and "var_value" you return here, but you are free to return the entire object if you want. Note that Asterisk will not pay attention to the "commented" field, so be sure not to return any objects that have a non-zero "commented" value.

In summary, static realtime is cumbersome, confusing, and not worth it. Stay clear unless you just really need to use it.

If your realtime store does not provide objects for the specified file, then you may either return an empty body or an HTTP error.

require

The "require" resource is used by Asterisk to test that a particular parameter for a realtime object is of a type it expects.

Asterisk sends an HTTP POST with body parameters describing what type it expects for a specific parameter. Here is an example of such a request:

```
POST /queue_members/require HTTP/1.1
User-Agent: asterisk-libcurl-agent/1.0
Host: localhost:8000
Accept: */*
Content-Length: 42
Content-Type: application/x-www-form-urlencoded

paused=integer1%3A1&uniqueid=uinteger2%3A5
```

Decoded, the body is "paused=integer1:1&uniqueid=uinteger2:5". The types that Asterisk can ask for are the following:

- An integer type can be indicated by any of the following
  - integer1
  - integer2
  - integer3
  - integer4
  - integer8
- An unsigned integer type can be indicated by any of the following
  - uinteger1
  - uinteger2
  - uinteger3
  - uinteger4
  - uinteger8
- char: A string
- date: A date
- datetime: A datetime
- float: a floating point number

It is undocumented what the meaning of the number after each of the "integer" and "uinteger" types means. If I'm guessing, it's the number of bytes allocated for the type.

The number after the colon for each parameter represents the minimum width, in digits for integer types and characters for char types, for each parameter.

In the example above, Asterisk is requiring that queue members' "paused" parameter be an integer type that can hold at least 1 digit and their "uniqueid" parameter be an unsigned integer type that can hold at least 5 digits.

Note that the purpose of Asterisk requesting the "require" resource is because Asterisk is going to attempt to **send** data of the type indicated to the realtime store. When receiving such a request, it is completely up to how you are storing your data to determine how to respond. If you are using a schema-less store for your data, then trying to test for width and type for each parameter is pointless, so you may as well just return successfully. If you are using something that has a schema you can check, then you should be sure that your realtime store can accommodate the data Asterisk will send. If your schema cannot accommodate the data, then this is an ideal time to modify the data schema if it is possible.

Respond with a "0" body to indicate success or a "-1" body to indicate failure. Here is an example response:

```
HTTP/1.1 200 OK
Date: Sat, 15 Mar 2014 18:57:23 GMT
Content-Length: 1
Content-Type: text/html

0
```

Other Information

If you are interested in looking more in-depth into Asterisk's cURL realtime backend, you can find a reference example of an HTTP server in `contrib/scripts/dbsep.cgi` written by Tilghman Lesher, who also wrote Asterisk's realtime cURL support. This script works by converting the HTTP request from Asterisk into a SQL query for a relational database. The configuration for the database used in the `dbsep.cgi` script is detailed in the `configs/dbsep.conf.sample` file in the Asterisk source.

Followme - Realtime

Followme is now realtime-enabled.

To use, you must define two backend data structures, with the following fields:

```
followme:
 name                Name of this followme entry.  Specified when invoking the FollowMe
                     application in the dialplan.  This field is the only one which is
                     mandatory.  All of the other fields will inherit the default from
                     followme.conf, if not specified in this data resource.
 musicclass OR       The musiconhold class used for the caller while waiting to be
 musiconhold OR      connected.
    music
 context             Dialplan context from which to dial numbers
 takecall            DTMF used to accept the call and be connected.  For obvious reasons,
                     this needs to be a single digit, '*', or '#'.
 declinecall         DTMF used to refuse the call, sending it onto the next step, if any.
 call_from_prompt    Prompt to play to the callee, announcing the call.
 norecording_prompt  The alternate prompt to play to the callee, when the caller
                     refuses to leave a name (or the option isn't set to allow them).
 options_prompt      Normally, "press 1 to accept, 2 to decline".
 hold_prompt         Message played to the caller while dialing the followme steps.
 status_prompt       Normally, "Party is not at their desk".
 sorry_prompt        Normally, "Unable to locate party".
```

```
followme_numbers:
 name                Name of this followme entry.  Must match the name above.
 ordinal             An integer, specifying the order in which these numbers will be
                     followed.
 phonenumber         The telephone number(s) you would like to call, separated by '&'.
 timeout             Timeout associated with this step.  See the followme documentation
                     for more information on how this value is handled.
```

LDAP Realtime Driver

Asterisk Realtime Lightweight Directory Access Protocol (LDAP) Driver

With this driver Asterisk, using the Realtime Database Configuration, can access and update information in an LDAP directory. Asterisk can configure SIP/IAX2 users, extensions, queues, queue members, and entire configuration files. This guide assumes you have a working knowledge of LDAP and have an LDAP server with authentication already setup. Asterisk requires read and write permissions to update the directory.

See configs/res_ldap.conf.sample for a configuration file sample.
See contrib/scripts for the LDAP schema and ldif files needed for the LDAP server.

> ⚠ To use static realtime with certain core configuration files the realtime backend you wish to use must be preloaded in `modules.conf`.

From within your Asterisk source directory:

```
cd contrib/scripts
sudo cp asterisk.ldap-schema /etc/ldap/schema/
sudo service slapd restart
sudo ldapadd -Y EXTERNAL -H ldapi:/// -f ./asterisk.ldif
```

Let's edit the extconfig.conf file to specify LDAP as our realtime storage engine and where Asterisk will look for data.

```
sippeers = ldap,"ou=sip,dc=example,dc=domain",sip
sipusers = ldap,"ou=sip,dc=example,dc=domain",sip
extensions = ldap,"ou=extensions,dc=example,dc=domain",extensions
```

> ⚠ You'll want to reference the Asterisk res_ldap.conf file which holds the LDAP mapping configuration when building your own record schema.

Basic sip users record layout which will need to be saved to a file (we'll use 'createduser.ldif' here as an example). This example record is for sip user '1000'. This example record is for sip user '1000'.

```
dn: cn=1000,ou=sip,dc=digium,dc=internal
objectClass: AsteriskAccount
objectClass: AsteriskExtension
objectClass: AsteriskSIPUser
objectClass: top
AstAccountName: sip user
cn: 1000
AstAccountDefaultUser: 0
AstAccountExpirationTimestamp: 0
AstAccountFullContact: 0
AstAccountHost: dynamic
AstAccountIPAddress: 0
AstAccountLastQualifyMilliseconds: 0
AstAccountPort: 0
AstAccountRegistrationServer: 0
AstAccountType: 0
AstAccountUserAgent: 0
AstExtension: 1000
```

Let's add the record to the LDAP server:

```
sudo ldapadd -D "cn=admin,dc=example,dc=domain" -x -W -f createduser.ldif
```

When creating your own record schema, you'll obviously want to incorporate authentication. Asterisk + LDAP requires that the user secrets be stored as an MD5 hash. MD5 hashes can be created using 'md5sum'.

For AstAccountRealmedPassword authentication use this.

```
printf "<secret composed of username, realm, and password goes here>" | md5sum
```

For AstMD5secret authentication use this.

```
printf "password" | md5sum
```

ODBC

Under Construction

Top-level page for everything about configuring ODBC connectivity, res_odbc, func_odbc, etc

Configuring res_odbc

Overview

The **res_odbc** module for Asterisk can provide Asterisk with connectivity to various database backends through ODBC (a database abstraction layer). Asterisk features such as Asterisk Realtime Architecture, Call Detail Records, Channel Event Logging, can connect to a database through res_odbc.

More details on specific options within configuration are provided in the sample configuration file included with Asterisk source.

We'll provide a brief guide here on how to get the res_odbc.so module configured to connect to an existing ODBC installation.

Recompile Asterisk to build required modules

You'll need to rebuild Asterisk with the needed modules.

Other pages on the wiki describe that process:

Building and Installing Asterisk

Using Menuselect to Select Asterisk Options

When using menuselect, verify that the **func_odbc** (you'll probably be using that one) and **res_odbc** (required) modules will be built. Then, build Asterisk and make sure those modules were built and exist in **/usr/lib/asterisk/modules** (or whatever directory you use).

Configure res_odbc.conf to connect to your ODBC installation

Find the configuration file, which should typically be located at /etc/asterisk/res_odbc.conf and provide a basic configuration such as:

```
[asterisk]
enabled => yes
dsn => your-configured-dsn-name
username => your-database-username
password => insecurepassword
pre-connect => yes
```

Then start up Asterisk and assuming res_odbc loads properly on the CLI you can use odbc show to verify a DSN is configured and shows up:

```
rnewton-office-lab*CLI> odbc show
ODBC DSN Settings
-----------------
  Name:   asterisk
  DSN:    your-configured-dsn-name
    Last connection attempt: 1969-12-31 18:00:00
```

To verify the connection works you should use func_odbc or something similar to query the data source from Asterisk.

Troubleshooting

If you don't have the **odbc** command at the CLI, check that

- The res_odbc.so module exists and has proper permissions in /usr/lib/asterisk/modules/
- Your modules.conf to make sure the module isn't noloaded or being prevented from loading somehow
- Debug during Asterisk startup to look for messages regarding res_odbc.conf (see logger.conf to get things setup)

If you the **odbc show** output shows "Connected: No" then you'll want to try connecting to your ODBC installation from other methods to verify it is working. The Linux tool **isql** is good for that.

Getting Asterisk Connected to MySQL via ODBC

Connect Asterisk to a MySQL back-end through ODBC

This is a short tutorial on how to quickly setup Asterisk to use MySQL, the ODBC MySQL connector and ODBC. We'll use CentOS 6 as the OS in this tutorial. However, the same essential steps apply for any popular Linux distro.

Installing and Configuring MySQL

There are three basic steps to install and configure MySQL for Asterisk.

- Install MySQL server package and start the DB service.
- Secure the installation if appropriate.
- Configure a user and database for Asterisk in MySQL

On This Page

Install MySQL server package and start the DB service

```
# sudo yum install mysql-server
# sudo service mysqld start
```

Secure the installation if appropriate

If you intend to push this install into production or practice as if you were then you will want to use the mysql_secure_installation script to apply some basic security.

```
# sudo /usr/bin/mysql_secure_installation
```

Configure a user and database for Asterisk in MySQL

If you want to use a GUI to manage your database then now is the time to set that GUI up and use it to create your asterisk user. Otherwise we will provide basic instructions for user setup below.

We'll have you login into the mysql command console, create a user, create a database and then assign appropriate permissions for the asterisk user. First, login using the root password you set earlier.

```
# mysql -u root -p
```

Now verify you are at the MySQL command prompt. It should look like "mysql>". Then enter the following commands:

```
# CREATE USER 'asterisk'@'%' IDENTIFIED BY 'replace_with_strong_password';
# CREATE DATABASE asterisk;
# GRANT ALL PRIVILEGES ON asterisk.* TO 'asterisk'@'%';
# exit
```

After each of the CREATE and GRANT commands you should see output indicating that the Query was OK including many rows were affected.

If you want, you can test out the new permissions by logging in as your user to the asterisk database and then logout again.

```
# mysql -u asterisk -p asterisk
# exit
```

Install ODBC and the MySQL ODBC connector

Be sure you have followed the previous sections as we presume you already have MySQL installed on your CentOS server along with a database and user for Asterisk configured. The database name should be 'asterisk' and the username should be 'asterisk'.

Install the latest unixODBC and GNU Libtool Dynamic Module Loader packages

The development packages are necessary as well, since later Asterisk will need to use them when building ODBC related modules.

```
# sudo yum install unixODBC unixODBC-devel libtool-ltdl libtool-ltdl-devel
```

Install the latest MySQL ODBC connector

```
# sudo yum install mysql-connector-odbc
```

Configure ODBC and the MySQL ODBC connector

Configure odbcinst.ini for ODBC

With recent UnixODBC versions the configuration should already be done for you in the /etc/odbcinst.ini file.

Verify that you have the following configuration:

```
# Driver from the mysql-connector-odbc package
# Setup from the unixODBC package
[MySQL]
Description     = ODBC for MySQL
Driver          = /usr/lib/libmyodbc5.so
Setup           = /usr/lib/libodbcmyS.so
Driver64        = /usr/lib64/libmyodbc5.so
Setup64         = /usr/lib64/libodbcmyS.so
FileUsage       = 1
Pooling         = Yes
CPTimeout       = 120
```

⚠ We configure pooling for the connection as that is probably desired by most users. Consult the unixODBC documentation to learn more about the function of ODBC connection pooling.

There may also be configuration for PostgreSQL which you can comment out if you are not planning to setup PostgreSQL as well. Comments begin the line with a hash (#) symbol.

You can also call **`odbcinst`** to query the driver, verifying that the configuration is found.

```
# odbcinst -q -d
```

The output should read simply "[MySQL]"

Configure the MySQL ODBC connector

Now we'll configure the /etc/odbc.ini file to create a DSN (Data Source Name) for Asterisk. The file may be empty, so you'll have to copy-paste from this example or write this from scratch.

Add the following to /etc/odbc.ini

```
[asterisk-connector]
Description = MySQL connection to 'asterisk' database
Driver = MySQL
Database = asterisk
Server = localhost
Port = 3306
Socket = /var/lib/mysql/mysql.sock
```

You may want to verify that mysql.sock is actually in the location specific here. It will differ on some systems depending on your configuration.

Test the ODBC Data Source Name connection

Now is a good time to test your database by connecting to it and performing a query. The unixODBC package provides **`isql`**; a command line utility that allows you to connect to the Data Source, send SQL commands to it and receive results back. The syntax used is:

isql -v *dsn_name db_username db_password*

So, for our purposes you would enter:

```
# isql -v asterisk-connector asterisk replace_with_strong_password
```

It is important to use the -v flag so that if isql runs into a problem you will be alerted of any diagnostics or errors available.

At this point you should get an SQL prompt. Run the following command:

```
SQL> select 1
```

You should see some simple results if the query is successful. Then you can exit.

```
SQL> select 1
+---------------------+
| 1                   |
+---------------------+
| 1                   |
+---------------------+
SQLRowCount returns 1
1 rows fetched

SQL> quit
```

Configuring Asterisk to Use the New ODBC and MySQL Install

Now you have a MySQL database, ODBC and an ODBC MySQL connector installed and basically configured. The next step is to recompile Asterisk so that the ODBC modules which required the previously mentioned items can now be built. Once those modules exist, then you can configure the proper configuration files in Asterisk depending on what information you want to write to or read from MySQL.

Getting the right Asterisk modules

If you already had Asterisk installed from source and the modules you need are already selected by default in menuselect - then the recompilation process could be as simple as navigating to the Asterisk source and running a few commands.

```
# cd ~/asterisk-source/
# ./configure
# make && make install
```

Otherwise you should follow the typical Asterisk installation process to make sure modules such as res_odbc, res_config_odbc, cdr_odbc, cdr_adaptive_odbc and func_odbc have their dependencies fulfilled and that they will be built.

See Building and Installing Asterisk and Using Menuselect to Select Asterisk Options.

Configuring Asterisk's ODBC connection

The basic configuration for an Asterisk ODBC connection is handled in res_odbc.conf. You should check out the Configuring res_odbc page and follow it using the DSN and database username and password you setup earlier.

After you have the connection set up in Asterisk you are ready to then configure your database tables with the proper schema depending on what exactly you want to do with them. Asterisk comes with some helpful tools to do this, such as Alembic. See the Managing Realtime Databases with Alembic section to get started with Alembic if you are working towards an Asterisk Realtime setup.

SQLite Tables

```
/*
 * res_config_sqlite - SQLite 2 support for Asterisk
 *
 * This module can be used as a static/RealTime configuration module, and a CDR
 * handler.  See the Doxygen documentation for a detailed description of the
 * module, and the configs/ directory for the sample configuration file.
 */

/*
 * Tables for res_config_sqlite.so.
 */

/*
 * RealTime static table.
 */
CREATE TABLE ast_config (
 id  INTEGER,
 cat_metric INT(11)  NOT NULL DEFAULT 0,
 var_metric INT(11)  NOT NULL DEFAULT 0,
 commented TINYINT(1) NOT NULL DEFAULT 0,
 filename VARCHAR(128) NOT NULL DEFAULT '',
 category VARCHAR(128) NOT NULL DEFAULT 'default',
 var_name VARCHAR(128) NOT NULL DEFAULT '',
 var_val  TEXT  NOT NULL DEFAULT '',
 PRIMARY KEY (id)
);

CREATE INDEX ast_config__idx__cat_metric  ON ast_config(cat_metric);
CREATE INDEX ast_config__idx__var_metric  ON ast_config(var_metric);
CREATE INDEX ast_config__idx__filename_commented ON ast_config(filename, commented);

/*
 * CDR table (this table is automatically created if non existent).
 */
CREATE TABLE ast_cdr (
 id  INTEGER,
 clid  VARCHAR(80) NOT NULL DEFAULT '',
 src  VARCHAR(80) NOT NULL DEFAULT '',
 dst  VARCHAR(80) NOT NULL DEFAULT '',
 dcontext VARCHAR(80) NOT NULL DEFAULT '',
 channel  VARCHAR(80) NOT NULL DEFAULT '',
 dstchannel VARCHAR(80) NOT NULL DEFAULT '',
 lastapp  VARCHAR(80) NOT NULL DEFAULT '',
 lastdata VARCHAR(80) NOT NULL DEFAULT '',
 start  DATETIME NOT NULL DEFAULT '0000-00-00 00:00:00',
 answer  DATETIME NOT NULL DEFAULT '0000-00-00 00:00:00',
 end  DATETIME NOT NULL DEFAULT '0000-00-00 00:00:00',
 duration INT(11)  NOT NULL DEFAULT 0,
 billsec  INT(11)  NOT NULL DEFAULT 0,
 disposition VARCHAR(45) NOT NULL DEFAULT '',
 amaflags INT(11)  NOT NULL DEFAULT 0,
 accountcode VARCHAR(20) NOT NULL DEFAULT '',
 uniqueid VARCHAR(32) NOT NULL DEFAULT '',
 userfield VARCHAR(255) NOT NULL DEFAULT '',
 PRIMARY KEY (id)
);

/*
 * SIP RealTime table.
 */
CREATE TABLE ast_sip (
 id  INTEGER,
 commented TINYINT(1) NOT NULL DEFAULT 0,
 name  VARCHAR(80) NOT NULL DEFAULT '',
 host  VARCHAR(31) NOT NULL DEFAULT '',
 nat  VARCHAR(5) NOT NULL DEFAULT 'no',
 type  VARCHAR(6) NOT NULL DEFAULT 'friend',
 accountcode VARCHAR(20)   DEFAULT NULL,
 amaflags VARCHAR(13)   DEFAULT NULL,
 callgroup VARCHAR(10)   DEFAULT NULL,
 callerid VARCHAR(80)   DEFAULT NULL,
 cancallforward CHAR(3)    DEFAULT 'yes',
 directmedia CHAR(3)    DEFAULT 'yes',
 context  VARCHAR(80)   DEFAULT NULL,
 defaultip VARCHAR(15)   DEFAULT NULL,
 dtmfmode VARCHAR(7)   DEFAULT NULL,
 fromuser VARCHAR(80)   DEFAULT NULL,
 fromdomain VARCHAR(80)   DEFAULT NULL,
 insecure VARCHAR(4)   DEFAULT NULL,
 language CHAR(2)    DEFAULT NULL,
 mailbox  VARCHAR(50)   DEFAULT NULL,
 md5secret VARCHAR(80)   DEFAULT NULL,
 deny  VARCHAR(95)   DEFAULT NULL,
 permit  VARCHAR(95)   DEFAULT NULL,
 mask  VARCHAR(95)   DEFAULT NULL,
 musiconhold VARCHAR(100)   DEFAULT NULL,
 pickupgroup VARCHAR(10)   DEFAULT NULL,
 qualify  CHAR(3)    DEFAULT NULL,
 regexten VARCHAR(80)   DEFAULT NULL,
 restrictcid CHAR(3)    DEFAULT NULL,
 rtptimeout CHAR(3)    DEFAULT NULL,
 rtpholdtimeout CHAR(3)    DEFAULT NULL,
 secret  VARCHAR(80)   DEFAULT NULL,
 setvar  VARCHAR(100)   DEFAULT NULL,
 disallow VARCHAR(100)   DEFAULT 'all',
 allow  VARCHAR(100)   DEFAULT 'g729,ilbc,gsm,ulaw,alaw',
 fullcontact VARCHAR(80) NOT NULL DEFAULT '',
 ipaddr  VARCHAR(15) NOT NULL DEFAULT '',
 port  INT(11)  NOT NULL DEFAULT 0,
 regserver VARCHAR(100)   DEFAULT NULL,
 regseconds INT(11)  NOT NULL DEFAULT 0,
 username VARCHAR(80) NOT NULL DEFAULT '',
 PRIMARY KEY (id)
 UNIQUE  (name)
);

CREATE INDEX ast_sip__idx__commented ON ast_sip(commented);

/*
 * Dialplan RealTime table.
 */
CREATE TABLE ast_exten (
 id  INTEGER,
 commented TINYINT(1) NOT NULL DEFAULT 0,
 context  VARCHAR(80) NOT NULL DEFAULT '',
 exten  VARCHAR(40) NOT NULL DEFAULT '',
 priority INT(11)  NOT NULL DEFAULT 0,
 app  VARCHAR(128) NOT NULL DEFAULT '',
 appdata  VARCHAR(128) NOT NULL DEFAULT '',
 PRIMARY KEY (id)
);
```

```
CREATE INDEX ast_exten__idx__commented   ON ast_exten(commented);
CREATE INDEX ast_exten__idx__context_exten_priority ON ast_exten(context, exten, priority);
```

Storing Voicemail in PostgreSQL via ODBC

How to get ODBC storage with PostgreSQL working with Voicemail

*Install PostgreSQL, PostgreSQL-devel, unixODBC, and unixODBC-devel, and PostgreSQL-ODBC. Make sure PostgreSQL is running and listening on a TCP socket.

*Log into your server as root, and then type:

```
[root@localhost ~]# su - postgres
```

This will log you into the system as the "postgres" user, so that you can create a new role and database within the PostgreSQL database system. At the new prompt, type:

```
$ createuser -s -D -R -l -P -e asterisk
Enter password for new role:
Enter it again:
```

Obviously you should enter a password when prompted. This creates the database role (or user).

Next we need to create the asterisk database. Type:

```
$ createdb -O asterisk -e asterisk
```

This creates the database and sets the owner of the database to the asterisk role.

Next, make sure that you are using md5 authentication for the database user. The line in my /var/lib/pgsql/data/pg_hba.conf looks like:

```
# "local" is for Unix domain socket connections only
local   asterisk    asterisk                           md5
local   all         all                               ident sameuser
# IPv4 local connections:
host    all         all         127.0.0.1/32          md5
```

As soon as you're done editing that file, log out as the postgres user.

- Make sure you have the PostgreSQL odbc driver setup in /etc/odbcinst.ini. Mine looks like:

```
[PostgreSQL]
Description     = ODBC for PostgreSQL
Driver          = /usr/lib/libodbcpsql.so
Setup           = /usr/lib/libodbcpsqlS.so
FileUsage       = 1
```

You can confirm that unixODBC is seeing the driver by typing:

```
[jsmith2@localhost tmp]$ odbcinst -q -d
[PostgreSQL]
```

- Setup a DSN in /etc/odbc.ini, pointing at the PostgreSQL database and driver. Mine looks like:

```
[testing]
Description           = ODBC Testing
Driver                = PostgreSQL
Trace                 = No
TraceFile             = sql.log
Database              = asterisk
Servername            = 127.0.0.1
UserName              = asterisk
Password              = supersecret
Port                  = 5432
ReadOnly              = No
RowVersioning         = No
ShowSystemTables      = No
ShowOidColumn         = No
FakeOidIndex          = No
ConnSettings          =
```

You can confirm that unixODBC sees your DSN by typing:

```
[jsmith2@localhost tmp]$ odbcinst -q -s
[testing]
```

- Test your database connectivity through ODBC. If this doesn't work, something is wrong with your ODBC setup.

```
[jsmith2@localhost tmp]$ echo "select 1" | isql -v testing
+---------------------------------------+
| Connected!                            |
|                                       |
| sql-statement                         |
| help [tablename]                      |
| quit                                  |
|                                       |
+---------------------------------------+
SQL> +------------+
| ?column?   |
+------------+
| 1          |
+------------+
SQLRowCount returns 1
1 rows fetched
```

If your ODBC connectivity to PostgreSQL isn't working, you'll see an error message instead, like this:

```
[jsmith2@localhost tmp]$ echo "select 1" | isql -v testing
[S1000][unixODBC]Could not connect to the server;
Could not connect to remote socket.
[ISQL]ERROR: Could not SQLConnect
bash: echo: write error: Broken pipe
```

- Compile Asterisk with support for ODBC voicemail. Go to your Asterisk source directory and run `make menuselect`. Under "Voicemail Build Options", enable "ODBC_STORAGE". See doc/README.odbcstorage for more information

Recompile Asterisk and install the new version.

- Once you've recompiled and re-installed Asterisk, check to make sure res_odbc.so has been compiled.

```
localhost*CLI> show modules like res_odbc.so
Module                         Description                                 Use Count
res_odbc.so                    ODBC Resource                               0
1 modules loaded
```

- Now it's time to get Asterisk configured. First, we need to tell Asterisk about our ODBC setup. Open /etc/asterisk/res_odbc.conf and add the following:

```
[postgres]
enabled => yes
dsn => testing
pre-connect => yes
```

- At the Asterisk CLI, unload and then load the res_odbc.so module. (You could restart Asterisk as well, but this way makes it easier to tell what's happening.) Notice how it says it's connected to "postgres", which is our ODBC connection as defined in res_odbc.conf, which points to the "testing" DSN in ODBC.

```
localhost*CLI> unload res_odbc.so
Jan  2 21:19:36 WARNING[8130]: res_odbc.c:498 odbc_obj_disconnect: res_odbc: disconnected 0 from postgres [testing]
Jan  2 21:19:36 NOTICE[8130]: res_odbc.c:589 unload_module: res_odbc unloaded.
localhost*CLI> load res_odbc.so
 Loaded /usr/lib/asterisk/modules/res_odbc.so => (ODBC Resource)
  == Parsing '/etc/asterisk/res_odbc.conf': Found
Jan  2 21:19:40 NOTICE[8130]: res_odbc.c:266 load_odbc_config: Adding ENV var: INFORMIXSERVER=my_special_database
Jan  2 21:19:40 NOTICE[8130]: res_odbc.c:266 load_odbc_config: Adding ENV var: INFORMIXDIR=/opt/informix
Jan  2 21:19:40 NOTICE[8130]: res_odbc.c:295 load_odbc_config: registered database handle 'postgres' dsn->[testing]
Jan  2 21:19:40 NOTICE[8130]: res_odbc.c:555 odbc_obj_connect: Connecting postgres
Jan  2 21:19:40 NOTICE[8130]: res_odbc.c:570 odbc_obj_connect: res_odbc: Connected to postgres [testing]
Jan  2 21:19:40 NOTICE[8130]: res_odbc.c:600 load_module: res_odbc loaded.
```

You can also check the status of your ODBC connection at any time from the Asterisk CLI:

```
localhost*CLI> odbc show
Name: postgres
DSN: testing
Connected: yes
```

- Now we can setup our voicemail table in PostgreSQL. Log into PostgreSQL and type (or copy and paste) the following:

```
--
-- First, let's create our large object type, called "lo"
--
CREATE FUNCTION loin (cstring) RETURNS lo AS 'oidin' LANGUAGE internal IMMUTABLE STRICT;
CREATE FUNCTION loout (lo) RETURNS cstring AS 'oidout' LANGUAGE internal IMMUTABLE STRICT;
CREATE FUNCTION lorecv (internal) RETURNS lo AS 'oidrecv' LANGUAGE internal IMMUTABLE STRICT;
CREATE FUNCTION losend (lo) RETURNS bytea AS 'oidrecv' LANGUAGE internal IMMUTABLE STRICT;

CREATE TYPE lo ( INPUT = loin, OUTPUT = loout, RECEIVE = lorecv, SEND = losend, INTERNALLENGTH = 4, PASSEDBYVALUE );
CREATE CAST (lo AS oid) WITHOUT FUNCTION AS IMPLICIT;
CREATE CAST (oid AS lo) WITHOUT FUNCTION AS IMPLICIT;

--
-- If we're not already using plpgsql, then let's use it!
--
CREATE TRUSTED LANGUAGE plpgsql;

--
-- Next, let's create a trigger to cleanup the large object table
-- whenever we update or delete a row from the voicemessages table
--

CREATE FUNCTION vm_lo_cleanup() RETURNS "trigger"
    AS $$
    declare
      msgcount INTEGER;
    begin
      --    raise notice 'Starting lo_cleanup function for large object with oid %',old.recording;
      -- If it is an update action but the BLOB (lo) field was not changed, dont do anything
      if (TG_OP = 'UPDATE') then
        if ((old.recording = new.recording) or (old.recording is NULL)) then
          raise notice 'Not cleaning up the large object table, as recording has not changed';
          return new;
        end if;
      end if;
      if (old.recording IS NOT NULL) then
        SELECT INTO msgcount COUNT(*) AS COUNT FROM voicemessages WHERE recording = old.recording;
        if (msgcount > 0) then
          raise notice 'Not deleting record from the large object table, as object is still referenced';
          return new;
        else
          perform lo_unlink(old.recording);
          if found then
            raise notice 'Cleaning up the large object table';
            return new;
          else
            raise exception 'Failed to cleanup the large object table';
            return old;
          end if;
        end if;
      else
        raise notice 'No need to cleanup the large object table, no recording on old row';
        return new;
      end if;
    end$$
    LANGUAGE plpgsql;

--
-- Now, let's create our voicemessages table
-- This is what holds the voicemail from Asterisk
--

CREATE TABLE voicemessages
(
  uniqueid serial PRIMARY KEY,
  msgnum int4,
  dir varchar(80),
  context varchar(80),
  macrocontext varchar(80),
  callerid varchar(40),
  origtime varchar(40),
  duration varchar(20),
  flag varchar(8),
  mailboxuser varchar(80),
  mailboxcontext varchar(80),
  recording lo,
  label varchar(30),
  "read" bool DEFAULT false
);

--
-- Let's not forget to make the voicemessages table use the trigger
--

CREATE TRIGGER vm_cleanup AFTER DELETE OR UPDATE ON voicemessages FOR EACH ROW EXECUTE PROCEDURE vm_lo_cleanup();
```

- Just as a sanity check, make sure you check the voicemessages table via the isql utility.

```
[jsmith2@localhost ODBC]$ echo "SELECT uniqueid, msgnum, dir, duration FROM voicemessages WHERE msgnum = 1" | isql testing
+---------------------------------------+
| Connected!                            |
|                                       |
| sql-statement                         |
| help [tablename]                      |
| quit                                  |
|                                       |
+---------------------------------------+
SQL>
+------------+------------+---------------------------------------------------------------------------------+--------------------
-+
| uniqueid   | msgnum     | dir                                                                             | duration
|
+------------+------------+---------------------------------------------------------------------------------+--------------------
-+
+------------+------------+---------------------------------------------------------------------------------+--------------------
-+
SQLRowCount returns 0
```

- Now we can finally configure voicemail in Asterisk to use our database. Open /etc/asterisk/voicemail.conf, and look in the [general] section. I've changed the format to gsm (as I can't seem to get WAV or wav working), and specify both the odbc connection and database table to use.

```
[general]
; Default formats for writing Voicemail
;format=g723sf|wav49|wav
format=gsm
odbcstorage=postgres
odbctable=voicemessages
```

You'll also want to create a new voicemail context called "odbctest" to do some testing, and create a sample mailbox inside that context. Add the following to the very bottom of voicemail.conf:

```
[odbctest]
101 => 5555,Example Mailbox
```

- Once you've updated voicemail.conf, let's make the changes take effect:

```
localhost*CLI> unload app_voicemail.so
  == Unregistered application 'VoiceMail'
  == Unregistered application 'VoiceMailMain'
  == Unregistered application 'MailboxExists'
  == Unregistered application 'VMAuthenticate'
localhost*CLI> load app_voicemail.so
 Loaded /usr/lib/asterisk/modules/app_voicemail.so => (Comedian Mail (Voicemail System))
  == Registered application 'VoiceMail'
  == Registered application 'VoiceMailMain'
  == Registered application 'MailboxExists'
  == Registered application 'VMAuthenticate'
  == Parsing '/etc/asterisk/voicemail.conf': Found
```

You can check to make sure your new mailbox exists by typing:

```
localhost*CLI> show voicemail users for odbctest
Context    Mbox  User                      Zone       NewMsg
odbctest   101   Example Mailbox                           0
```

- Now, let's add a new context called "odbc" to extensions.conf. We'll use these extensions to do some testing:

```
[odbc]
exten => 100,1,Voicemail(101@odbctest)
exten => 200,1,VoicemailMain(101@odbctest)
```

- Next, we need to point a phone at the odbc context. In my case, I've got a SIP phone called "linksys" that is registering to Asterisk, so I'm setting its context to the [odbc] context we created in the previous step. The relevant section of my sip.conf file looks like:

```
[linksys]
type=friend
secret=verysecret
disallow=all
allow=ulaw
allow=gsm
context=odbc
host=dynamic
qualify=yes
```

I can check to see that my linksys phone is registered with Asterisk correctly:

```
localhost*CLI> sip show peers like linksys
Name/username              Host            Dyn Nat ACL Port     Status
linksys/linksys            192.168.0.103    D          5060     OK (9 ms)
1 sip peers [1 online , 0 offline]
```

- At last, we're finally ready to leave a voicemail message and have it stored in our database! (Who'd have guessed it would be this much trouble?!?) Pick up the phone, dial extension 100, and leave yourself a voicemail message.
  In my case, this is what appeared on the Asterisk CLI:

```
localhost*CLI>
    -- Executing VoiceMail("SIP/linksys-10228cac", "101@odbctest") in new stack
    -- Playing 'vm-intro' (language 'en')
    -- Playing 'beep' (language 'en')
    -- Recording the message
    -- x=0, open writing:  /var/spool/asterisk/voicemail/odbctest/101/tmp/dlZunm format: gsm, 0x101f6534
    -- User ended message by pressing #
    -- Playing 'auth-thankyou' (language 'en')
  == Parsing '/var/spool/asterisk/voicemail/odbctest/101/INBOX/msg0000.txt': Found
```

Now, we can check the database and make sure the record actually made it into PostgreSQL, from within the psql utility.

```
[jsmith2@localhost ~]$ psql
Password:
Welcome to psql 8.1.4, the PostgreSQL interactive terminal.

Type:  \copyright for distribution terms
       \h for help with SQL commands
       \? for help with psql commands
       \g or terminate with semicolon to execute query
       \q to quit

asterisk=# SELECT * FROM voicemessages;
 uniqueid | msgnum |                       dir                        | context | macrocontext |       callerid        |
origtime  | duration | mailboxuser | mailboxcontext | recording | label | read | sip_id | pabx_id | iax_id
----------+--------+--------------------------------------------------+---------+--------------+-----------------------+---------
---+----------+-------------+----------------+-----------+-------+------+--------+---------+--------
       26 |      0 | /var/spool/asterisk/voicemail/odbctest/101/INBOX | odbc    |              | "linksys" <linksys> | 1167794179
| 7        | 101         | odbctest       | 16599     |       | f    |        |         |
(1 row)
```

Did you notice the the recording column is just a number? When a recording gets stuck in the database, the audio isn't actually stored in the voicemessages table. It's stored in a system table called the large object table. We can look in the large object table and verify that the object actually exists there:

```
asterisk=# \lo_list
    Large objects
  ID   | Description
-------+-------------
 16599 |
(1 row)
```

In my case, the OID is 16599. Your OID will almost surely be different. Just make sure the OID number in the recording column in the voicemessages table corresponds with a record in the large object table. (The trigger we added to
our voicemessages table was designed to make sure this is always the case.)

We can also pull a copy of the voicemail message back out of the database and write it to a file, to help us as we debug things:

```
asterisk=# \lo_export 16599 /tmp/odcb-16599.gsm
lo_export
```

We can even listen to the file from the Linux command line:

```
[jsmith2@localhost tmp]$ play /tmp/odcb-16599.gsm

Input Filename : /tmp/odcb-16599.gsm
Sample Size    : 8-bits
Sample Encoding: gsm
Channels       : 1
Sample Rate    : 8000

Time: 00:06.22 [00:00.00] of 00:00.00 (  0.0%) Output Buffer: 298.36K

Done.
```

- Last but not least, we can pull the voicemail message back out of the database by dialing extension 200 and entering "5555" at the password prompt. You should see something like this on the Asterisk CLI:

```
localhost*CLI>
    -- Executing VoiceMailMain("SIP/linksys-10228cac", "101@odbctest") in new stack
    -- Playing 'vm-password' (language 'en')
    -- Playing 'vm-youhave' (language 'en')
    -- Playing 'digits/1' (language 'en')
    -- Playing 'vm-INBOX' (language 'en')
    -- Playing 'vm-message' (language 'en')
    -- Playing 'vm-onefor' (language 'en')
    -- Playing 'vm-INBOX' (language 'en')
    -- Playing 'vm-messages' (language 'en')
    -- Playing 'vm-opts' (language 'en')
    -- Playing 'vm-first' (language 'en')
    -- Playing 'vm-message' (language 'en')
  == Parsing '/var/spool/asterisk/voicemail/odbctest/101/INBOX/msg0000.txt': Found
    -- Playing 'vm-received' (language 'en')
    -- Playing 'digits/at' (language 'en')
    -- Playing 'digits/10' (language 'en')
    -- Playing 'digits/16' (language 'en')
    -- Playing 'digits/p-m' (language 'en')
    -- Playing '/var/spool/asterisk/voicemail/odbctest/101/INBOX/msg0000' (language 'en')
    -- Playing 'vm-advopts' (language 'en')
    -- Playing 'vm-repeat' (language 'en')
    -- Playing 'vm-delete' (language 'en')
    -- Playing 'vm-toforward' (language 'en')
    -- Playing 'vm-savemessage' (language 'en')
    -- Playing 'vm-helpexit' (language 'en')
    -- Playing 'vm-goodbye' (language 'en')
```

That's it!

Jared Smith
2 Jan 2006
(updated 11 Mar 2007)

Distributed Device State

In the Key Concepts section States and Presence we discussed Asterisk Device State, Extension State and Hints.

The child pages here discuss the configuration of modules that allow device state to be distributed between multiple systems or instances of Asterisk.

In This Section

Corosync

Corosync

Corosync is an open source group messaging system typically used in clusters, cloud computing, and other high availability environments.

The project, at it's core, provides four C api features:

- A closed process group communication model with virtual synchrony guarantees for creating replicated state machines.
- A simple availability manager that restarts the application process when it has failed.
- A configuration and statistics in-memory database that provide the ability to set, retrieve, and receive change notifications of information.
- A quorum system that notifies applications when quorum is achieved or lost.

Corosync and Asterisk

Using Corosync together with res_corosync allows events to be shared amongst a local cluster of Asterisk servers. Specifically, the types of events that may be shared include:

- Device state
- Message Waiting Indication, or MWI (to allow voicemail to live on a server that is different from where the phones are registered)

Setup and Configuration

Corosync

- Installation Debian / Ubuntu

  ```
  apt-get install corosync corosync-dev
  ```

  Red Hat / Fedora

  ```
  yum install corosync corosynclib corosynclib-devel
  ```

- Authkey To create an authentication key for secure communications between your nodes you need to do this on, what will be, the active node.

  ```
  corosync-keygen
  ```

  This creates a key in /etc/corosync/authkey.

  ```
  asterisk_active:~# scp /etc/corosync/authkey asterisk_standby:
  ```

  Now, on the standby node, you'll need to stick the authkey in it's new home and fix it's permissions / ownership.

  ```
  asterisk_standby:~# mv ~/authkey /etc/corosync/authkey
  asterisk_standby:~# chown root:root /etc/corosync/authkey
  asterisk_standby:~# chmod 400 /etc/corosync/authkey
  ```

- /etc/corosync/corosync.conf The interface section under the totem block defines the communication path(s) to the other Corosync processes running on nodes within the cluster. These can be either IPv4 or IPv6 ip addresses but can not be mixed and matched within an interface. Adjustments can be made to the cluster settings based on your needs and installation environment.
  - IPv4 Active Node Example

    ```
    totem {
            version: 2
            token: 160
            token_retransmits_before_loss_const: 3
            join: 30
            consensus: 300
            vsftype: none
            max_messages: 20
            threads: 0
            nodeid: 1
            rrp_mode: none
            interface {
                    ringnumber: 0
                    bindnetaddr: 192.168.1.0
                    mcastaddr: 226.94.1.1
                    mcastport: 5405
            }
    }
    ```

    Standby Node Example

```
totem {
        version: 2
        token: 160
        token_retransmits_before_loss_const: 3
        join: 30
        consensus: 300
        vsftype: none
        max_messages: 20
        threads: 0
        nodeid: 2
        rrp_mode: none
        interface {
                ringnumber: 0
                bindnetaddr: 192.168.1.0
                mcastaddr: 226.94.1.1
                mcastport: 5405
        }
}
```

- Start Corosync

```
service corosync start
```

Asterisk

- Installation In your Asterisk source directory:

```
./configure
make
make install
```

- /etc/asterisk/res_corosync.conf

```
;
; Sample configuration file for res_corosync.
;
; This module allows events to be shared amongst a local cluster of
; Asterisk servers.  Specifically, the types of events that may be
; shared include:
;
;   - Device State (for shared presence information)
;
;   - Message Waiting Indication, or MWI (to allow Voicemail to live on
;     a server that is different from where the phones are registered)
;
; For more information about Corosync, see: http://www.corosync.org/
;

[general]

;
;  Publish Message Waiting Indication (MWI) events from this server to the
;  cluster.
publish_event = mwi
;
;  Subscribe to MWI events from the cluster.
subscribe_event = mwi
;
;  Publish Device State (presence) events from this server to the cluster.
publish_event = device_state
;
;  Subscribe to Device State (presence) events from the cluster.
subscribe_event = device_state
;
```

In the general section of the res_corosync.conf file we are specifying which events we'd like to publish and subscribe to (at the moment this is either device_state or mwi).

- Verifying Installation If everything is setup correctly, you should see this output after executing a 'corosync show members' on the Asterisk CLI.

```
*CLI> corosync show members

=============================================================
=== Cluster members =========================================
=============================================================
===
=== Node 1
=== --> Group: asterisk
=== --> Address 1: <host #1 ip goes here>
===
=============================================================
```

After starting Corosync and Asterisk on your second node, the 'corosync show members' output should look something like this:

```
*CLI> corosync show members

=============================================================
=== Cluster members =========================================
=============================================================
===
=== Node 1
=== --> Group: asterisk
=== --> Address 1: <host #1 ip goes here>
=== Node 2
=== --> Group: asterisk
=== --> Address 1: <host #2 ip goes here>
===
=============================================================
```

Distributed Device State with AIS

1. Introduction

Various changes have been made related to "event handling" in Asterisk. One of the most important things included in these changes is the ability to share certain events between servers. The two types of events that can currently be shared between servers are:

1. **MWI** - *Message Waiting Indication* - This gives you a high performance option for letting servers in a cluster be aware of changes in the state of a mailbox. Instead of having each server have to poll an ODBC database, this lets the server that actually made the change to the mailbox generate an event which will get distributed to the other servers that have subscribed to this information.
2. **Device State** - This lets servers in a local cluster inform each other about changes in the state of a device on that particular server. When the state of a device changes on any server, the overall state of that device across the cluster will get recalculated. So, any subscriptions to the state of a device, such as hints in the dialplan or an application like Queue() which reads device state, will then reflect the state of a device across a cluster.

2. OpenAIS Installation

Description

The current solution for providing distributed events with Asterisk is done by using the AIS (Application Interface Specification), which provides an API for a distributed event service. While this API is standardized, this code has been developed exclusively against the open source implementation of AIS called OpenAIS.

For more information about OpenAIS, visit their web site http://www.openais.org/.

Install Dependencies

- Ubuntu
  - libnss3-dev
- Fedora
  - nss-devel

Download

Download the latest versions of Corosync and OpenAIS from http://www.corosync.org/ and http://www.openais.org/.

Compile and Install

```
$ tar xvzf corosync-1.2.8.tar.gz
$ cd corosync-1.2.8
$ ./configure
$ make
$ sudo make install
```

```
$ tar xvzf openais-1.1.4.tar.gz
$ cd openais-1.1.4
$ ./configure
$ make
$ sudo make install
```

3. OpenAIS Configuration

Basic OpenAIS configuration to get this working is actually pretty easy. Start by copying in a sample configuration file for Corosync and OpenAIS.

```
$ sudo mkdir -p /etc/ais
$ cd openais-1.1.4
$ sudo cp conf/openais.conf.sample /etc/ais/openais.conf
$ sudo mkdir -p /etc/corosync
$ cd corosync-1.2.8
$ sudo cp conf/corosync.conf.sample /etc/corosync/corosync.conf
```

Now, edit openais.conf using the editor of your choice.

```
$ ${EDITOR:-vim} /etc/ais/openais.conf
```

The only section that you should need to change is the totem - interface section.

/etc/ais/openais.conf

```
totem {
    ...
    interface {
        ringnumber: 0
        bindnetaddr: 10.24.22.144
        mcastaddr: 226.94.1.1
        mcastport: 5405
    }
}
```

The default mcastaddr and mcastport is probably fine. You need to change the bindnetaddr to match the address of the network interface that this node will use to communicate with other nodes in the cluster.

Now, edit /etc/corosync/corosync.conf, as well. The same change will need to be made to the totem-interface section in that file.

4. Running OpenAIS

While testing, I recommend starting the aisexec application in the foreground so that you can see debug messages that verify that the nodes have discovered each other and joined the cluster.

```
$ sudo aisexec -f
```

For example, here is some sample output from the first server after starting aisexec on the second server:

```
Nov 13 06:55:30 corosync [CLM   ] CLM CONFIGURATION CHANGE
Nov 13 06:55:30 corosync [CLM   ] New Configuration:
Nov 13 06:55:30 corosync [CLM   ]       r(0) ip(10.24.22.144)
Nov 13 06:55:30 corosync [CLM   ]       r(0) ip(10.24.22.242)
Nov 13 06:55:30 corosync [CLM   ] Members Left:
Nov 13 06:55:30 corosync [CLM   ] Members Joined:
Nov 13 06:55:30 corosync [CLM   ]       r(0) ip(10.24.22.242)
Nov 13 06:55:30 corosync [TOTEM ] A processor joined or left the membership and a new membership was formed.
Nov 13 06:55:30 corosync [MAIN  ] Completed service synchronization, ready to provide service.
```

5. Installing Asterisk

Install Asterisk as usual. Just make sure that you run the configure script after OpenAIS gets installed. That way, it will find the AIS header files and will let you build the res_ais module. Check menuselect to make sure that res_ais is going to get compiled and installed.

```
$ cd asterisk-source
$ ./configure

$ make menuselect
  ---> Resource Modules
```

If you have existing configuration on the system being used for testing, just be sure to install the addition configuration file needed for res_ais.

```
$ sudo cp configs/ais.conf.sample /etc/asterisk/ais.conf
```

6. Configuring Asterisk

First, ensure that you have a unique "entity ID" set for each server.

```
*CLI> core show settings
   ...
   Entity ID:                   01:23:45:67:89:ab
```

The code will attempt to generate a unique entity ID for you by reading MAC addresses off of a network interface. However, you can also set it manually in the [options] section of asterisk.conf.

```
$ sudo ${EDITOR:-vim} /etc/asterisk/asterisk.conf
```

asterisk.conf

```
[options]

entity_id=01:23:45:67:89:ab
```

Edit the Asterisk ais.conf to enable distributed events. For example, if you would like to enable distributed device state, you should add the following section to the file:

```
$ sudo ${EDITOR:-vim} /etc/asterisk/ais.conf
```

/etc/asterisk/ais.conf

```
[device_state]
type=event_channel
publish_event=device_state
subscribe_event=device_state
```

For more information on the contents and available options in this configuration file, please see the sample configuration file:

```
$ cd asterisk-source
$ less configs/ais.conf.sample
```

7. Basic Testing of Asterisk with OpenAIS

If you have OpenAIS successfully installed and running, as well as Asterisk with OpenAIS support successfully installed, configured, and running, then you are ready to test out some of the AIS functionality in Asterisk.

The first thing to test is to verify that all of the nodes that you think should be in your cluster are actually there. There is an Asterisk CLI command which will list the current cluster members using the AIS Cluster Membership Service
(CLM).

```
*CLI> ais clm show members

=============================================================
=== Cluster Members =========================================
=============================================================
===
=== ---------------------------------------------------------
=== Node Name: 10.24.22.144
=== ==> ID: 0x9016180a
=== ==> Address: 10.24.22.144
=== ==> Member: Yes
=== ---------------------------------------------------------
===
=== ---------------------------------------------------------
=== Node Name: 10.24.22.242
=== ==> ID: 0xf216180a
=== ==> Address: 10.24.22.242
=== ==> Member: Yes
=== ---------------------------------------------------------
===
=============================================================
```

If you're having trouble getting the nodes of the cluster to see each other, make sure you do not have firewall rules that are blocking the multicast traffic that is used to communicate amongst the nodes.

The next thing to do is to verify that you have successfully configured some event channels in the Asterisk ais.conf file. This command is related to the event service (EVT), so like the previous command, uses the syntax: `ais <service name> <command>`.

```
*CLI> ais evt show event channels

=============================================================
=== Event Channels ==========================================
=============================================================
===
=== ---------------------------------------------------------
=== Event Channel Name: device_state
=== ==> Publishing Event Type: device_state
=== ==> Subscribing to Event Type: device_state
=== ---------------------------------------------------------
===
=============================================================
```

8. Testing Distributed Device State

The easiest way to test distributed device state is to use the DEVICE_STATE() diaplan function. For example, you could have the following piece of dialplan on every server:

/etc/asterisk/extensions.conf

```
[devstate_test]

exten => 1234,hint,Custom:mystate
```

Now, you can test that the cluster-wide state of "Custom:mystate" is what you would expect after going to the CLI of each server and adjusting the state.

```
server1*CLI> dialplan set global DEVICE_STATE(Custom:mystate) NOT_INUSE
   ...

server2*CLI> dialplan set global DEVICE_STATE(Custom:mystate) INUSE
   ...
```

Various combinations of setting and checking the state on different servers can be used to verify that it works as expected. Also, you can see the status of the hint on each server, as well, to see how extension state would reflect the
state change with distributed device state:

```
server2*CLI> core show hints
    -= Registered Asterisk Dial Plan Hints =-
                   1234@devstate_test       : Custom:mystate        State:InUse           Watchers  0
```

One other helpful thing here during testing and debugging is to enable debug logging. To do so, enable debug on the console in /etc/asterisk/logger.conf. Also, enable debug at the Asterisk CLI.

```
*CLI> core set debug 1
```

When you have this debug enabled, you will see output during the processing of every device state change. The important thing to look for is where the known state of the device for each server is added together to determine the overall
state.

Distributed Device State with XMPP PubSub

1. Introduction

This document describes installing and utilizing XMPP PubSub events to distribute device state and message waiting indication (MWI) events between servers. The difference between this method and OpenAIS (see Distributed Device State with AIS) is that OpenAIS can only be used in low latency networks; meaning only on the LAN, and not across the internet.

If you plan on distributing device state or MWI across the internet, then you will require the use of XMPP PubSub events.

2. Tigase Installation

Description

Currently the only server supported for XMPP PubSub events is the Tigase open source XMPP/Jabber environment. This is the server that the various Asterisk servers will connect to in order to distribute the events. The Tigase server can even be clustered in order to provide high availability for your device state; however, that is beyond the scope of this document.

For more information about Tigase, visit their web site http://www.tigase.org/.

Download

To download the Tigase environment, get the latest version at http://www.tigase.org/content/tigase-downloads. Some distributions have Tigase packaged, as well.

Install

The Tigase server requires a working Java environment, including both a JRE (Java Runtime Environment) and a JDK (Java Development Kit), currently at least version 1.6.

For more information about how to install Tigase, see the web site http://www.tigase.org/content/quick-start.

2.1. Tigase Configuration

While installing Tigase, be sure you enable the PubSub module. Without it, the PubSub events won't be accepted by the server, and your device state will not be distributed.

There are a couple of things you need to configure in Tigase before you start it in order for Asterisk to connect. The first thing we need to do is generate the self-signed certificate. To do this we use the keytool application. More
information can be found here http://www.tigase.org/content/server-certificate.

Generating the keystore file

Generally, we need to run the following commands to generate a new keystore file.

```
# cd /opt/Tigase-4.3.1-b1858/certs
```

Be sure to change the 'yourdomain' to your domain.

```
# keytool -genkey -alias yourdomain -keystore rsa-keystore -keyalg RSA -sigalg MD5withRSA
```

The keytool application will then ask you for a password. Use the password 'keystore' as this is the default password that Tigase will use to load the keystore file.

You then need to specify your domain as the first value to be entered in the security certificate.

```
What is your first and last name?
  [Unknown]: asterisk.mydomain.tld
What is the name of your organizational unit?
  [Unknown]:
What is the name of your organization?
  [Unknown]:
What is the name of your City or Locality?
  [Unknown]:
What is the name of your State or Province?
  [Unknown]:
What is the two-letter country code for this unit?
  [Unknown]:
Is CN=asterisk.mydomain.tld, OU=Unknown, O=Unknown, L=Unknown, ST=Unknown, C=Unknown correct?
  [no]: yes
```

You will then be asked for another password, in which case you must just press enter for the same password as Tigase will not work without them being the same.

```
Enter key password for <mykey>
             (RETURN if same as keystore password):
```

Configuring init.properties

The next step is to configure the init.properties file which is used by Tigase to generate the tigase.xml file. Whenever you change the init.properties file because sure to remove the current tigase.xml file so that it will be regenerated at start up.

```
# cd /opt/Tigase-4.3.1-b1858/etc
```

Then edit the init.properties file and add the following:

```
config-type=--gen-config-def
--admins=admin@asterisk.mydomain.tld
--virt-hosts=asterisk.mydomain.tld
--debug=server
--user-db=derby
--user-db-uri=jdbc:derby:/opt/Tigase-4.3.1-b1858
--comp-name-1=pubsub
--comp-class-1=tigase.pubsub.PubSubComponent
```

Be sure to change the domain in the --admin and --virt-hosts options. The most important lines are --comp-name-1 and --comp-class-1 which tell Tigase to load the PubSub module.

2.2. Running Tigase

You can then start the Tigase server with the tigase.sh script.

```
# cd /opt/Tigase-4.3.1-b1858
# ./scripts/tigase.sh start etc/tigase.conf
```

2.3. Adding Buddies to Tigase

At this time, Asterisk is not able to automatically register your peers for you, so you'll need to use an external application to do the initial registration.

Pidgin is an excellent multi-protocol instant messenger application which supports XMPP. It runs on Linux, Windows, and OSX, and is open source. You can get Pidgin from http://www.pidgin.im

Then add the two buddies we'll use in Asterisk with Pidgin by connecting to the Tigase server. For more information about how to register new buddies, see the Pidgin documentation.

Once the initial registration is done and loaded into Tigase, you no longer need to worry about using Pidgin. Asterisk will then be able to load the peers into memory at start up.

The example peers we've used in the following documentation for our two nodes are:

```
server1@asterisk.mydomain.tld/astvoip1
server2@asterisk.mydomain.tld/astvoip2
```

3. Installing Asterisk

Install Asterisk as usual. However, you'll need to make sure you have the res_jabber module compiled, which requires the iksemel development library. Additionally, be sure you have the OpenSSL development library installed so you can connect securly to the Tigase server.

Make sure you check menuselect that res_jabber is selected so that it will compile.

```
# cd asterisk-source
# ./configure

# make menuselect
  ---> Resource Modules
```

If you don't have jabber.conf in your existing configuration, because sure to copy the sample configuration file there.

```
# cd configs
# cp jabber.conf.sample /etc/asterisk/jabber.conf
```

3.1. Configuring Asterisk

We then need to configure our servers to communicate with the Tigase server. We need to modify the jabber.conf file on the servers. The configurations below are for a 2 server setup, but could be expanded for additional servers easily.

The key note here is to note that the pubsub_node option needs to start with pubsub, so for example, pubsub.asterisk.mydomain.tld. Without the 'pubsub' your Asterisk system will not be able to distribute events.

Additionally, you will need to specify each of the servers you need to connec to using the 'buddy' option.

*Asterisk Server 1

jabber.conf on server1

```
[general]
debug=no                                ;;Turn on debugging by default.
;autoprune=yes                          ;;Auto remove users from buddy list. Depending on
your
                                        ;;setup (ie, using your personal Gtalk account
for a test)
                                        ;;you might lose your contacts list. Default is
'no'.
autoregister=yes                        ;;Auto register users from buddy list.
;collection_nodes=yes                   ;;Enable support for XEP-0248 for use with
                                        ;;distributed device state.  Default is 'no'.
;pubsub_autocreate=yes                  ;;Whether or not the PubSub server supports/is
using
                                        ;;auto-create for nodes.  If it is, we have to
                                        ;;explicitly pre-create nodes before publishing
them.
                                        ;;Default is 'no'.

[asterisk]
type=client
serverhost=asterisk.mydomain.tld
pubsub_node=pubsub.asterisk.mydomain.tld
username=server1@asterisk.mydomain.tld/astvoip1
secret=welcome
distribute_events=yes
status=available
usetls=no
usesasl=yes
buddy=server2@asterisk.mydomain.tld/astvoip2
```

Asterisk Server 2

jabber.conf on server2

```
[general]
debug=yes    ;;Turn on debugging by default.
;autoprune=yes    ;;Auto remove users from buddy list. Depending on your
     ;;setup (ie, using your personal Gtalk account for a test)
     ;;you might lose your contacts list. Default is 'no'.
autoregister=yes   ;;Auto register users from buddy list.
;collection_nodes=yes   ;;Enable support for XEP-0248 for use with
     ;;distributed device state.  Default is 'no'.
;pubsub_autocreate=yes   ;;Whether or not the PubSub server supports/is using
     ;;auto-create for nodes.  If it is, we have to
     ;;explicitly pre-create nodes before publishing them.
     ;;Default is 'no'.

[asterisk]
type=client
serverhost=asterisk.mydomain.tld
pubsub_node=pubsub.asterisk.mydomain.tld
username=server2@asterisk.mydomain.tld/astvoip2
secret=welcome
distribute_events=yes
status=available
usetls=no
usesasl=yes
buddy=server1@asterisk.mydomain.tld/astvoip1
```

4. Basic Testing of Asterisk with XMPP PubSub

Once you have Asterisk installed with XMPP PubSub, it is time to test it out.

We need to start up our first server and make sure we get connected to the XMPP server. We can verify this with an Asterisk console command to determine if we're connected.

On Asterisk 1 we can run 'jabber show connected' to verify we're connected to the XMPP server.

```
*CLI> jabber show connected
Jabber Users and their status:
       User: server1@asterisk.mydomain.tld/astvoip1     - Connected
----
   Number of users: 1
```

The command above has given us output which verifies we've connected our first server.

We can then check the state of our buddies with the 'jabber show buddies' CLI command.

```
*CLI> jabber show buddies
Jabber buddy lists
Client: server1@asterisk.mydomain.tld/astvoip1
 Buddy: server2@asterisk.mydomain.tld
  Resource: None
 Buddy: server2@asterisk.mydomain.tld/astvoip2
  Resource: None
```

The output above tells us we're not connected to any buddies, and thus we're not distributing state to anyone (or getting it from anyone). That makes sense since we haven't yet started our other server.

Now, let's start the other server and verify the servers are able to establish a connection between each other.

On Asterisk 2, again we run the 'jabber show connected' command to make sure we've connected successfully to the XMPP server.

```
*CLI> jabber show connected
Jabber Users and their status:
       User: server2@asterisk.mydomain.tld/astvoip2     - Connected
----
   Number of users: 1
```

And now we can check the status of our buddies.

```
*CLI> jabber show buddies
Jabber buddy lists
Client: server2@scooter/astvoip2
 Buddy: server1@asterisk.mydomain.tld
  Resource: astvoip1
   node: http://www.asterisk.org/xmpp/client/caps
   version: asterisk-xmpp
   Jingle capable: yes
  Status: 1
  Priority: 0
 Buddy: server1@asterisk.mydomain.tld/astvoip1
  Resource: None
```

Excellent! So we're connected to the buddy on Asterisk 1, and we could run the same command on Asterisk 1 to verify the buddy on Asterisk 2 is seen.

5. Testing Distributed Device State

The easiest way to test distributed device state is to use the DEVICE_STATE() diaplan function. For example, you could have the following piece of dialplan on every server:

```
[devstate_test]

exten => 1234,hint,Custom:mystate

exten => set_inuse,1,Set(DEVICE_STATE(Custom:mystate)=INUSE)
exten => set_not_inuse,1,Set(DEVICE_STATE(Custom:mystate)=NOT_INUSE)

exten => check,1,NoOp(Custom:mystate is ${DEVICE_STATE(Custom:mystate)})
```

Now, you can test that the cluster-wide state of "Custom:mystate" is what you would expect after going to the CLI of each server and adjusting the state.

```
server1*CLI> console dial set_inuse@devstate_test
   ...

server2*CLI> console dial check@devstate_test
    -- Executing [check@devstate_test:1] NoOp("OSS/dsp", "Custom:mystate is INUSE") in new stack
```

Various combinations of setting and checking the state on different servers can be used to verify that it works as expected. Also, you can see the status of the hint on each server, as well, to see how extension state would reflect the
state change with distributed device state:

```
server2*CLI> core show hints
    -= Registered Asterisk Dial Plan Hints =-
                   1234@devstate_test       : Custom:mystate        State:InUse           Watchers  0
```

One other helpful thing here during testing and debugging is to enable debug logging. To do so, enable debug on the console in /etc/asterisk/logger.conf. Also, enable debug at the Asterisk CLI.

```
*CLI> core set debug 1
```

When you have this debug enabled, you will see output during the processing of every device state change. The important thing to look for is where the known state of the device for each server is added together to determine the overall
state.

6. Notes On Large Installations

On larger installations where you want a fully meshed network of buddies (i.e. all servers have all the buddies of the remote servers), you may want some method of distributing those buddies automatically so you don't need to modify
all servers (N+1) every time you add a new server to the cluster.

The problem there is that you're confined by what's allowed in XEP-0060, and unfortunately that means modifying affiliations by individual JID (as opposed to the various subscription access models, which are more flexible).

See here for details http://xmpp.org/extensions/xep-0060.html#owner-affiliations

One method for making this slightly easier is to utilize the #exec functionality in configuration files, and dynamically generate the buddies via script that pulls the information from a database, or to #include a file which is automatically generated on all the servers when you add a new node to the cluster.

Unfortunately this still requires a reload of res_jabber.so on all the servers, but this could also be solved through the use of the Asterisk Manager Interface (AMI).

So while this is not the ideal situation, it is programmatically solvable with existing technologies and features found in Asterisk today.

Simple Network Management Protocol (SNMP) Support

Asterisk SNMP Support

Rudimentary support for SNMP access to Asterisk is available. To build this, one needs to have Net-SNMP development headers and libraries on the build system, including any libraries Net-SNMP depends on.

Note that on some (many?) Linux-distributions the dependency list in the net-snmp-devel list is not complete, and additional packages will need to be installed. This is usually seen as configure failing to detect net-snmp-devel as the configure script does a sanity check of the net-snmp build environment, based on the output of 'net-snmp-config --agent-libs'.

To see what your distribution requires, run:

```
'net-snmp-config --agent-libs'.
```

You will receive a response similar to the following:

```
 -L/usr/lib -lnetsnmpmibs -lnetsnmpagent -lnetsnmphelpers -lnetsnmp -ldl
 -lrpm -lrpmio -lpopt -lz -lcrypto -lm -lsensors -L/usr/lib/lib -lwrap
 -Wl,-E -Wl,-rpath,/usr/lib/perl5/5.8.8/i386-linux-thread-multi/CORE
 -L/usr/local/lib
 /usr/lib/perl5/5.8.8/i386-linux-thread-multi/auto/DynaLoader/DynaLoader.a
 -L/usr/lib/perl5/5.8.8/i386-linux-thread-multi/CORE -lperl -lresolv -lnsl
 -ldl -lm -lcrypt -lutil -lpthread -lc
```

The packages required may include the following:

- bzip2-devel
- lm_sensors-devel
- newt-devel

SNMP support comes in two varieties – as a sub-agent to a running SNMP daemon using the AgentX protocol, or as a full standalone agent. If you wish to run a full standalone agent, Asterisk must run as root in
order to bind to port 161.

Configuring access when running as a full agent is something that is left as an exercise to the reader.

To enable access to the Asterisk SNMP subagent from a master SNMP daemon, one will need to enable AgentX support, and also make sure that Asterisk will be able to access the Unix domain socket. One way of doing this is to add the following to /etc/snmp/snmpd.conf:

```


# Enable AgentX support
master agentx

# Set permissions on AgentX socket and containing
# directory such that process in group 'asterisk'
# will be able to connect
agentXPerms  0660 0550 nobody asterisk
```

This assumes that you run Asterisk under group 'asterisk' (and does not care what user you run as).

Asterisk MIB Definitions

```
ASTERISK-MIB DEFINITIONS ::= BEGIN

IMPORTS
 OBJECT-TYPE, MODULE-IDENTITY, Integer32, Counter32, TimeTicks,
 Unsigned32, Gauge32
  FROM SNMPv2-SMI

 TEXTUAL-CONVENTION, DisplayString, TruthValue
  FROM SNMPv2-TC

 digium
  FROM DIGIUM-MIB;

asterisk MODULE-IDENTITY
 LAST-UPDATED "200806202025Z"
 ORGANIZATION "Digium, Inc."
 CONTACT-INFO
  "Mark A. Spencer
  Postal: Digium, Inc.
          445 Jan Davis Drive
          Huntsville, AL 35806
          USA
     Tel: +1 256 428 6000
   Email: markster@digium.com

  Thorsten Lockert
  Postal: Voop AS
          Boehmergaten 42
   NO-5057 Bergen
   Norway
     Tel: +47 5598 7200
   Email: tholo@voop.no"
 DESCRIPTION
  "Asterisk is an Open Source PBX.  This MIB defined
  objects for managing Asterisk instances."
 REVISION "200806202025Z"
 DESCRIPTION
  "smilint police --
   Add missing imports; fix initial capitalization
   of enumeration elements; add missing range
   restrictions for Integer32 indices, correct
   spelling of astChanCidANI in its definition.
   Addresses bug 12905. - jeffg@opennms.org"
 REVISION "200708211450Z"
 DESCRIPTION
  "Add total and current call counter statistics."
 REVISION "200603061840Z"
 DESCRIPTION
  "Change audio codec identification from 3kAudio to
  Audio3k to conform better with specification.

  Expand on contact information."
 REVISION "200602041900Z"
 DESCRIPTION
  "Initial published revision."
 ::= { digium 1 }

asteriskVersion  OBJECT IDENTIFIER ::= { asterisk 1 }
asteriskConfiguration OBJECT IDENTIFIER ::= { asterisk 2 }
asteriskModules  OBJECT IDENTIFIER ::= { asterisk 3 }
asteriskIndications OBJECT IDENTIFIER ::= { asterisk 4 }
asteriskChannels OBJECT IDENTIFIER ::= { asterisk 5 }

-- asteriskVersion

astVersionString OBJECT-TYPE
 SYNTAX  DisplayString
 MAX-ACCESS read-only
 STATUS  current
 DESCRIPTION
  "Text version string of the version of Asterisk that
  the SNMP Agent was compiled to run against."
 ::= { asteriskVersion 1 }

astVersionTag OBJECT-TYPE
 SYNTAX  Unsigned32
 MAX-ACCESS read-only
 STATUS  current
 DESCRIPTION
  "SubVersion revision of the version of Asterisk that
  the SNMP Agent was compiled to run against -- this is
  typically 0 for release-versions of Asterisk."
 ::= { asteriskVersion 2 }

-- asteriskConfiguration
```

```
astConfigUpTime OBJECT-TYPE
 SYNTAX  TimeTicks
 MAX-ACCESS read-only
 STATUS  current
 DESCRIPTION
  "Time ticks since Asterisk was started."
 ::= { asteriskConfiguration 1 }

astConfigReloadTime OBJECT-TYPE
 SYNTAX  TimeTicks
 MAX-ACCESS read-only
 STATUS  current
 DESCRIPTION
  "Time ticks since Asterisk was last reloaded."
 ::= { asteriskConfiguration 2 }

astConfigPid OBJECT-TYPE
 SYNTAX  Integer32
 MAX-ACCESS read-only
 STATUS  current
 DESCRIPTION
  "The process id of the running Asterisk process."
 ::= { asteriskConfiguration 3 }

astConfigSocket OBJECT-TYPE
 SYNTAX  DisplayString
 MAX-ACCESS read-only
 STATUS  current
 DESCRIPTION
  "The control socket for giving Asterisk commands."
 ::= { asteriskConfiguration 4 }

astConfigCallsActive OBJECT-TYPE
 SYNTAX  Gauge32
 MAX-ACCESS read-only
 STATUS  current
 DESCRIPTION
  "The number of calls currently active on the Asterisk PBX."
 ::= { asteriskConfiguration 5 }

astConfigCallsProcessed OBJECT-TYPE
 SYNTAX  Counter32
 MAX-ACCESS read-only
 STATUS  current
 DESCRIPTION
  "The total number of calls processed through the Asterisk PBX since last
  restart."
 ::= { asteriskConfiguration 6 }

-- asteriskModules

astNumModules OBJECT-TYPE
 SYNTAX  Integer32
 MAX-ACCESS read-only
 STATUS  current
 DESCRIPTION
  "Number of modules currently loaded into Asterisk."
 ::= { asteriskModules 1 }

-- asteriskIndications

astNumIndications OBJECT-TYPE
 SYNTAX  Integer32
 MAX-ACCESS read-only
 STATUS  current
 DESCRIPTION
  "Number of indications currently defined in Asterisk."
 ::= { asteriskIndications 1 }

astCurrentIndication OBJECT-TYPE
 SYNTAX  DisplayString
 MAX-ACCESS read-only
 STATUS  current
 DESCRIPTION
  "Default indication zone to use."
 ::= { asteriskIndications 2 }

astIndicationsTable OBJECT-TYPE
 SYNTAX  SEQUENCE OF AstIndicationsEntry
 MAX-ACCESS not-accessible
 STATUS  current
 DESCRIPTION
  "Table with all the indication zones currently know to
  the running Asterisk instance."
 ::= { asteriskIndications 3 }

astIndicationsEntry OBJECT-TYPE
 SYNTAX  AstIndicationsEntry
 MAX-ACCESS not-accessible
 STATUS  current
 DESCRIPTION
  "Information about a single indication zone."
 INDEX  { astIndIndex }
 ::= { astIndicationsTable 1 }

AstIndicationsEntry ::= SEQUENCE {
 astIndIndex  Integer32,
 astIndCountry  DisplayString,
 astIndAlias  DisplayString,
 astIndDescription DisplayString
}

astIndIndex OBJECT-TYPE
 SYNTAX  Integer32 (1 .. 2147483647)
 MAX-ACCESS read-only
 STATUS  current
 DESCRIPTION
  "Numerical index into the table of indication zones."
 ::= { astIndicationsEntry 1 }

astIndCountry OBJECT-TYPE
 SYNTAX  DisplayString
 MAX-ACCESS read-only
 STATUS  current
 DESCRIPTION
  "Country for which the indication zone is valid,
  typically this is the ISO 2-letter code of the country."
 ::= { astIndicationsEntry 2 }

astIndAlias OBJECT-TYPE
 SYNTAX  DisplayString
 MAX-ACCESS read-only
 STATUS  current
 DESCRIPTION
  ""
 ::= { astIndicationsEntry 3 }

astIndDescription OBJECT-TYPE
 SYNTAX  DisplayString
 MAX-ACCESS read-only
 STATUS  current
 DESCRIPTION
  "Description of the indication zone, usually the full
  name of the country it is valid for."
 ::= { astIndicationsEntry 4 }

-- asteriskChannels

astNumChannels OBJECT-TYPE
 SYNTAX  Gauge32
 MAX-ACCESS read-only
 STATUS  current
 DESCRIPTION
  "Current number of active channels."
 ::= { asteriskChannels 1 }

astChanTable OBJECT-TYPE
 SYNTAX  SEQUENCE OF AstChanEntry
 MAX-ACCESS not-accessible
 STATUS  current
 DESCRIPTION
  "Table with details of the currently active channels
  in the Asterisk instance."
 ::= { asteriskChannels 2 }

astChanEntry OBJECT-TYPE
 SYNTAX  AstChanEntry
 MAX-ACCESS not-accessible
 STATUS  current
 DESCRIPTION
  "Details of a single channel."
 INDEX  { astChanIndex }
 ::= { astChanTable 1 }

AstChanEntry ::= SEQUENCE {
 astChanIndex  Integer32,
 astChanName  DisplayString,
 astChanLanguage  DisplayString,
 astChanType  DisplayString,
 astChanMusicClass DisplayString,
 astChanBridge  DisplayString,
 astChanMasq  DisplayString,
 astChanMasqr  DisplayString,
 astChanWhenHangup TimeTicks,
 astChanApp  DisplayString,
 astChanData  DisplayString,
 astChanContext  DisplayString,
 astChanMacroContext DisplayString,
 astChanMacroExten DisplayString,
 astChanMacroPri  Integer32,
 astChanExten  DisplayString,
 astChanPri  Integer32,
 astChanAccountCode DisplayString,
 astChanForwardTo DisplayString,
 astChanUniqueId  DisplayString,
 astChanCallGroup Unsigned32,
 astChanPickupGroup Unsigned32,
 astChanState  INTEGER,
 astChanMuted  TruthValue,
 astChanRings  Integer32,
 astChanCidDNID  DisplayString,
 astChanCidNum  DisplayString,
 astChanCidName  DisplayString,
 astChanCidANI  DisplayString,
 astChanCidRDNIS  DisplayString,
 astChanCidPresentation DisplayString,
 astChanCidANI2  Integer32,
 astChanCidTON  Integer32,
 astChanCidTNS  Integer32,
 astChanAMAFlags  INTEGER,
 astChanADSI  INTEGER,
 astChanToneZone  DisplayString,
 astChanHangupCause INTEGER,
 astChanVariables DisplayString,
 astChanFlags  BITS,
 astChanTransferCap INTEGER
}

astChanIndex OBJECT-TYPE
 SYNTAX  Integer32 (1 .. 2147483647)
 MAX-ACCESS read-only
 STATUS  current
 DESCRIPTION
  "Index into the channel table."
 ::= { astChanEntry 1 }

astChanName OBJECT-TYPE
 SYNTAX  DisplayString
 MAX-ACCESS read-only
 STATUS  current
 DESCRIPTION
  "Name of the current channel."
 ::= { astChanEntry 2 }

astChanLanguage OBJECT-TYPE
 SYNTAX  DisplayString
 MAX-ACCESS read-only
 STATUS  current
 DESCRIPTION
  "Which language the current channel is configured to
  use -- used mainly for prompts."
 ::= { astChanEntry 3 }

astChanType OBJECT-TYPE
 SYNTAX  DisplayString
 MAX-ACCESS read-only
 STATUS  current
 DESCRIPTION
  "Underlying technology for the current channel."
 ::= { astChanEntry 4 }

astChanMusicClass OBJECT-TYPE
 SYNTAX  DisplayString
 MAX-ACCESS read-only
 STATUS  current
 DESCRIPTION
  "Music class to be used for Music on Hold for this
  channel."
 ::= { astChanEntry 5 }

astChanBridge OBJECT-TYPE
 SYNTAX  DisplayString
 MAX-ACCESS read-only
 STATUS  current
 DESCRIPTION
  "Which channel this channel is currently bridged (in a
  conversation) with."
 ::= { astChanEntry 6 }

astChanMasq OBJECT-TYPE
 SYNTAX  DisplayString
 MAX-ACCESS read-only
 STATUS  current
 DESCRIPTION
  "Channel masquerading for us."
 ::= { astChanEntry 7 }
```

```
astChanMasqr OBJECT-TYPE
 SYNTAX  DisplayString
 MAX-ACCESS read-only
 STATUS  current
 DESCRIPTION
  "Channel we are masquerading for."
 ::= { astChanEntry 8 }

astChanWhenHangup OBJECT-TYPE
 SYNTAX  TimeTicks
 MAX-ACCESS read-only
 STATUS  current
 DESCRIPTION
  "How long until this channel will be hung up."
 ::= { astChanEntry 9 }

astChanApp OBJECT-TYPE
 SYNTAX  DisplayString
 MAX-ACCESS read-only
 STATUS  current
 DESCRIPTION
  "Current application for the channel."
 ::= { astChanEntry 10 }

astChanData OBJECT-TYPE
 SYNTAX  DisplayString
 MAX-ACCESS read-only
 STATUS  current
 DESCRIPTION
  "Arguments passed to the current application."
 ::= { astChanEntry 11 }

astChanContext OBJECT-TYPE
 SYNTAX  DisplayString
 MAX-ACCESS read-only
 STATUS  current
 DESCRIPTION
  "Current extension context."
 ::= { astChanEntry 12 }

astChanMacroContext OBJECT-TYPE
 SYNTAX  DisplayString
 MAX-ACCESS read-only
 STATUS  current
 DESCRIPTION
  "Current macro context."
 ::= { astChanEntry 13 }

astChanMacroExten OBJECT-TYPE
 SYNTAX  DisplayString
 MAX-ACCESS read-only
 STATUS  current
 DESCRIPTION
  "Current macro extension."
 ::= { astChanEntry 14 }

astChanMacroPri OBJECT-TYPE
 SYNTAX  Integer32
 MAX-ACCESS read-only
 STATUS  current
 DESCRIPTION
  "Current macro priority."
 ::= { astChanEntry 15 }

astChanExten OBJECT-TYPE
 SYNTAX  DisplayString
 MAX-ACCESS read-only
 STATUS  current
 DESCRIPTION
  "Current extension."
 ::= { astChanEntry 16 }

astChanPri OBJECT-TYPE
 SYNTAX  Integer32
 MAX-ACCESS read-only
 STATUS  current
 DESCRIPTION
  "Current priority."
 ::= { astChanEntry 17 }

astChanAccountCode OBJECT-TYPE
 SYNTAX  DisplayString
 MAX-ACCESS read-only
 STATUS  current
 DESCRIPTION
  "Account Code for billing."
 ::= { astChanEntry 18 }
```

```
astChanForwardTo OBJECT-TYPE
 SYNTAX  DisplayString
 MAX-ACCESS read-only
 STATUS  current
 DESCRIPTION
  "Where to forward to if asked to dial on this
  interface."
 ::= { astChanEntry 19 }

astChanUniqueId OBJECT-TYPE
 SYNTAX  DisplayString
 MAX-ACCESS read-only
 STATUS  current
 DESCRIPTION
  "Unique Channel Identifier."
 ::= { astChanEntry 20 }

astChanCallGroup OBJECT-TYPE
 SYNTAX  Unsigned32
 MAX-ACCESS read-only
 STATUS  current
 DESCRIPTION
  "Call Group."
 ::= { astChanEntry 21 }

astChanPickupGroup OBJECT-TYPE
 SYNTAX  Unsigned32
 MAX-ACCESS read-only
 STATUS  current
 DESCRIPTION
  "Pickup Group."
 ::= { astChanEntry 22 }

astChanState OBJECT-TYPE
 SYNTAX  INTEGER {
  stateDown(0),
  stateReserved(1),
  stateOffHook(2),
  stateDialing(3),
  stateRing(4),
  stateRinging(5),
  stateUp(6),
  stateBusy(7),
  stateDialingOffHook(8),
  statePreRing(9)
 }
 MAX-ACCESS read-only
 STATUS  current
 DESCRIPTION
  "Channel state."
 ::= { astChanEntry 23 }

astChanMuted OBJECT-TYPE
 SYNTAX  TruthValue
 MAX-ACCESS read-only
 STATUS  current
 DESCRIPTION
  "Transmission of voice data has been muted."
 ::= { astChanEntry 24 }

astChanRings OBJECT-TYPE
 SYNTAX  Integer32
 MAX-ACCESS read-only
 STATUS  current
 DESCRIPTION
  "Number of rings so far."
 ::= { astChanEntry 25 }

astChanCidDNID OBJECT-TYPE
 SYNTAX  DisplayString
 MAX-ACCESS read-only
 STATUS  current
 DESCRIPTION
  "Dialled Number ID."
 ::= { astChanEntry 26 }

astChanCidNum OBJECT-TYPE
 SYNTAX  DisplayString
 MAX-ACCESS read-only
 STATUS  current
 DESCRIPTION
  "Caller Number."
 ::= { astChanEntry 27 }

astChanCidName OBJECT-TYPE
 SYNTAX  DisplayString
 MAX-ACCESS read-only
 STATUS  current
 DESCRIPTION
```

```
  "Caller Name."
 ::= { astChanEntry 28 }

astChanCidANI OBJECT-TYPE
 SYNTAX  DisplayString
 MAX-ACCESS read-only
 STATUS  current
 DESCRIPTION
  "ANI"
 ::= { astChanEntry 29 }

astChanCidRDNIS OBJECT-TYPE
 SYNTAX  DisplayString
 MAX-ACCESS read-only
 STATUS  current
 DESCRIPTION
  "Redirected Dialled Number Service."
 ::= { astChanEntry 30 }

astChanCidPresentation OBJECT-TYPE
 SYNTAX  DisplayString
 MAX-ACCESS read-only
 STATUS  current
 DESCRIPTION
  "Number Presentation/Screening."
 ::= { astChanEntry 31 }

astChanCidANI2 OBJECT-TYPE
 SYNTAX  Integer32
 MAX-ACCESS read-only
 STATUS  current
 DESCRIPTION
  "ANI 2 (info digit)."
 ::= { astChanEntry 32 }

astChanCidTON OBJECT-TYPE
 SYNTAX  Integer32
 MAX-ACCESS read-only
 STATUS  current
 DESCRIPTION
  "Type of Number."
 ::= { astChanEntry 33 }

astChanCidTNS OBJECT-TYPE
 SYNTAX  Integer32
 MAX-ACCESS read-only
 STATUS  current
 DESCRIPTION
  "Transit Network Select."
 ::= { astChanEntry 34 }

astChanAMAFlags OBJECT-TYPE
 SYNTAX  INTEGER {
  default(0),
  omit(1),
  billing(2),
  documentation(3)
 }
 MAX-ACCESS read-only
 STATUS  current
 DESCRIPTION
  "AMA Flags."
 ::= { astChanEntry 35 }

astChanADSI OBJECT-TYPE
 SYNTAX  INTEGER {
  unknown(0),
  available(1),
  unavailable(2),
  offHookOnly(3)
 }
 MAX-ACCESS read-only
 STATUS  current
 DESCRIPTION
  "Whether or not ADSI is detected on CPE."
 ::= { astChanEntry 36 }

astChanToneZone OBJECT-TYPE
 SYNTAX  DisplayString
 MAX-ACCESS read-only
 STATUS  current
 DESCRIPTION
  "Indication zone to use for channel."
 ::= { astChanEntry 37 }

astChanHangupCause OBJECT-TYPE
 SYNTAX  INTEGER {
  notDefined(0),
  unregistered(3),
  normal(16),
  busy(17),
  noAnswer(19),
  congestion(34),
  failure(38),
  noSuchDriver(66)
 }
 MAX-ACCESS read-only
 STATUS  current
 DESCRIPTION
  "Why is the channel hung up."
 ::= { astChanEntry 38 }

astChanVariables OBJECT-TYPE
 SYNTAX  DisplayString
 MAX-ACCESS read-only
 STATUS  current
 DESCRIPTION
  "Channel Variables defined for this channel."
 ::= { astChanEntry 39 }

astChanFlags OBJECT-TYPE
 SYNTAX  BITS {
  wantsJitter(0),
  deferDTMF(1),
  writeInterrupt(2),
  blocking(3),
  zombie(4),
  exception(5),
  musicOnHold(6),
  spying(7),
  nativeBridge(8),
  autoIncrementingLoop(9)
 }
 MAX-ACCESS read-only
 STATUS  current
 DESCRIPTION
  "Flags set on this channel."
 ::= { astChanEntry 40 }

astChanTransferCap OBJECT-TYPE
 SYNTAX  INTEGER {
  speech(0),
  digital(8),
  restrictedDigital(9),
  audio3k(16),
  digitalWithTones(17),
  video(24)
 }
 MAX-ACCESS read-only
 STATUS  current
 DESCRIPTION
  "Transfer Capabilities for this channel."
 ::= { astChanEntry 41 }

astNumChanTypes OBJECT-TYPE
 SYNTAX  Integer32
 MAX-ACCESS read-only
 STATUS  current
 DESCRIPTION
  "Number of channel types (technologies) supported."
 ::= { asteriskChannels 3 }

astChanTypeTable OBJECT-TYPE
 SYNTAX  SEQUENCE OF AstChanTypeEntry
 MAX-ACCESS not-accessible
 STATUS  current
 DESCRIPTION
  "Table with details of the supported channel types."
 ::= { asteriskChannels 4 }

astChanTypeEntry OBJECT-TYPE
 SYNTAX  AstChanTypeEntry
 MAX-ACCESS not-accessible
 STATUS  current
 DESCRIPTION
  "Information about a technology we support, including
  how many channels are currently using this technology."
 INDEX  { astChanTypeIndex }
 ::= { astChanTypeTable 1 }

AstChanTypeEntry ::= SEQUENCE {
 astChanTypeIndex Integer32,
 astChanTypeName  DisplayString,
 astChanTypeDesc  DisplayString,
 astChanTypeDeviceState Integer32,
 astChanTypeIndications Integer32,
 astChanTypeTransfer Integer32,
 astChanTypeChannels Gauge32
```

```
}

astChanTypeIndex OBJECT-TYPE
 SYNTAX  Integer32 (1 .. 2147483647)
 MAX-ACCESS read-only
 STATUS  current
 DESCRIPTION
  "Index into the table of channel types."
 ::= { astChanTypeEntry 1 }

astChanTypeName OBJECT-TYPE
 SYNTAX  DisplayString
 MAX-ACCESS read-only
 STATUS  current
 DESCRIPTION
  "Unique name of the technology we are describing."
 ::= { astChanTypeEntry 2 }

astChanTypeDesc OBJECT-TYPE
 SYNTAX  DisplayString
 MAX-ACCESS read-only
 STATUS  current
 DESCRIPTION
  "Description of the channel type (technology)."
 ::= { astChanTypeEntry 3 }

astChanTypeDeviceState OBJECT-TYPE
 SYNTAX  TruthValue
 MAX-ACCESS read-only
 STATUS  current
 DESCRIPTION
  "Whether the current technology can hold device states."
 ::= { astChanTypeEntry 4 }

astChanTypeIndications OBJECT-TYPE
 SYNTAX  TruthValue
 MAX-ACCESS read-only
 STATUS  current
 DESCRIPTION
  "Whether the current technology supports progress indication."
 ::= { astChanTypeEntry 5 }

astChanTypeTransfer OBJECT-TYPE
 SYNTAX  TruthValue
 MAX-ACCESS read-only
 STATUS  current
 DESCRIPTION
  "Whether the current technology supports transfers, where
  Asterisk can get out from inbetween two bridged channels."
 ::= { astChanTypeEntry 6 }

astChanTypeChannels OBJECT-TYPE
 SYNTAX  Gauge32
 MAX-ACCESS read-only
 STATUS  current
 DESCRIPTION
  "Number of active channels using the current technology."
 ::= { astChanTypeEntry 7 }

astChanScalars OBJECT IDENTIFIER ::= { asteriskChannels 5 }

astNumChanBridge OBJECT-TYPE
       SYNTAX          Gauge32
       MAX-ACCESS      read-only
       STATUS          current
       DESCRIPTION
               "Number of channels currently in a bridged state."
       ::= { astChanScalars 1 }

END
```

Digium MIB Definitions

```
DIGIUM-MIB DEFINITIONS ::= BEGIN

IMPORTS
 enterprises, MODULE-IDENTITY
  FROM SNMPv2-SMI;

digium MODULE-IDENTITY
 LAST-UPDATED "200806202000Z"
 ORGANIZATION "Digium, Inc."
 CONTACT-INFO
  "Mark Spencer
  Email: markster@digium.com"
 DESCRIPTION
  "The Digium private-enterprise MIB"
 REVISION "200806202000Z"
 DESCRIPTION
  "Corrected imports and missing revision for last update.
   Addresses bug 12905. - jeffg@opennms.org"
 REVISION "200602041900Z"
 DESCRIPTION
  "Initial revision."
 ::= { enterprises 22736 }

END
```

Speech Recognition API

The Asterisk Speech Recognition API

The generic speech recognition engine is implemented in the res_speech.so module. This module connects through the API to speech recognition software, that is not included in the module.

To use the API, you must load the res_speech.so module before any connectors. For your convenience, there is a preload line commented out in the modules.conf sample file.

Dialplan Applications:

The dialplan API is based around a single speech utilities application file, which exports many applications to be used for speech recognition. These include an application to prepare for speech recognition, activate a grammar, and play back a sound file while waiting for the person to speak. Using a combination of these applications you can easily make a dialplan use speech recognition without worrying about what speech recognition engine is being used.

SpeechCreate(Engine Name)

This application creates information to be used by all the other applications. It must be called before doing any speech recognition activities such as activating a grammar. It takes the engine name to use as the argument, if not specified the default engine will be used.

If an error occurs are you are not able to create an object, the variable ERROR will be set to 1. You can then exit your speech recognition specific context and play back an error message, or resort to a DTMF based IVR.

SpeechLoadGrammar(Grammar Name|Path)

Loads grammar locally on a channel. Note that the grammar is only available as long as the channel exists, and you must call SpeechUnloadGrammar before all is done or you may cause a memory leak. First argument is the grammar name that it will be loaded as and second argument is the path to the grammar.

SpeechUnloadGrammar(Grammar Name)

Unloads a locally loaded grammar and frees any memory used by it. The only argument is the name of the grammar to unload.

SpeechActivateGrammar(Grammar Name)

This activates the specified grammar to be recognized by the engine. A grammar tells the speech recognition engine what to recognize, and how to portray it back to you in the dialplan. The grammar name is the only argument to this application.

SpeechStart()

Tell the speech recognition engine that it should start trying to get results from audio being fed to it. This has no arguments.

SpeechBackground(Sound File|Timeout)

This application plays a sound file and waits for the person to speak. Once they start speaking playback of the file stops, and silence is heard. Once they stop talking the processing sound is played to indicate the speech recognition engine is working. Note it is possible to have more then one result. The first argument is the sound file and the second is the timeout. Note the timeout will only start once the sound file has stopped playing.

SpeechDeactivateGrammar(Grammar Name)

This deactivates the specified grammar so that it is no longer recognized. The only argument is the grammar name to deactivate.

SpeechProcessingSound(Sound File)

This changes the processing sound that SpeechBackground plays back when the speech recognition engine is processing and working to get results. It takes the sound file as the only argument.

SpeechDestroy()

This destroys the information used by all the other speech recognition applications. If you call this application but end up wanting to recognize more speech, you must call SpeechCreate again before calling any other application. It takes no arguments.

Getting Result Information:

The speech recognition utilities module exports several dialplan functions that you can use to examine results.

- `${SPEECH(status)}` - Returns `1` if `SpeechCreate` has been called. This uses the same check that applications do to see if a speech object is setup. If it returns `0` then you know you can not use other speech applications.
- `${SPEECH(spoke)}` - Returns `1` if the speaker spoke something, or `0` if they were silent.
- `${SPEECH(results)}` - Returns the number of results that are available.
- `${SPEECH_SCORE(result number)}` - Returns the score of a result.
- `${SPEECH_TEXT(result number)}` - Returns the recognized text of a result.
- `${SPEECH_GRAMMAR(result number)}` - Returns the matched grammar of the result.
- `SPEECH_ENGINE(name)=value` - Sets a speech engine specific attribute.

Dialplan Flow:

1. Create a speech recognition object using `SpeechCreate()`
2. Activate your grammars using `SpeechActivateGrammar(Grammar Name)`
3. Call `SpeechStart()` to indicate you are going to do speech recognition immediately
4. Play back your audio and wait for recognition using `SpeechBackground(Sound File|Timeout)`
5. Check the results and do things based on them
6. Deactivate your grammars using `SpeechDeactivateGrammar(Grammar Name)`
7. Destroy your speech recognition object using `SpeechDestroy()`

Dialplan Examples:

This is pretty cheeky in that it does not confirmation of results. As well the way the grammar is written it returns the person's extension instead of their name so we can just do a Goto based on the result text.

Grammar: company-directory.gram

```
#ABNF 1.0;
language en-US;
mode voice;
tag-format <lumenvox/1.0>;
root $company_directory;

$josh = ((Joshua | Josh) [Colp]):"6066";
$mark = (Mark [Spencer] | Markster):"4569";
$kevin = (Kevin [Fleming]):"2567";

$company_directory = ($josh | $mark | $kevin) { $ = $$ };
```

Dialplan logic

extensions.conf

```
[dial-by-name]
exten => s,1,SpeechCreate()
exten => s,2,SpeechActivateGrammar(company-directory)
exten => s,3,SpeechStart()
exten => s,4,SpeechBackground(who-would-you-like-to-dial)
exten => s,5,SpeechDeactivateGrammar(company-directory)
exten => s,6,Goto(internal-extensions-${SPEECH_TEXT(0)})
```

Useful Dialplan Tidbits

A simple macro that can be used for confirm of a result. Requires some sound files. ARG1 is equal to the file to play back after "I heard..." is played.

extensions.conf

```
[macro-speech-confirm]
exten => s,1,SpeechActivateGrammar(yes_no)
exten => s,2,Set(OLDTEXT0=${SPEECH_TEXT(0)})
exten => s,3,Playback(heard)
exten => s,4,Playback(${ARG1})
exten => s,5,SpeechStart()
exten => s,6,SpeechBackground(correct)
exten => s,7,Set(CONFIRM=${SPEECH_TEXT(0)})
exten => s,8,GotoIf($["${SPEECH_TEXT(0)}" = "1"]?9:10)
exten => s,9,Set(CONFIRM=yes)
exten => s,10,Set(CONFIRMED=${OLDTEXT0})
exten => s,11,SpeechDeactivateGrammar(yes_no)
```

The Asterisk Speech Recognition C API

The module `res_speech.so` exports a C based API that any developer can use to speech recognize enable their application. The API gives greater control, but requires the developer to do more on their end in comparison to the dialplan speech utilities.

For all API calls that return an integer value, a non-zero value indicates an error has occurred.

Creating a speech structure

```
struct ast_speech *ast_speech_new(char *engine_name, int format)

struct ast_speech *speech = ast_speech_new(NULL, AST_FORMAT_SLINEAR);
```

This will create a new speech structure that will be returned to you. The speech recognition engine name is optional and if NULL the default one will be used. As well for now format should always be AST_FORMAT_SLINEAR.

Activating a grammar

```
int ast_speech_grammar_activate(struct ast_speech *speech, char *grammar_name)

res = ast_speech_grammar_activate(speech, "yes_no");
```

This activates the specified grammar on the speech structure passed to it.

Start recognizing audio

```
void ast_speech_start(struct ast_speech *speech)

ast_speech_start(speech);
```

This essentially tells the speech recognition engine that you will be feeding audio to it from then on. It MUST be called every time before you start feeding audio to the speech structure.

Send audio to be recognized

```
int ast_speech_write(struct ast_speech *speech, void *data, int len)

res = ast_speech_write(speech, fr->data, fr->datalen);
```

This writes audio to the speech structure that will then be recognized. It must be written signed linear only at this time. In the future other formats may be supported.

Checking for results

The way the generic speech recognition API is written is that the speech structure will undergo state changes to indicate progress of recognition. The states are outlined below:

- `AST_SPEECH_STATE_NOT_READY` - The speech structure is not ready to accept audio
- `AST_SPEECH_STATE_READY` - You may write audio to the speech structure
- `AST_SPEECH_STATE_WAIT` - No more audio should be written, and results will be available soon.
- `AST_SPEECH_STATE_DONE` - Results are available and the speech structure can only be used again by calling `ast_speech_start`

It is up to you to monitor these states. Current state is available via a variable on the speech structure. (`state`)

Knowing when to stop playback

If you are playing back a sound file to the user and you want to know when to stop play back because the individual started talking use the following.

```
ast_test_flag(speech, AST_SPEECH_QUIET) /* This will return a positive value when the
person has started talking. */
```

Getting results

```
struct ast_speech_result *ast_speech_results_get(struct ast_speech *speech)

struct ast_speech_result *results = ast_speech_results_get(speech);
```

This will return a linked list of result structures. A result structure looks like the following:

```
struct ast_speech_result {
    char *text;                      /*!< Recognized text */
    int score;                       /*!< Result score */
    char *grammar;                   /*!< Matched grammar */
    struct ast_speech_result *next; /*!< List information */
};
```

Freeing a set of results

```
int ast_speech_results_free(struct ast_speech_result *result)

res = ast_speech_results_free(results);
```

This will free all results on a linked list. Results MAY NOT be used as the memory will have been freed.

Deactivating a grammar

```
int ast_speech_grammar_deactivate(struct ast_speech *speech, char *grammar_name)

res = ast_speech_grammar_deactivate(speech, "yes_no");
```

This deactivates the specified grammar on the speech structure.

Destroying a speech structure

```
int ast_speech_destroy(struct ast_speech *speech)

res = ast_speech_destroy(speech);
```

This will free all associated memory with the speech structure and destroy it with the speech recognition engine.

Loading a grammar on a speech structure

```
int ast_speech_grammar_load(struct ast_speech *speech, char *grammar_name, char *grammar)

res = ast_speech_grammar_load(speech, "builtin:yes_no", "yes_no");
```

Unloading a grammar on a speech structure

If you load a grammar on a speech structure it is preferred that you unload it as well, or you may cause a memory leak. Don't say I didn't warn you.

```
int ast_speech_grammar_unload(struct ast_speech *speech, char *grammar_name)

res = ast_speech_grammar_unload(speech, "yes_no");
```

This unloads the specified grammar from the speech structure.

Utilizing the StatsD Dialplan Application

Utilizing the StatsD Dialplan Application

This page will document how to use the StatsD dialplan application. Statistics will be logged anytime that a call is made to an extension, within the dialplan, that uses the StatsD application.

Overview

StatsD is a daemon used to aggregate statistics by using different metrics and then summarizing these statistics in a way that can be useful to the users. One of StatsD's most useful capabilities is its ability to use a graphing back-end to display the statistics graphically. StatsD makes this very simple by accepting statistics through a short, one-line command and then grouping and arranging the statistics for you.

This StatsD application is a dialplan application that is used to send statistics automatically whenever a call is made to an extension that employs the application. The user must provide the arguments to the application in the dialplan, but after that, the application will send statistics to StatsD without requiring the user to perform anymore actions whenever a call comes through that extension.

Setup

To send statistics to a StatsD server, you first need to have a StatsD server able to accept the metrics you are sending. This does not have to be the same machine that contains your asterisk instance.

Installing StatsD on a machine will make the machine a StatsD server. A backend can then be installed on the same machine as a way of viewing the statistics that StatsD receives. StatsD already has the capability built in to support a backend. Most backends are graphs that you can view through a browser and allow you to track statistics that you have sent in real time.

After a StatsD server has been set up, all that is needed is to provide the IP address of the server and the port to *statsd.conf*. You will now have a StatsD server that can accept statistics and your dialplan application will send statistics directly to this server. If a backend is configured on the StatsD server, then the backend will automatically interact with StatsD when statistics are sent to the server.

If you wish to set up your own StatsD server, then you can download StatsD from here: https://github.com/etsy/statsd.

A list of available backends to be used is available here: https://github.com/etsy/statsd/wiki/Backends.

Requirements

Only a few requirements are needed for working with the StatsD application.

- You need a statsd capable backend to receive statistics.
- Build Asterisk with the res_statsd module. This provides the StatsD dialplan application.
- Configure and enable StatsD support in Asterisk's statsd.conf

Configuration

To send statistics from the dialplan application to a StatsD server, the only options that you need from *statsd.conf* are *enabled* and *server*.

- *enabled*- whether or not StatsD support is enabled in Asterisk. To use StatsD, this must be set to yes.
- *server*- the address of the StatsD server. A port is not required, and if one is not provided, will be used 8125 as the default port.

statsd.conf

```
[general]
enabled = yes   ; When set to yes, statsd support is enabled
server = 127.0.0.1  ; server[:port] of statsd server to use.
      ; If not specified, the port is 8125
;prefix =    ; Prefix to prepend to all metrics
;add_newline = no  ; Append a newline to every event. This is
      ; useful if you want to run a fake statsd
      ; server using netcat (nc -lu 8125)
```

If you wish to add a port, such as 8126, to the server address, then you would add it at the end of the address with a colon, like so: *127.0.0.1:8126*.

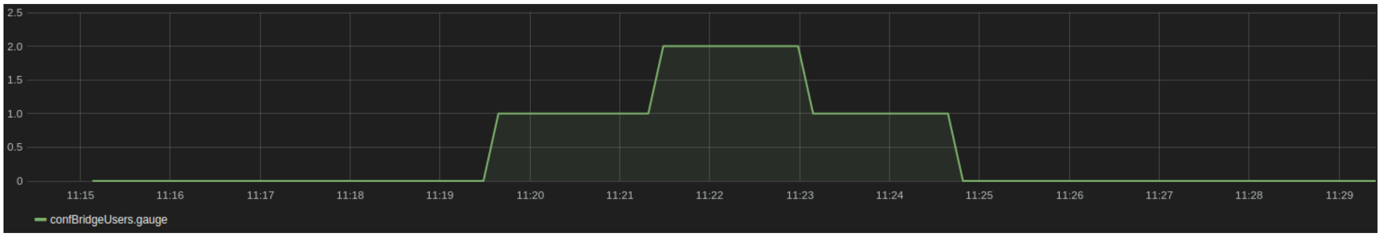
extension.conf

```
[default]

exten => 100,1,NoOp()
 same => n,StatsD(g,confBridgeUsers,+1,1)
 same => n,Set(CHANNEL(hangup_handler_push)=default,200,1);
 same => n,ConfBridge(1000)
 same => n,StatsD(g,confBridgeUsers,-1,1)
 same => n,Hangup()


exten => 200,1,NoOp()
 same => n,StatsD(g,confBridgeUsers,-1,1)
 same => n,Hangup()
```

Example

The image below is an example of how calling into the dialplan provided above would send statistics to a StatsD server.

- The number of members in the conference initially is at 0. When someone calls into the conference, the gauge *confBridgeUsers* is incremented by 1, and the graph below shows the count of *confBridgeUsers* to be 1.
- When another person calls in and joins the conference, the count is incremented to 2, and the graph displays that two people are in the conference.
- When one person hangs up or is kicked from the conference, the count will decrement, showing that only one person remains in the conference.
- When the final person hangs up, the count of *confBridgeUsers* is decremented again, and the value of *confBridgeUsers* is again 0.



Note: this graph is not a part of StatsD, but is rather a backend that StatsD can be configured to use. This configuration would occur on the StatsD server.

Codec Opus

Configuration Options

The Opus codec for Asterisk exposes a few configuration options that allow adjustments to be made to the encoder. The following options can be used to define custom format types within the *codecs.conf* file. These custom format types can then be specified in the "allow" line of an endpoint.

Option Name	Description	Default
type	Must be of type "opus".	"" (empty string)
packet_loss	Encoder's packet loss percentage. Can be any number between *0* and *100*, inclusive. A higher value results in more loss resistance.	0
complexity	Encoder's computational complexity. Can be any number between *0* and *10*, inclusive. Note, 10 equals the highest complexity.	10
signal	Encoder's signal type. Aids in mode selection on the encoder: Can be any of the following: *auto*, *voice*, *music*.	auto
application	Encoder's application type. Can be any of the following: *voip*, *audio*, *low_delay*.	voip
max_playback_rate*	Sets the "maxplaybackrate" format parameter on the SDP and also limits the bandwidth on the encoder. Any value between *8000* and *48000* (inclusive) is valid, however typically it should match one of the usual opus bandwidths. Below is a mapping of values to bandwidth: 8000 Narrow Band 8001 – 16000 Medium Band 16001 – 24000 Wide Band 24001 – 32000 Super Wide Band 32001 – 48000 Full Band	48000
max_bandwidth	Sets an upper bandwidth bound on the encoder. Can be any of the following: narrow, medium, wide, super_wide, full.	full
bitrate*	Specify the maximum average bitrate (sdp parameter "maxaveragebitrate"). Any value between 500 and 512000 is valid. The following values are also allowed: *auto*, *max*.	auto
cbr*	Sets the "cbr" (constant bit rate) format parameter on SDP. Also tells the encoder to use a constant bit rate. A value of *no (0* or *false* also work) represents a variable bit rate whereas *yes* (*1* or *true* also work) represents a constant bit rate.	0
fec*	Sets the "useinbandfec" format parameter on the SDP. If set, and applicable, the encoder will add in-band forward error correction data. A value of *no (0* or *false* also work) represents disabled whereas *yes* (*1* or *true* a lso work) represents enabled.	0
dtx*	Sets the "usedtx" format on the SDP. A value of *no (0* or *false* also work) represents disabled whereas *yes* (*1* or *true* also work) represents enabled (usedtx).	0

*If the format parameter is set to its default it will not show up in the fmtp attribute line.

Examples

Limit the maximum playback rate of any endpoint that allow "opus" and don't include any forward error correction.

```
[opus]
type=opus
max_playback_rate=8000 ; Limit the bandwidth on the encoder to narrow band
fec=no ; Do not include in-band forward error correction data
```

Limit endpoints that allow "myopus" to wide band and use a constant bit rate:

```
[myopus]
type=opus
bitrate=16000 ; Maximum encoded bit rate used
cbr=yes ; The encoder will use a constant bit rate
```

WebRTC

Configuring Asterisk for WebRTC Clients

Overview

This tutorial will walk you through configuring Asterisk to service WebRTC clients.

You will...

- Modify or create an Asterisk HTTPS TLS server.
- Create a PJSIP WebSocket transport.
- Create PJSIP Endpoint, AOR and Authentication objects that represent a WebRTC client.

Prerequisites

Asterisk Installation:

You should have a working chan_pjsip based Asterisk installation to start with and for purposes of this tutorial, it must be version 15.5 or greater. Either install Asterisk from your distribution's packages or preferably install Asterisk from source. Either way, there are a few modules over and above the standard ones that must be present for websockets and WebRTC to work:

- res_crypto
- res_http_websocket
- res_pjsip_transport_websocket
- codec_opus (optional but highly recommended for high quality audio)

We recommend installing Asterisk from source because it's easy to make sure those modules are built and installed.

Certificates:

Technically, a client can use WebRTC over an unsecured websocket to connect to Asterisk. In practice though, most browsers will require a TLS based websocket to be used. You can use self-signed certificates to set up the Asterisk TLS server but getting browsers to accept them is tricky so if you're able, we highly recommend getting trusted certificates from an organization such as LetsEncrypt.

If you already have certificate files (certificate, key, CA certificate), whether self-signed or trusted, you can skip the rest of this section. If you need to generate a self-signed certificate, read on.

Create Certificates

Asterisk provides a utility script, **ast_tls_cert** in its **contrib/scripts** source directory. We will use it to make a self-signed certificate authority and a server certificate for Asterisk, signed by our new authority.

From the Asterisk source directory run the following commands. You'll be prompted to set a a pass phrase for the CA key, then you'll be asked for that same pass phrase a few times. Use anything you can easily remember. The pass phrase is indicated below with "********". Replace "pbx.example.com" with your PBX's hostname or ip address. Replace "My Organization" as appropriate.

```
$ sudo mkdir /etc/asterisk/keys
$ sudo contrib/scripts/ast_tls_cert -C pbx.example.com -O "My Organization" -d
/etc/asterisk/keys

No config file specified, creating '/etc/asterisk/keys/tmp.cfg'
You can use this config file to create additional certs without
re-entering the information for the fields in the certificate
Creating CA key /etc/asterisk/keys/keys/ca.key
Generating RSA private key, 4096 bit long modulus
............................................................................++
.....................++
e is 65537 (0x010001)
Enter pass phrase for /etc/asterisk/keys/ca.key:********
Verifying - Enter pass phrase for /etc/asterisk/keys/ca.key:********
Creating CA certificate /etc/asterisk/keys/ca.crt
Enter pass phrase for /etc/asterisk/keys/ca.key:********
Creating certificate /etc/asterisk/keys/asterisk.key
Generating RSA private key, 1024 bit long modulus
........++++++
............++++++
e is 65537 (0x010001)
Creating signing request /etc/asterisk/keys/asterisk.csr
Creating certificate /etc/asterisk/keys/asterisk.crt
Signature ok
subject=CN = pbx.example.com, O = My Organization
Getting CA Private Key
Enter pass phrase for /etc/asterisk/keys/ca.key:********
Combining key and crt into /etc/asterisk/keys/asterisk.pem


$ ls -l /etc/asterisk/keys
total 32
-rw------- 1 root root 1204 Mar  4  2019 asterisk.crt
-rw------- 1 root root  574 Mar  4  2019 asterisk.csr
-rw------- 1 root root  887 Mar  4  2019 asterisk.key
-rw------- 1 root root 2091 Mar  4  2019 asterisk.pem
-rw------- 1 root root  149 Mar  4  2019 ca.cfg
-rw------- 1 root root 1736 Mar  4  2019 ca.crt
-rw------- 1 root root 3311 Mar  4  2019 ca.key
-rw------- 1 root root  123 Mar  4  2019 tmp.cfg
```

We'll use the asterisk.crt and asterisk.key files later to configure the HTTP server.

Asterisk Configuration

Configure Asterisk's built-in HTTP daemon

To communicate with websocket clients, Asterisk uses its built-in HTTP daemon. Configure **/etc/asterisk/http.conf** as follows:

```
[general]
enabled=yes
bindaddr=0.0.0.0
bindport=8088
tlsenable=yes
tlsbindaddr=0.0.0.0:8089
tlscertfile=/etc/asterisk/keys/asterisk.crt
tlsprivatekey=/etc/asterisk/keys/asterisk.key
```

If you have not used the generated self-signed certificates produced in the "Create Certificates" section then you will need to set the "tlscertfile" and "tlsprivatekey" to the path of your own certificates if they differ.

Now start or restart Asterisk and make sure the TLS server is running by issuing the following CLI command:

```
*CLI>  http show status

HTTP Server Status:
Prefix:
Server: Asterisk/GIT-16-a84c257cd6
Server Enabled and Bound to [::]:8088

HTTPS Server Enabled and Bound to [::]:8089

Enabled URI's:
/test_media_cache/... => HTTP Media Cache Test URI
/guimohdir_rh => HTTP POST mapping
/httpstatus => Asterisk HTTP General Status
/phoneprov/... => Asterisk HTTP Phone Provisioning Tool
/amanager => HTML Manager Event Interface w/Digest authentication
/backups => HTTP POST mapping
/arawman => Raw HTTP Manager Event Interface w/Digest authentication
/manager => HTML Manager Event Interface
/rawman => Raw HTTP Manager Event Interface
/static/... => Asterisk HTTP Static Delivery
/amxml => XML Manager Event Interface w/Digest authentication
/mxml => XML Manager Event Interface
/moh => HTTP POST mapping
/ari/... => Asterisk RESTful API
/ws => Asterisk HTTP WebSocket
<there may be more>
```

Note that the HTTPS Server is enabled and bound to `[::]:8089` and that the `/ws` URI is enabled.

Configure PJSIP

If you're not already familiar with configuring Asterisk's chan_pjsip driver, visit the res_pjsip configuration page.

PJSIP WSS Transport

Although the HTTP daemon does the heavy lifting for websockets, we still need to define a basic PJSIP Transport for websockets.

/etc/asterisk/pjsip.conf

```
[transport-wss]
type=transport
protocol=wss
bind=0.0.0.0
; All other transport parameters are ignored for wss transports.
```

PJSIP Endpoint, AOR and Auth

We now need to create the basic PJSIP objects that represent the client. In this example, we'll call the client `webrtc_client` but you can use any name you like, such as an extension number. Only the minimum options needed for a working configuration are shown. NOTE: It's normal for multiple objects in pjsip.conf to have the same name as long as the types differ.

/etc/asterisk/pjsip.conf

```
[webrtc_client]
type=aor
max_contacts=5
remove_existing=yes

[webrtc_client]
type=auth
auth_type=userpass
username=webrtc_client
password=webrtc_client ; This is a completely insecure password!  Do NOT expose this
                       ; system to the Internet without utilizing a better password.

[webrtc_client]
type=endpoint
aors=webrtc_client
auth=webrtc_client
dtls_auto_generate_cert=yes
webrtc=yes
; Setting webrtc=yes is a shortcut for setting the following options:
; use_avpf=yes
; media_encryption=dtls
; dtls_verify=fingerprint
; dtls_setup=actpass
; ice_support=yes
; media_use_received_transport=yes
; rtcp_mux=yes
context=default
disallow=all
allow=opus,ulaw
```

An explanation of each of these settings parameters can be found on the Asterisk 15 Configuration_res_pjsip page. Briefly:

- Declare an endpoint that references our previously-made aor and auth.
- Notify Asterisk to expect the AVPF profile (secure RTP)
- Setup the DTLS method of media encryption.
- Specify which certificate files to use for TLS negotiations with this endpoint and our verification and setup methods.
- Enable ICE support
- Tell Asterisk to send media across the same transport that we receive it from.
- Enable mux-ing of RTP and RTCP events onto the same socket.
- Place received calls from this endpoint into an Asterisk Dialplan context called "default"
- And setup codecs by first disabling all and then selectively enabling Opus (presuming that you installed the Opus codec for Asterisk as mentioned at the beginning of this tutorial), then G.711 u-law.

Restart Asterisk

Restart Asterisk to pick up the changes and if you have a firewall, don't forget to allow TCP port 8089 through so your client can connect.

Wrapup

At this point, your WebRTC client should be able to register and make calls. If you've used self-signed certificates however, your browser may not allow the connection and because the attempt is not from a normal URI supplied by the user, the user might not even be notified that there's an issue. You *may* be able to get the browser to accept the certificate by visiting "https://pbx.example.com:8089/ws" directly. This will usually result in a warning from the browser and may give you the opportunity to accept the self-signed certificate and/or create an exception.   If you generated your certificate from a pre-existing local Certificate Authority, you can also import that Certificate Authority's certificate into your trusted store but that procedure is beyond the scope of this document.

WebRTC tutorial using SIPML5

Tutorial Overview

This tutorial demonstrates basic WebRTC support and functionality within Asterisk. Asterisk will be configured to support a remote WebRTC client, the sipml5 client, for the purposes of making calls to/from Asterisk within a web browser. You must be running a recent (as of September 2018) version of a Mozilla or Chromium based web browser.

Setup Asterisk

Follow the instructions at Configuring Asterisk for WebRTC Clients before proceeding, The rest of this tutorial assumes that your PBX is reachable at `pbx.example.com` and that the client is known as `webrtc_client`.

Configure Asterisk Dialplan

We'll make a simple dialplan for receiving a test call from the sipml5 client.

/etc/asterisk/extensions.conf

```
[default]
exten => 200,1,Answer()
same => n,Playback(demo-congrats)
same => n,Hangup()
```

This instructs Asterisk to Answer a call to "200," to play a file named "demo-congrats" (included in Asterisk's core sound file packages), and to hang up. To make the extension active, either restart Asterisk or issue a "dialplan reload" command from the Asterisk CLI.

Browsers and WSS

When using WSS as a transport, Chrome and Firefox will not allow you, by default, to connect using WSS to a server with a self-signed certificate. Rather, you'll have to install a publicly-signed certificate into Asterisk. Or, you'll have to import the the self-signed certificate we made earlier into your browser's keychain, which is outside the scope of this Wiki.

Or, for Firefox and Chrome, you can open a separate browser tab and point it to Asterisk's HTTPs server's TLS port and WS path, e.g. `https://pbx.example.com:8089/ws`, and you can manually confirm the security exception.

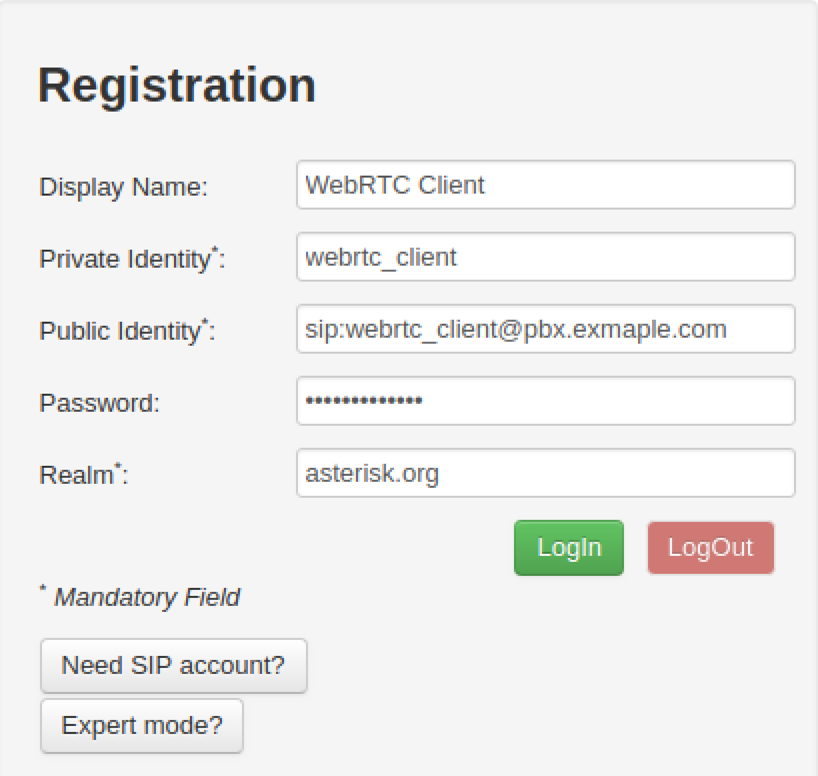
Configure SIPML5

SIPML5 is a useful client for testing Asterisk. Many real-world users explore other options that may include rolling your own client.

Next, visit https://sipml5.org - you'll be redirected to https://www.doubango.org/sipml5/

Once there, click the "Enjoy our live demo" link to be directed to the sipml5 client.

In the Registration box, use configuration similar to the following:



Registration

Display Name: WebRTC Client

Private Identity*: webrtc_client

Public Identity*: sip:webrtc_client@pbx.exmaple.com

Password: ••••••••••••••

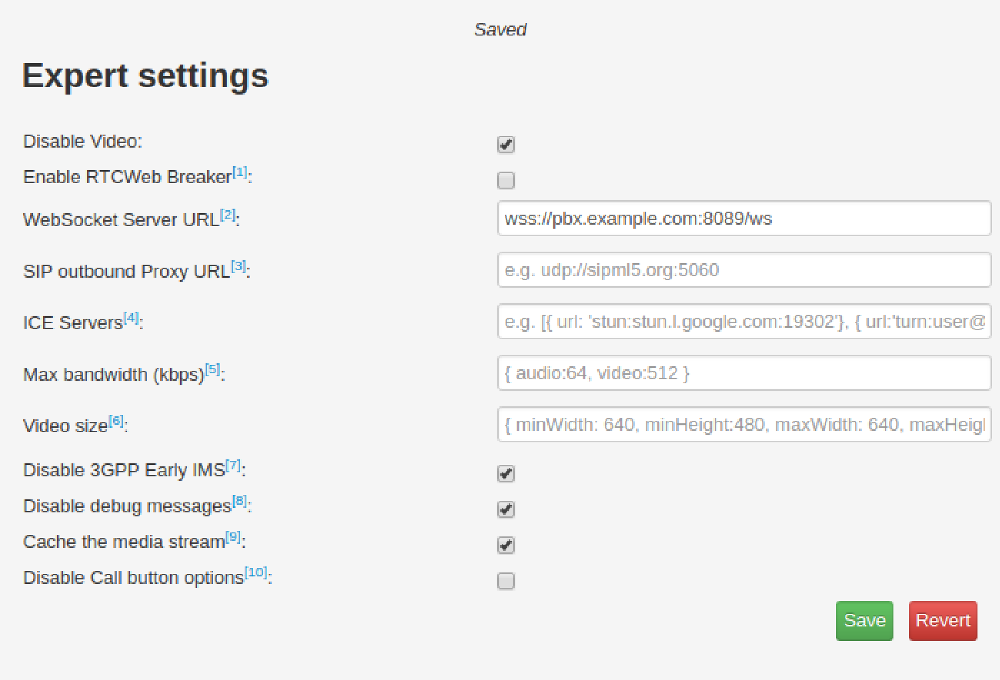
Realm*: asterisk.org

* *Mandatory Field*

Here, we have input the following:

- Display Name is a free-form string
- Private Identity is our username from our PJSIP auth object
- Public Identity is in the format:
  - sip : (name of our PJSIP aor object) @ (IP Address of the Asterisk system)
- Password is our password from our PJSIP auth object
- Realm is "asterisk.org"

Next, click the "Expert mode?" form button. It will open a new browser tab. In the Expert settings box, use a configuration similar to the following:



Saved

Expert settings

Disable Video: ☑

Enable RTCWeb Breaker[1]: ☐

WebSocket Server URL[2]: wss://pbx.example.com:8089/ws

SIP outbound Proxy URL[3]: e.g. udp://sipml5.org:5060

ICE Servers[4]: e.g. [{ url: 'stun:stun.l.google.com:19302'}, { url:'turn:user@

Max bandwidth (kbps)[5]: { audio:64, video:512 }

Video size[6]: { minWidth: 640, minHeight:480, maxWidth: 640, maxHeigl

Disable 3GPP Early IMS[7]: ☑

Disable debug messages[8]: ☑

Cache the media stream[9]: ☑

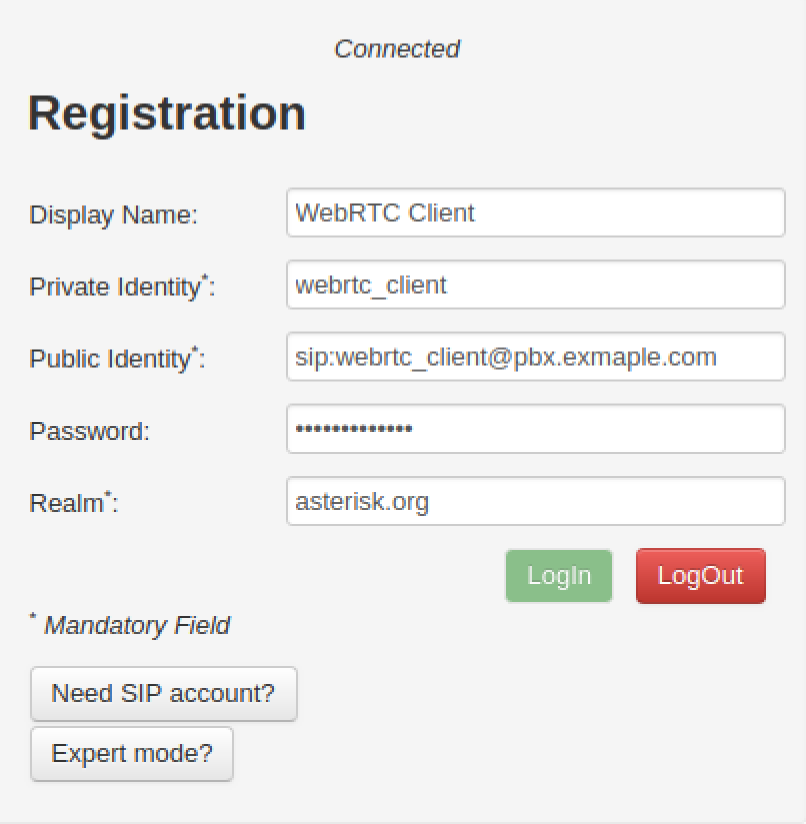
Disable Call button options[10]: ☐

Here, we have made the following changes:

- Checked the "Disable Video" box
- Filled in the WebSocket Server URL using the format:
    - wss : // (ip address of asterisk) : 8089 / ws
- Checked the "Disable 3GPP Early IMS" box

Click "Save" and return to the other demo tab with the Registration box.

Next, click "Login" and you should see *Connected* as such:



Connected

Registration

Display Name: WebRTC Client

Private Identity*: webrtc_client

Public Identity*: sip:webrtc_client@pbx.exmaple.com

Password: **************

Realm*: asterisk.org

You should see a corresponding connection happen on the Asterisk CLI. You can log into the Asterisk CLI by performing:

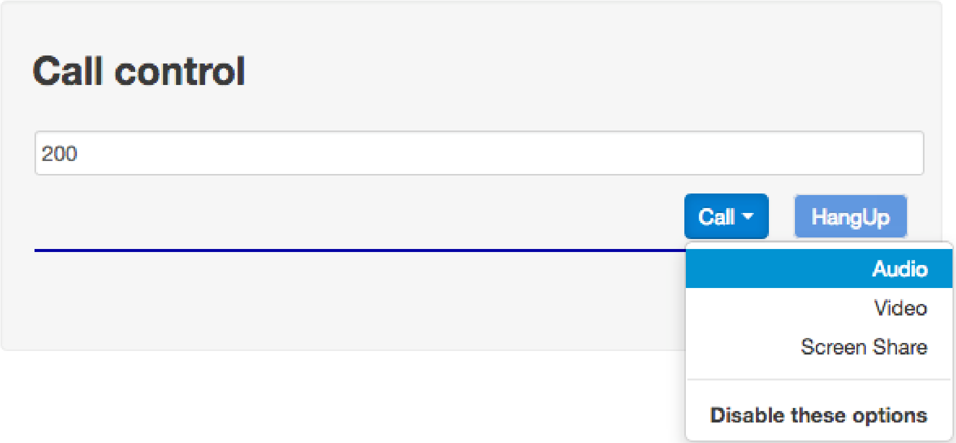
```
# asterisk -vvvr
```

Then, you can LogOut and Login and see something like:

```
== WebSocket connection from '192.168.147.245:49976' for protocol 'sip' accepted using version '13'
-- Added contact 'sips:webrtc_client@192.168.147.245:49976;transport=ws;rtcweb-breaker=no' to AOR 'webrtc_client' with
expiration of 200 seconds
== Endpoint webrtc_client is now Reachable
```

Make a test call

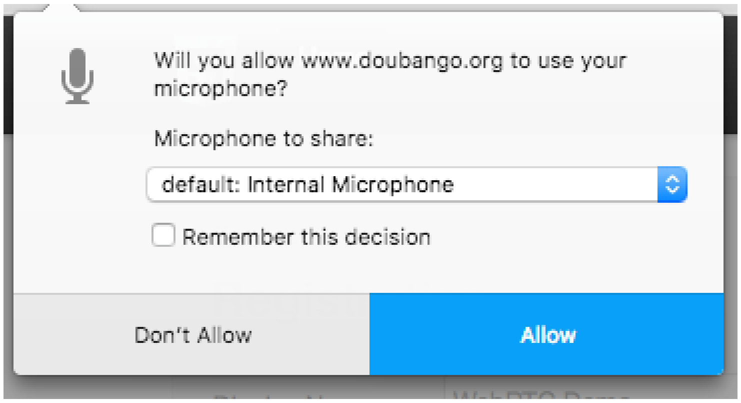
In the sipml5 Call control box input **200**. Then press the Call button. You'll see a drop-down:



Call control

200

Select "Audio" to continue. Once you do this, Firefox will display a popup asking permission to use your microphone:



Will you allow www.doubango.org to use your microphone?

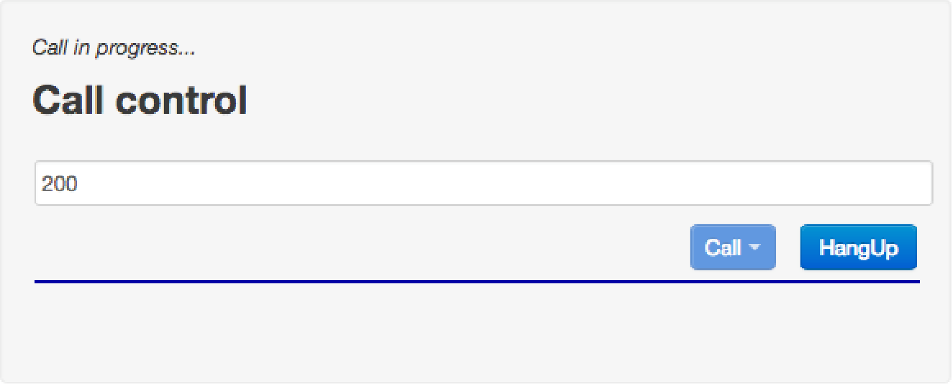
Microphone to share:

default: Internal Microphone

□ Remember this decision

Click "Allow."

Next, the Call control box will indicate that the call is proceeding:

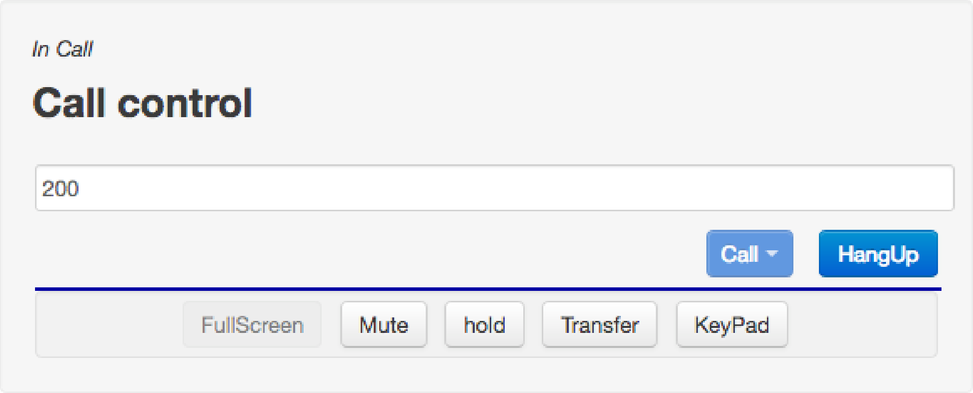


Call in progress...

Call control

200

Finally, when the call is connected, you will see *In Call*:



and you will hear "Congratulations, you have successfully installed and executed the Asterisk open source PBX..."

You've just made your first call via WebRTC using Asterisk!

Installing and Configuring CyberMegaPhone

Introduction

At AstriDevCon 2017, Digium introduced a sample WebRTC Video Conference Web Application called CyberMegaPhone (CMP2K). This document will walk you through installing the application and configuring it and Asterisk as a simple video conference server.

Prerequisites

Before proceeding, follow the instructions for Configuring Asterisk for WebRTC Clients and then use SIPML5 to test your connectivity by following the instructions at WebRTC tutorial using SIPML5. The instructions below assume you've completed those steps. Don't forget, Asterisk 15.5 or better is required.

You'll also need a working webcam and microphone on your client computer. CMP2K will not connect unless both are available.

Get The Code

The CyberMegaPhone (CMP2K) code is located in Asterisk's public Github repository at https://github.com/asterisk/cyber_mega_phone_2k. You can either download the code as a zip file or clone the repository using git. Which ever way you choose, download it now to the directory of your choice. We'll use `/usr/src/asterisk/cyber_mega_phone_2k` in the instructions below.

From an installation perspective, that's all there is to it. It's just configuration from now on.

As a reminder, we'll be using `pbx.example.com` as our hostname so substitute it with your own hostname or ip address.

Configure Asterisk

(Re)Configure the Asterisk HTTP Server

The CMP2K software needs to be served by a TLS capable web server. The easiest way to do this by far is to simply use Asterisk's built-in HTTP server. Here's what we need to add...

/etc/asterisk/http.conf

```
; Existing definition
[general]
enabled=yes
bindaddr=0.0.0.0
bindport=8088
tlsenable=yes
tlsbindaddr=0.0.0.0:8089
tlscertfile=<your_cert_file>
tlsprivatekey=<your_key_file>
tlscafile=<your_ca_cert_file>

; Add the following if not already present
; Allow the HTTP server to serve static content from /var/lib/asterisk/static-http
enablestatic = yes
; Create an alias that will allow us to easily load the client in a web browser.
redirect = /cmp2k /static/cyber_mega_phone_2k/index.html
```

Restart Asterisk or issue the CLI command "`config reload /etc/asterisk/http.conf`"

Now check that the configuration was applied. From the Asterisk CLI...

Asterisk CLI

```
*CLI> http show status
HTTP Server Status:
Prefix:
Server: Asterisk/GIT-16-a84c257cd6
Server Enabled and Bound to [::]:8088


HTTPS Server Enabled and Bound to [::]:8089

Enabled URI's:
/test_media_cache/... => HTTP Media Cache Test URI
/guimohdir_rh => HTTP POST mapping
/httpstatus => Asterisk HTTP General Status
/phoneprov/... => Asterisk HTTP Phone Provisioning Tool
/amanager => HTML Manager Event Interface w/Digest authentication
/backups => HTTP POST mapping
/arawman => Raw HTTP Manager Event Interface w/Digest authentication
/manager => HTML Manager Event Interface
/rawman => Raw HTTP Manager Event Interface
/static/... => Asterisk HTTP Static Delivery
/amxml => XML Manager Event Interface w/Digest authentication
/mxml => XML Manager Event Interface
/moh => HTTP POST mapping
/ari/... => Asterisk RESTful API
/ws => Asterisk HTTP WebSocket

Enabled Redirects:
  /cmp2k => /static/cyber_mega_phone_2k/index.html
```

Notice that there's a new Redirect entry.

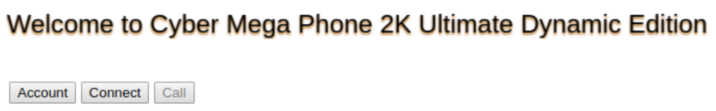
For security reasons, the HTTP server will not serve arbitrary paths so the `/static/cyber_mega_phone_2k/index.html` path will actually resolve to is `/var/lib/asterisk/static-http/cyber_mega_phone_2k/index.html`. You can either move the CMP2K directory that you downloaded to `/var/lib/asterisk/static-http` or you can simply create a symlink to it as follows:

Shell Prompt

```
# cd /var/lib/asterisk/static-http
# ln -s /usr/src/asterisk/cyber_mega_phone_2k
```

OK, let's test. From your web browser, visit `https://pbx.example.com:8089/cmp2k` remembering to substitute your hostname or ip address as appropriate.

Did you get?...



Welcome to Cyber Mega Phone 2K Ultimate Dynamic Edition

Great.

(Re)Configure PJSIP

In the Configuring Asterisk for WebRTC Clients tutorial, you created a PJSIP Endpoint named "webrtc_client". We need to modify that definition for our purposes.

/etc/asterisk/pjsip.conf

```
[webrtc_client]
type=endpoint
aors=webrtc_client
auth=webrtc_client
dtls_auto_generate_cert=yes
webrtc=yes
context=default
disallow=all
; We need to allow more codecs.
; vp8, vp9 and h264 are video pass-through codecs.
; No special Asterisk modules are required to support them.
allow=opus,g722,ulaw,vp9,vp8,h264
; Since video conferencing makes use of the Streams functionality added in Asterisk 15
; we need to indicate the maximum number of streams allowed for audio and video.
max_audio_streams = 1
max_video_streams = 15
```

You may already have some of the config from previous webrtc endpoints for certificates, keys, encryption, ice support etc and think you don't need to add the magical `webrtc=yes` but you do! The `webrtc=yes` flag does more than just shortcut already existing flags which are needed for proper SFU support.

There are two more Asterisk changes we need to make so no need to restart Asterisk just yet.

Configure app_confbridge

The sample `confbridge.conf` file is enough to get you going with one exception. In the `default_bridge` section, we need to set `video_mode=sfu`.

/etc/asterisk/confbridge.conf

```
[default_bridge]
type=bridge
; other stuff
; SFU is Selective Forwarding Mode
; Basically all participant's video streams are relayed to all other participants.
video_mode = sfu
```

One more change...

Configure extensions.conf

Now we need to configure an extension that, when dialed, will put us into the video conference bridge, so add the following to extensions.conf

/etc/asterisk/extensions.conf

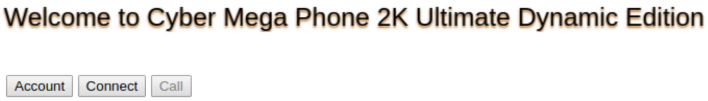
```
[default]
exten =
my_video_conference,1,Confbridge(MYCONF,default_bridge,default_user,sample_user_menu)
```

NOW, restart Asterisk!

Join the Conference Bridge!

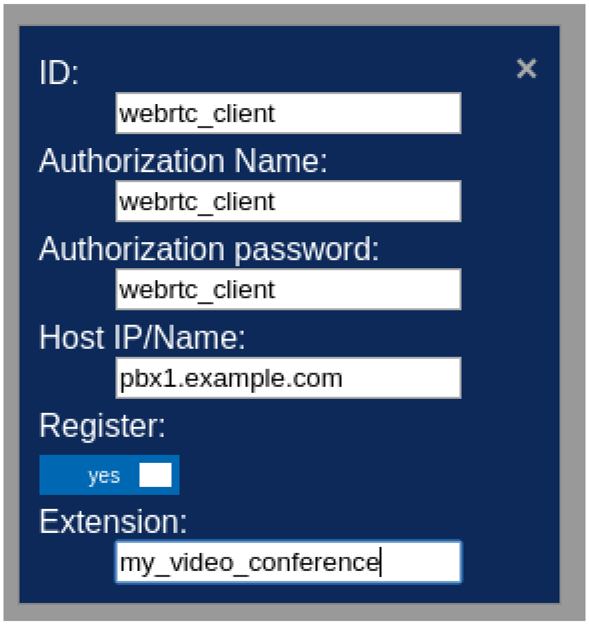
Open a browser window and visit `https://pbx.example.com:8089/cmp2k` remembering to substitute your hostname or ip address as appropriate.

You should be back at the page you got when you tested earlier...



Welcome to Cyber Mega Phone 2K Ultimate Dynamic Edition

Click the `Account` button and fill in the details as follows...



ID: webrtc_client

Authorization Name: webrtc_client

Authorization password: webrtc_client

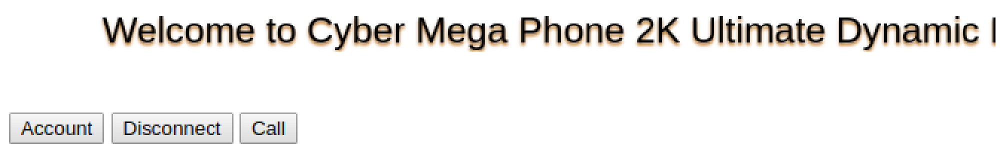
Host IP/Name: pbx1.example.com

Register: yes

Extension: my_video_conference

Now click anywhere outside the edit box or click the `X` in the upper right corner to save the information and you'll be be back to the previous page.

Now click `Connect` and you should see the button change to `Disconnect`

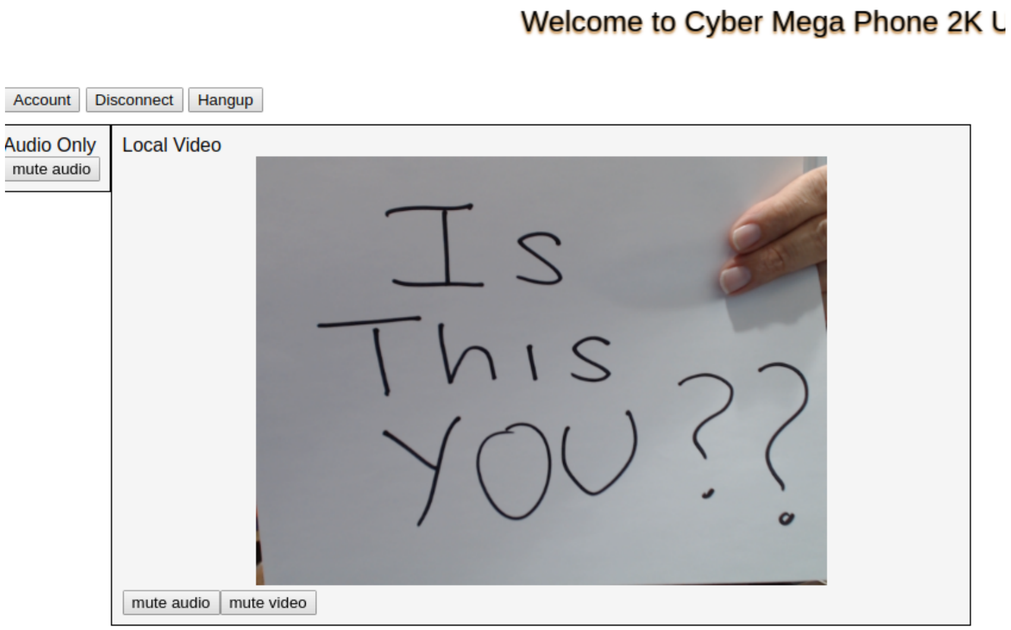


You should have also seen an `== Endpoint webrtc_client is now Reachable` message on the Asterisk console (if you were looking).

Finally, click `Call`...

You may be prompted to allow access to your microphone and camera. If so, allow them both.

Now, did you hear the `You are the only person in this conference` prompt? Do you see yourself in the video preview window?



That's it!

Bonus Points

Have a friend or co-worker join the bridge. They can use the same `webrtc_client` credentials.

Recommendations

If you experience audio issues, it may be a good idea to turn on the jitterbuffer. This can cause the audio to be slightly delayed, but will also eliminate problems such as bursty audio packets causing disruptions. You can enable this option in confbridge.conf for a user, or you can do it through the dialplan before placing the user in the conference by using the JITTERBUFFER dialplan function for a more fine tuned experience.

Deployment

Basic PBX Functionality

In this section, we're going to guide you through the basic setup of a very primitive PBX. After you finish, you'll have a basic PBX with two phones that can dial each other. In later modules, we'll go into more detail on each of these steps, but in the meantime, this will give you a basic system on which you can learn and experiement.

The Most Basic PBX

Requirements and Assumptions

While it won't be anything to brag about, this basic PBX that you will build from Asterisk will help you learn the fundamentals of configuring Asterisk.

For this exercise, we're going to assume that you have the following:

A machine, virtual or real, with Asterisk already installed.

Got here without installing Asterisk? Head back to the Installation Asterisk section. Be sure to install the SIP Channel Driver module that you want to use for SIP connectivity. This tutorial will cover using chan_sip and res_pjsip/chan_pjsip.

Two or more phones which speak the SIP voice-over-IP protocol.

There are a wide variety of SIP phones available in many different shapes and sizes, and if your budget doesn't allow for you to buy phones, feel free to use a free soft phone. Softphones are simply computer programs which run on your computer and emulate a real phone, and communicate with other devices across your network, just like a real voice-over-IP phone would. If you do use a soft phone, remember to watch out for software firewalls blocking traffic from or to the system hosting the softphone.

Creating SIP Accounts

In order for our phones to communicate with each other, we need to configure an account for each phone in the channel driver which corresponds to the protocol they'll be using. Since the phones are using the SIP protocol, we actually have two options for a SIP channel driver, the configuration file would be **sip.conf** for chan_sip, or **pjsip.conf** for chan_pjsip/res_pjsip (res_pjsip actually provides the configuration). You may already know that chan_pjsip is only available in Asterisk 12 or later. These files reside in the Asterisk configuration directory, which is typically **/etc/asterisk**. We'll include separate instructions for each channel driver below, so you have the option of using either.

> For the sake of the examples in this section, you should only configure and use one SIP channel driver at a time. Advanced configuration is needed to use two different SIP channel drivers simultaneously.

Use of templates

In the below examples of channel driver configuration we will use templates. Using templates can make the configuration seem a bit more confusing at first, but in the long run it will simplify managing larger groups of extensions. Templates are used to define sections of options or settings that can then be inherited by a child section. Read about templates here. The short story is that parents are defined with a bang in parentheses after their section name, and children specify a parent in the parentheses after their section name.

On this page

Configuring chan_sip

Open **sip.conf** with your favorite text editor, and spend a minute or two looking at the sample file. (Don't let it overwhelm you — the sample **sip.conf** has a **lot** of data in it, and can be overwhelming at first glance.) Notice that there are a couple of sections at the top of the configuration, such as [general] and [authentication], which control the overall functionality of the channel driver. Below those sections, there are sections which correspond to SIP accounts on the system.

Now copy the sample sip.conf to something like sip.conf.sample and create a new blank sip.conf to work with.

Create the following sections of configuration in the sip.conf file. Let's name your phones **Alice** and **Bob**, so that we can easily differentiate between them.

```
[general]
transport=udp

[friends_internal](!)
type=friend
host=dynamic
context=from-internal
disallow=all
allow=ulaw

[demo-alice](friends_internal)
secret=verysecretpassword ; put a strong, unique password here instead

[demo-bob](friends_internal)
secret=othersecretpassword ; put a strong, unique password here instead
```

You can read what each of the options do in the sip.conf sample file. However, let's take a look at why we are using each option here.

In the general section:

- transport=udp: sets the general transport used by all chan_sip accounts defined in configuration if they don't have their own transport defined. We want to use UDP

In the template for friends_internal:

- type=friend: we are using the friend type to make things easy. See the sip.conf.sample for more explanation on the section types available in sip.conf
- host=dynamic: our phones will register to Asterisk. Otherwise we would define the IP address of the phone here.
- context=from-internal: When Asterisk receives a call from this phone, it'll look for the dialed extension number inside this context within dialplan (/etc/asterisk/extensions.conf typically)
- disallow=all; Don't allow any codecs to be used except what is set in 'allow'
- allow=ulaw; Only allow the ulaw codec to be used.

In the sections for demo-alice and demo-bob:

- secret=verysecretpassword: This is the authentication password the phone needs to use when authenticating against Asterisk.

Note that in chan_sip configuration, the authentication username for each SIP account is the section name itself. That is "demo-alice" is the name you'll have your phone authenticate against when registering. This will be covered more in the next wiki section on registering phones to Asterisk.

Configuring chan_pjsip

Take a minute to look over the pjsip.conf sample file if you haven't already. Then backup your pjsip.conf file and create a new blank one.

Add the following configuration:

```
[transport-udp]
type=transport
protocol=udp
bind=0.0.0.0

;Templates for the necessary config sections

[endpoint_internal](!)
type=endpoint
context=from-internal
disallow=all
allow=ulaw

[auth_userpass](!)
type=auth
auth_type=userpass

[aor_dynamic](!)
type=aor
max_contacts=1

;Definitions for our phones, using the templates above

[demo-alice](endpoint_internal)
auth=demo-alice
aors=demo-alice
[demo-alice](auth_userpass)
password=unsecurepassword ; put a strong, unique password here instead
username=demo-alice
[demo-alice](aor_dynamic)

[demo-bob](endpoint_internal)
auth=demo-bob
aors=demo-bob
[demo-bob](auth_userpass)
password=unsecurepassword ; put a strong, unique password here instead
username=demo-bob
[demo-bob](aor_dynamic)
```

This configuration is roughly comparable to the chan_sip configuration in its end result. The pjsip configuration is a little more complex as the channel driver's architecture allows for more flexibility in configuration, so things tend be more modular and broken out.

Each configuration section has a type.

In the **transport** type section we configure the following:

- type=transport; This defines a transport that can be used by other configuration objects/sections, such as endpoints
- protocol=udp; We want to use the UDP protocol
- bind=0.0.0.0; We want to communicate over the most appropriate available interfaces.

In the **endpoint** template, endpoint_internal:

- type=endpoint; This defines an endpoint; a primary configuration section that you could think of as a profile of settings for a SIP connection
- context=from-internal; When Asterisk receives a call from this phone, it'll look for the dialed extension number inside this context within dialplan (/etc/asterisk/extensions.conf typically)
- disallow=all; Don't allow any codecs to be used except what is set in 'allow'
- allow=ulaw; Only allow the ulaw codec to be used.

In theauth template, auth_userpass:

- type=auth; This defines an auth section which describes credentials used for authentication. Any particular endpoint or registration will typically reference an auth section by name.
- auth_type=userpass; This tells Asterisk to use the 'username' and 'password' options set in this auth. In this case we'll be setting those in the children auths that inherit this template.

In the **aor** template, aor_dynamic:

- type=aor; This defines an aor section which describe location information making up an Address of Record. Endpoints make use of an AOR so that Asterisk knows where to send calls to the endpoint.
- max_contacts=1; We want to allow up to a maximum of one registration to this AOR. That is, we are only going to have one phone registering for each AOR.

In the AOR type configuration section (see type=aor) we purposefully don't define any contacts, and instead we set "max_contacts=1", this allows us to register phones to the AORs configured for each endpoint. This accomplishes similar behavior to "host=dynamic" for account entries in the older chan_sip driver.

After the templates, you'll see the definitions of the **demo-alice** and **demo-bob** endpoints. In those, all we do is tie them to their own auth and aor sections. In their **auth** sections we set their unique usernames and passwords. In their aor sections we don't set anything, because the necessary settings were already set in the template.

If you want to explore pjsip configuration more deeply after you finish this PBX tutorial, check out the Configuring res_pjsip wiki section.

Loading the configuration

We have more configuration to do in the next section, so there is no need to load configuration yet. However if you do want to go ahead and load the configuration; note that for some channel drivers, like chan_pjsip, certain configuration sections (transport, system) can't be reloaded at runtime. Therefore it is easiest to restart Asterisk completely here since we have made set transport settings.

```
#service asterisk restart
or
#asterisk -rx "core restart now"
```

Registering Phones to Asterisk

The next step is to configure the phones themselves to communicate with Asterisk. The way we have configured the accounts in the SIP channel driver, Asterisk will expect the phones to **register** to it. Registration is simply a mechanism where a phone communicates "Hey, I'm Bob's phone... here's my username and password. Oh, and if you get any calls for me, I'm at this particular IP address."

Configuring your particular phone is obviously beyond the scope of this guide, but here are a list of common settings you're going to want to set in your phone, so that it can communicate with Asterisk:

- **Registrar/Registration Server** - The location of the server which the phone should register to. This should be set to the IP address of your Asterisk system.
- **SIP User Name/Account Name/Address** - The SIP username on the remote system. This should be set to demo-alice on one phone and demo-bob on the other. This username corresponds directly to the section name in square brackets in sip.conf.
- **SIP Authentication User/Auth User** - On Asterisk-based systems, this will be the same as the SIP user name above.
- **Proxy Server/Outbound Proxy Server** - This is the server with which your phone communicates to make outside calls. This should be set to the IP address of your Asterisk system.

When using chan_sip you can tell whether or not your phone has registered successfully to Asterisk by checking the output of the **sip show peers** command at the Asterisk CLI. If the **Host** column says **(Unspecified)**, the phone has not yet registered. On the other hand, if the **Host** column contains an IP address and the **Dyn** column contains the letter **D**, you know that the phone has successfully registered.

```
server*CLI> sip show peers
Name/username              Host            Dyn NAT ACL Port     Status
demo-alice                 (Unspecified)    D       A  5060     Unmonitored
demo-bob                   192.168.5.105    D       A  5060     Unmonitored
2 sip peers [Monitored: 0 online, 0 offline Unmonitored: 2 online, 0 offline]
```

In the example above, you can see that Alice's phone has not registered, but Bob's phone has registered.

For chan_pjsip you can use **pjsip show endpoints**.

Debugging SIP Registrations

If you're having troubles getting a phone to register to Asterisk, make sure you watch the Asterisk CLI with the verbosity level set to at least three while you reboot the phone. You'll likely see error messages indicating what the problem is, like in this example:

```
NOTICE[22214]: chan_sip.c:20824 handle_request_register: Registration from '"Alice"&nbsp;
<sip:demo-alice@192.168.5.50>' failed for '192.168.5.103' - Wrong password
```

As you can see, Asterisk has detected that the password entered into the phone doesn't match the secret setting in the [demo-alice] section of sip.conf.

Creating Dialplan Extensions

The last things we need to do to enable Alice and Bob to call each other is to configure a couple of extensions in the dialplan.

What is an Extension?

When dealing with Asterisk, the term extension does not represent a physical device such as a phone. An extension is simply a set of actions in the dialplan which may or may not write a physical device. In addition to writing a phone, an extensions might be used for such things auto-attendant menus and conference bridges. In this guide we will be careful to use the words phone or device when referring to the physical phone, and extension when referencing the set of instructions in the Asterisk dialplan.

Let's take a quick look at the dialplan, and then add two extensions.

Open **extensions.conf**, and take a quick look at the file. Near the top of the file, you'll see some general-purpose sections named [general] and [globals]. Any sections in the dialplan beneath those two sections is known as a **context**. The sample **extensions.conf** file has a number of other contexts, with names like [demo] and [default].

We cover the concept of contexts more in Dialplan, but for now you should know that each phone or outside connection in Asterisk points at a single context. If the dialed extension does not exist in the specified context, Asterisk will reject the call. That means it is important to understand that the **context** option in your sip.conf or pjsip.conf configuration is what tells Asterisk to direct the call from the endpoint to the context we build in the next step.

Go to the bottom of your **extensions.conf** file, and add a new context named **[from-internal]** since from-internal is what we configured for the context option in the Creating SIP Accounts page.

Naming Your Dialplan Contexts

There's nothing special about the name **from-internal** for this context. It could have been named **strawberry_milkshake**, and it would have behaved exactly the same way. It is considered best practice, however, to name your contexts for the types of extensions that are contained in that context. Since this context contains extensions that will be dialing from inside the network, we'll call it from-internal.

Underneath that context name, we'll create an extesion numbered **6001** which attempts to ring Alice's phone for twenty seconds, and an extension **6002** which attempts to rings Bob's phone for twenty seconds.

```
[from-internal]
exten=>6001,1,Dial(SIP/demo-alice,20)
exten=>6002,1,Dial(SIP/demo-bob,20)
```

Each channel driver can have its own way of dialling it. The above example is for use when dialing chan_sip extensions. If you are using PJSIP then you would dial "PJSIP/demo-alice" and "PJSIP/demo-bob" respectively.

After adding that section to **extensions.conf**, go to the Asterisk command-line interface and tell Asterisk to reload the dialplan by typing the command **dialplan reload**. You can verify that Asterisk successfully read the configuration file by typing **dialplan show from-internal** at the CLI.

```
server*CLI> dialplan show from-internal
[ Context 'from-internal' created by 'pbx_config' ]
  '6001' =>         1. Dial(SIP/demo-alice,20)                    [pbx_config]
  '6002' =>         1. Dial(SIP/demo-bob,20)                      [pbx_config]

-= 2 extensions (2 priorities) in 1 context. =-
```

Now we're ready to make a test call!

Learn more about dialplan format in the Contexts, Extensions, and Priorities section.

Making a Phone Call

At this point, you should be able to pick up Alice's phone and dial extension **6002** to call Bob, and dial **6001** from Bob's phone to call Alice. As you make a few test calls, be sure to watch the Asterisk command-line interface (and ensure that your verbosity is set to a value three or higher) so that you can see the messages coming from Asterisk, which should be similar to the ones below:

```
server*CLI>     -- Executing [6002@from-internal:1] Dial("SIP/demo-alice-00000000", "SIP/demo-bob,20") in new stack
    -- Called demo-bob
    -- SIP/demo-bob-00000001 is ringing
    -- SIP/demo-bob-00000001 answered SIP/demo-alice-00000000
    -- Native bridging SIP/demo-alice-00000000 and SIP/demo-bob-00000001
  == Spawn extension (from-internal, 6002, 1) exited non-zero on 'SIP/demo-alice-00000000'
```

As you can see, Alice called extension **6002** in the [from-internal] context, which in turn used the **Dial** application to call Bob's phone. Bob's phone rang, and then answered the call. Asterisk then bridged the two calls (one call from Alice to Asterisk, and the other from Asterisk to Bob), until Alice hung up the phone.

At this point, you have a very basic PBX. It has two extensions which can dial each other, but that's all. Before we move on, however, let's review a few basic troubleshooting steps that will help you be more successful as you learn about Asterisk.

Basic PBX Troubleshooting

The most important troubleshooting step is to set your verbosity level to three (or higher), and watch the command-line interface for errors or warnings as calls are placed.

To ensure that your SIP phones are registered, type **sip show peers**(chan_sip), or **pjsip show endpoints**(chan_pjsip) at the Asterisk CLI.

To see which context your SIP phones will send calls to, type **sip show users**(chan_sip) or **pjsip show endpoint <endpoint name>**(chan_pjsip).

To ensure that you've created the extensions correctly in the **[from-internal]** context in the dialplan, type **dialplan show from-internal**.

To see which extension will be executed when you dial extension **6002**, type **dialplan show 6002@from-internal**.

Auto-attendant and IVR Menus

In this section, we'll cover the how to build voice menus, often referred to as auto-attedants and IVR menus. IVR stands for *Interactive Voice Response*, and is used to describe a system where a caller navigates through a system by using the touch-tone keys on their phone keypad.

When the caller presses a key on their phone keypad, the phone emits two tones, known as DTMF tones. DTMF stands for *Dual Tone Multi-Frequency*. Asterisk recognizes the DTMF tones and responds accordingly.

Let's dive in and learn how to build IVR menus in the Asterisk dialplan!

Background and WaitExten Applications

The **Background()** application plays a sound prompt, but listens for DTMF input. Asterisk then tries to find an extension in the current dialplan context that matches the DTMF input. If it finds a matching extension, Asterisk will send the call to that extension.

The Background() application takes the name of the sound prompt as the first parameter just like the Playback() application, so remember not to include the file extension.

✓ **Multiple Prompts**
If you have multiple prompts you'd like to play during the Background() application, simply concatenate them together with the ampersand (&) character, like this:

```
exten => 6123,1,Background(prompt1&prompt2&prompt3)
```

One problems you may encounter with the **Background()** application is that you may want Asterisk to wait a few more seconds after playing the sound prompt. In order to do this, you can call the **WaitExten()** application. You'll usually see the **WaitExten()** application called immediately after the **Background()** application. The first parameter to the **WaitExten()** application is the number of seconds to wait for the caller to enter an extension. If you don't supply the first parameter, Asterisk will use the built-in response timeout (which can be modified with the **TIMEOUT()** dialplan function).

```
[auto_attendant]
exten => start,1,Verbose(2,Incoming call from ${CALLERID(all)})
   same => n,Playback(silence/1)
   same => n,Background(prompt1&prompt2&prompt3)
   same => n,WaitExten(10)
   same => n,Goto(timeout-handler,1)

exten => timeout-handler,1)
   same => n,Dial(${GLOBAL(OPERATOR)},30)
   same => n,Voicemail(operator@default,${IF($[${DIALSTATUS} = BUSY]?b:u)})
   same => n,Hangup()
```

Goto Application and Priority Labels

Before we create a simple auto-attendant menu, let's cover a couple of other useful dialplan applications. The **Goto()** application allows us to jump from one position in the dialplan to another. The parameters to the **Goto()** application are slightly more complicated than with the other applications we've looked at so far, but don't let that scare you off.

The **Goto()** application can be called with either one, two, or three parameters. If you call the **Goto()** application with a single parameter, Asterisk will jump to the specified priority (or its label) within the current extension. If you specify two parameters, Asterisk will read the first as an extension within the current context to jump to, and the second parameter as the priority (or label) within that extension. If you pass three parameters to the application, Asterisk will assume they are the context, extension, and priority (respectively) to jump to.

```
[StartingContext]
exten => 100,1,Goto(monkeys)
   same => n,NoOp(We skip this)
   same => n(monkeys),Playback(tt-monkeys)
   same => n,Hangup()

exten => 200,1,Goto(start,1)  ; play tt-weasels then tt-monkeys

exten => 300,1,Goto(start,monkeys) ; only play tt-monkeys

exten => 400,1,Goto(JumpingContext,start,1)  ; play hello-world

exten => start,1,NoOp()
   same => n,Playback(tt-weasels)
   same => n(monkeys),Playback(tt-monkeys)

[JumpingContext]
exten => start,1,NoOp()
   same => n,Playback(hello-world)
   same => n,Hangup()
```

SayDigits, SayNumber, SayAlpha, and SayPhonetic Applications

While not exactly related to auto-attendant menus, we'll introduce some applications to read back various pieces of information back to the caller. The **SayDigits()** and **SayNumber()** applications read the specified number back to caller. To use the **SayDigits()** and **SayNumber()** application simply pass it the number you'd like it to say as the first parameter.

The **SayDigits()** application reads the specified number one digit at a time. For example, if you called **SayDigits(123)**, Asterisk would read back "one two three". On the other hand, the **SayNumber()** application reads back the number as if it were a whole number. For example, if you called **SayNumber(123)** Asterisk would read back "one hundred twenty three".

The **SayAlpha()** and **SayPhonetic()** applications are used to spell an alphanumeric string back to the caller. The **SayAlpha()** reads the specified string one letter at a time. For example, **SayAlpha(hello)** would read spell the word "hello" one letter at a time. The **SayPhonetic()** spells back a string one letter at a time, using the international phonetic alphabet. For example, **SayPhonetic(hello)** would read back "Hotel Echo Lima Lima Oscar".

We'll use these four applications to read back various data to the caller througout this guide. In the meantime, please feel free to add some sample extensions to your dialplan to try out these applications. Here are some examples:

```
exten => 6592,1,SayDigits(123)
exten => 6593,1,SayNumber(123)
exten => 6594,1,SayAlpha(hello)
exten => 6595,1,SayPhonetic(hello)
```

Creating a Simple IVR Menu

Let's go ahead and apply what we've learned about the various dialplan applications by building a very simple auto-attendant menu. It is common practice to create an auto-attendant or IVR menu in a new context, so that it remains independant of the other extensions in the dialplan. Please add the following to your dialplan (the **extensions.conf** file) to create a new **demo-menu** context. In this new context, we'll create a simple menu that prompts you to enter one or two, and then it will read back what you're entered.

> **Sample Sound Prompts**
> Please note that the example below (and many of the other examples in this guide) use sound prompts that are part of the *extra* sounds packages. If you didn't install the extra sounds earlier, now might be a good time to do that.

```
[demo-menu]
exten => s,1,Answer(500)
   same => n(loop),Background(press-1&or&press-2)
   same => n,WaitExten()

exten => 1,1,Playback(you-entered)
   same => n,SayNumber(1)
   same => n,Goto(s,loop)

exten => 2,1,Playback(you-entered)
   same => n,SayNumber(2)
   same => n,Goto(s,loop)
```

Before we can use the demo menu above, we need to add an extension to the **[docs:users]** context to redirect the caller to our menu. Add this line to the **[docs:users]** context in your dialplan:

```
exten => 6598,1,Goto(demo-menu,s,1)
```

Reload your dialplan, and then try dialing extension **6598** to test your auto-attendant menu.

Handling Special Extensions

We have the basics of an auto-attendant created, but now let's make it a bit more robust. We need to be able to handle special situations, such as when the caller enters an invalid extension, or doesn't enter an extension at all. Asterisk has a set of special extensions for dealing with situations like there. They all are named with a single letter, so we recommend you don't create any other extensions named with a single letter. You can read about all the Special Dialplan Extensions on the wiki.

Let's add a few more lines to our **[docs:demo-menu]** context, to handle invalid entries and timeouts. Modify your **[docs:demo-menu]** context so that it matches the one below:

```
[demo-menu]
exten => s,1,Answer(500)
   same => n(loop),Background(press-1&or&press-2)
   same => n,WaitExten()

exten => 1,1,Playback(you-entered)
   same => n,SayNumber(1)
   same => n,Goto(s,loop)

exten => 2,1,Playback(you-entered)
   same => n,SayNumber(2)
   same => n,Goto(s,loop)

exten => i,1,Playback(option-is-invalid)
   same => n,Goto(s,loop)

exten => t,1,Playback(are-you-still-there)
   same => n,Goto(s,loop)
```

Now dial your auto-attendant menu again (by dialing extension **6598**), and try entering an invalid option (such as **3**) at the auto-attendant menu. If you watch the Asterisk command-line interface while you dial and your verbosity level is three or higher, you should see something similar to the following:

```
  -- Executing [6598@users:1] Goto("SIP/demo-alice-00000008", "demo-menu,s,1") in new stack
  -- Goto (demo-menu,s,1)
  -- Executing [s@demo-menu:1] Answer("SIP/demo-alice-00000008", "500") in new stack
  -- Executing [s@demo-menu:2] BackGround("SIP/demo-alice-00000008", "press-1&or&press-2") in new stack
  -- <SIP/demo-alice-00000008> Playing 'press-1.gsm' (language 'en')
  -- <SIP/demo-alice-00000008> Playing 'or.gsm' (language 'en')
  -- <SIP/demo-alice-00000008> Playing 'press-2.gsm' (language 'en')
  -- Invalid extension '3' in context 'demo-menu' on SIP/demo-alice-00000008
  -- Executing [i@demo-menu:1] Playback("SIP/demo-alice-00000008", "option-is-invalid") in new stack
  -- <SIP/demo-alice-00000008> Playing 'option-is-invalid.gsm' (language 'en')
  -- Executing [i@demo-menu:2] Goto("SIP/demo-alice-00000008", "s,loop") in new stack
  -- Goto (demo-menu,s,2)
  -- Executing [s@demo-menu:2] BackGround("SIP/demo-alice-00000008", "press-1&or&press-2") in new stack
  -- <SIP/demo-alice-00000008> Playing 'press-1.gsm' (language 'en')
  -- <SIP/demo-alice-00000008> Playing 'or.gsm' (language 'en')
  -- <SIP/demo-alice-00000008> Playing 'press-2.gsm' (language 'en')
```

If you don't enter anything at the auto-attendant menu and instead wait approximately ten seconds, you should hear (and see) Asterisk go to the **t** extension as well.

Record Application

For creating your own auto-attendant or IVR menus, you're probably going to want to record your own custom prompts. An easy way to do this is with the **Record()** application. The **Record()** application plays a beep, and then begins recording audio until you press the hash key (**#**) on your keypad. It then saves the audio to the filename specified as the first parameter to the application and continues on to the next priority in the extension. If you hang up the call before pressing the hash key, the audio will not be recorded. For example, the following extension records a sound prompt called **custom-menu** in the **gsm** format in the **en/** sub-directory, and then plays it back to you.

```
exten => 6597,1,Answer(500)
   same => n,Record(en/custom-menu.gsm)
   same => n,Wait(1)
   same => n,Playback(custom-menu)
   same => n,Hangup()
```

> **Recording Formats**
> When specifiying a file extension when using the **Record()** application, you must choose a file extension which represents one of the supported file formats in Asterisk. For the complete list of file formats supported in your Asterisk installation, type **core show file formats** at the Asterisk command-line interface.

You've now learned the basics of how to create a simple auto-attendant menu. Now let's build a more practical menu for callers to be able to reach Alice or Bob or the dial-by-name directory.

Procedure 216.1. Building a Practical Auto-Attendant Menu

1. Add an extension 6599 to the [docs:users] context which sends the calls to a new context we'll build called [docs:day-menu]. Your extension should look something like:

    -
      ```
      exten=>6599,1,Goto(day-menu,s,1)
      ```
2. Add a new context called **[docs:day-menu]**, with the following contents:

    -
      ```
      [day-menu]
      exten => s,1,Answer(500)
      same => n(loop),Background(custom-menu)
      same => n,WaitExten()

      exten => 1,1,Goto(users,6001,1)

      exten => 2,1,Goto(users,6002,1)

      exten => 9,1,Directory(vm-demo,users,fe)
      exten => *,1,VoiceMailMain(@vm-demo)

      exten => i,1,Playback(option-is-invalid)
      same => n,Goto(s,loop)

      exten => t,1,Playback(are-you-still-there)
      same => n,Goto(s,loop)
      ```

1. Dial extension **6597** to record your auto-attendant sound prompt. Your sound prompt should say something like "Thank you for calling! Press one for Alice, press two for Bob, or press 9 for a company directory". Press the hash key (**#**) on your keypad when you're finished recording, and Asterisk will play it back to you. If you don't like it, simply dial extension **6597** again to re-record it.
2. Dial extension **6599** to test your auto-attendant menu.

In just a few lines of code, you've created your own auto-attendant menu. Feel free to experiment with your auto-attendant menu before moving on to the next section.

Adding Voice Mail to Dialplan Extensions

Adding voicemail to the extensions is quite simple. The Asterisk voicemail module provides two key applications for dealing with voice mail. The first, named **VoiceMail()**, allows a caller to leave a voice mail message in the specified mailbox. The second, called **VoiceMailMain()**, allows the mailbox owner to retrieve their messages and change their greetings.

Configuring Voice Mail Boxes

Now that we've covered the two main voice mail applications, let's look at the voicemail configuration. Voice mail options and mailboxes are configured in the **voicemail.conf** configuration file. This file has three major sections:

The [general] section

Near the top of **voicemail.conf**, you'll find the **[general]** section. This section of the configuration file controls the general aspects of the voicemail system, such as the maximum number of messages per mailbox, the maximum length of a voicemail message, and so forth. Feel free to look at the sample **voicemail.conf** file for more details about the various settings.

The [zonemessages] section

The **[zonemessages]** section is used to define various timezones around the world. Each mailbox can be assigned to a particular time zone, so that times and dates are announced relative to their local time. The time zones specified in this section also control the way in which times and dates are announced, such as reading the time of day in 24-hour format.

Voice Mail Contexts

After the **[general]** and **[zonemessages]** sections, any other bracketed section is a voice mail context. Within each context, you can define one or more mailbox. To define a mailbox, we set a mailbox number, a PIN, the mailbox owner's name, the primary email address, a secondary email address, and a list of mailbox options (separated by the pipe character), as shown below:

```
mailbox=>pin,full name,email address,short email address,mailbox options
```

By way of explanation, the short email address is an email address that will receive shorter email notifications suitable for mobile devices such as cell phones and pagers. It will never receive attachments.

To add voice mail capabilities to extensions **6001** and **6002**, add these three lines to the bottom of **voicemail.conf**.

```
[vm-demo]
6001 => 8762,Alice
Jones,alice@example.com,alice2@example.com,attach=no|tz=central|maxmsg=10
6002 => 9271,Bob Smith,bob@example.com,bob2@example.com,attach=yes|tz=eastern
```

Now that we've defined the mailboxes, we can go into the Asterisk CLI and type **voicemail reload** to get Asterisk to reload the **voicemail.conf** file. We can also verify that the new mailboxes have been created by typing **voicemail show users**.

```
server*CLI> voicemail reload
Reloading voicemail configuration...
server*CLI> voicemail show users
Context    Mbox  User                      Zone       NewMsg
default    general New User                                0
default    1234  Example Mailbox                           0
other      1234  Company2 User                             0
vm-demo    6001  Alice Jones               central         0
vm-demo    6002  Bob Smith                 eastern         0
5 voicemail users configured.
```

Now that we have mailboxes defined, let's add a priority to extensions **6001** and **6002** which will allow callers to leave voice mail in their respective mailboxes. We'll also add an extension **6500** to allow Alice and Bob to check their voicemail messages. Please modify your **[users]** context in **extensions.conf** to look like the following:

```
[users]
exten => 6000,1,Answer(500)
exten => 6000,n,Playback(hello-world)
exten => 6000,n,Hangup()

exten => 6001,1,Dial(SIP/demo-alice,20)
exten => 6001,n,VoiceMail(6001@vm-demo,u)

exten => 6002,1,Dial(SIP/demo-bob,20)
exten => 6002,n,VoiceMail(6002@vm-demo,u)

exten => 6500,1,Answer(500)
exten => 6500,n,VoiceMailMain(@vm-demo)
```

Reload the dialplan by typing **dialplan reload** at the Asterisk CLI. You can then test the voice mail system by dialing from one phone to the other and waiting twenty seconds. You should then be connected to the voicemail system, where you can leave a message. You should also be able to dial extension **6500** to retrieve the voicemail message. When prompted, enter the mailbox number and PIN number of the mailbox.

While in the **VoiceMainMain()** application, you can also record the mailbox owner's name, unavailable greeting, and busy greeting by pressing 0 at the voicemail menu. Please record at least the name greeting for both Alice and Bob before continuing on to the next section.

Go into lots of detail about the voicemail interface? How to move between messages, move between folders, forward messages, etc?

Deployment In Your Network

Under Construction

Top-level page for pages on dealing with NAT, Firewalling and more in relation to Asterisk.

Emergency Calling

Under Construction

Top-level page for resources, tutorials or ideas regarding emergency dialing and calling, E-911, etc.

Important Security Considerations

The pages in this section provide specific warnings about security that are pertinent to Asterisk. Just because you're already familiar with securing your Linux machine, doesn't mean you can skip this section.

> ⓘ 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.

Network Security

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 signaling. The SIP channel supports TLS encryption of the signaling, as well as SRTP (encrypted media).

Dialplan Security

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.

Log Security

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.

Asterisk Security Webinars

Asterisk VoIP Security - Part 1 of 3

VoIP Fraud: Current Threats From A Law Enforcement Perspective
Special Agent Michael McAndrews, FBI

Asterisk VoIP Security - Part 2 of 3

VoIP Security Best Practices
Dan York, Chairman, Best Practices Group, VoIP Security Alliance

Asterisk VoIP Security - Part 3 of 3

Securing Asterisk Systems
Jared Smith, Training Manager, Digium

Privacy Configuration

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

FTC Don't Call List

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.

Fighting 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".

Fighting Empty Caller ID

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.)

Using Welcome Menus for Privacy

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,s,1:s,7)
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.

Making life difficult for telemarketers

I have developed an elaborate script to torture Telemarketers, and entertain friends.

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 ...!

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:

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

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.

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.

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.
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".

Call Screening 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.

Screening Calls with 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.

Internationalization and Localization

Under Construction

Top-level page for pages discussing Asterisk's capabilities for adapting to various language environments for text, sound prompts and more.

Asterisk Sounds Packages

Asterisk utilizes a variety of sound prompts that are available in several file formats and languages. Multiple languages and formats can be installed on the same system, and Asterisk will utilize prompts from languages installed, and will automatically pick the least CPU intensive format that is available on the system (based on codecs in use, in additional to the codec and format modules installed and available).

In addition to the prompts available with Asterisk, you can create your own sets of prompts and utilize them as well. This document will tell you how the prompts available for Asterisk are created so that the prompts you create can be as close and consistent in the quality and volume levels as those shipped with Asterisk.

Getting the Sounds Tools

The sounds tools are available in the publicly accessible repotools repository. You can check these tools out with Git via the following command:

```
# git clone https://gerrit.asterisk.org/repotools
```

The sound tools are available in the subdirectory sound_tools/ which contains the following directories:

- audiofilter
- makeg722
- scripts

About the Sounds Tools

The following sections will describe the sound tools in more detail and explain what they are used for in the sounds package creation process.

audiofilter

The audiofilter application is used to "tune" the sound files in such a way that they sound good when being used while in a compressed format. The values in the scripts for creating the sound files supplied in repotools is essentially a high-pass filter that drops out audio below 100Hz (or so).

(There is an ITU specification that states for 8KHz audio that is being compressed frequencies below a certain threshold should be removed because they make the resulting compressed audio sound worse than it should.)

The audiofilter application is used by the 'converter' script located in the scripts subdirectory of repotools/sound_tools. The values being passed to the audiofilter application is as follows:

```
audiofilter -n 0.86916 -1.73829 0.86916 -d 1.00000 -1.74152 0.77536
```

The two options -n and -d are 'numerator' and 'denominator'. Per the author, Jean-Marc Valin, "These values are filter coefficients (-n means numerator, -d is denominator) expressed in the z-transform domain. There represent an elliptic filter that I designed with Octave such that 'the result sounds good'."

makeg722

The makeg722 application is used by the 'converters' script to generate the G.722 sound files that are shipped with Asterisk. It starts with the RAW sound files and then converts them to G.722.

scripts

The scripts folder is where all the magic happens. These are the scripts that the Asterisk open source team use to build the packaged audio files for the various formats that are distributed with Asterisk.

- chkcore - used to check that the contents of core-sounds-lang.txt are in sync
- chkextra - same as above, but checks the extra sound files
- mkcore - script used to generate the core sounds packages
- mkextra - script used to generate the extra sounds packages
- mkmoh - script used to generate the music on hold packages
- converters - script used to convert the master files to various formats

Sound Prompt Searching based on Channel Language

How Asterisk Searches for Sound Prompts Based on Channel Language

Each channel in Asterisk can be assigned a language by the channel driver. The channel's language code is split, piece by piece (separated by underscores), and used to build paths to look for sound prompts. Asterisk then uses the first file that is found.

This means that if we set the language to en_GB_female_BT, for example, Asterisk would search for files in:

.../sounds/en_GB_female_BT

.../sounds/en_GB_female

.../sounds/en_GB

.../sounds/en

.../sounds

This scheme makes it easy to add new sound prompts for various language variants, while falling back to a more general prompt if there is no prompt recorded in the more specific variant.

Troubleshooting

Under Construction

This page will be reworked to become a top level page for troubleshooting articles, including a page for overall troubleshooting approach and specific sub-pages on troubleshooting certain sub-systems.

If you're able to get the command-line examples above working, feel free to skip this section. Otherwise, let's look at troubleshooting connections to the Asterisk CLI.

The most common problem that people encounter when learning the Asterisk command-line interface is that sometimes they're not able to connect to the Asterisk service running in the background. For example, let's say that Fred starts the Asterisk service, but then isn't able to connect to it with the CLI:

```
[root@server ~]# service asterisk start
Starting asterisk:                                          [  OK  ]
[root@server ~]# asterisk -r
Asterisk version, Copyright (C) 1999 - 2010 Digium, Inc. and others.
Created by Mark Spencer <markster@digium.com>
=========================================================================
Unable to connect to remote asterisk (does /var/run/asterisk/asterisk.ctl exist?)
```

What does this mean? It most likely means that Asterisk did not remain running between the time that the service was started and the time Fred tried to connect to the CLI (even if it was only a matter of a few seconds.) This could be caused by a variety of things, but the most common is a broken configuration file.

To diagnose Asterisk start-up problems, we'll start Asterisk in a special mode, known as **console** mode. In this mode, Asterisk does not run as a background service or daemon, but instead runs directly in the console. To start Asterisk in console mode, pass the **-c** parameter to the **asterisk** application. In this case, we also want to turn up the verbosity, so we can see any error messages that might indicate why Asterisk is unable to start.

```
[root@server ~]# asterisk -vvvc
Asterisk version, Copyright (C) 1999 - 2010 Digium, Inc. and others.
Created by Mark Spencer <markster@digium.com>
Asterisk comes with ABSOLUTELY NO WARRANTY; type 'core show warranty' for details.
This is free software, with components licensed under the GNU General Public
License version 2 and other licenses; you are welcome to redistribute it under
certain conditions. Type 'core show license' for details.
=========================================================================
  == Parsing '/etc/asterisk/asterisk.conf':   == Found
  == Parsing '/etc/asterisk/extconfig.conf':   == Found
  == Parsing '/etc/asterisk/logger.conf':   == Found
  == Parsing '/etc/asterisk/asterisk.conf':   == Found
  Asterisk Dynamic Loader Starting:
  == Parsing '/etc/asterisk/modules.conf':   == Found
...
```

Carefully look for any errors or warnings that are printed to the CLI, and you should have enough information to solve whatever problem is keeping Asterisk from starting up.

Running Asterisk in Console Mode
We don't recommend you use Asterisk in console mode on a production system, but simply use it for debugging, especially when debugging start-up problems. On production systems, run Asterisk as a background service.

SIP Retransmissions

What is the problem with SIP retransmits?

Sometimes you get messages in the console like these:

```
retrans_pkt: Hanging up call XX77yy  - no reply to our critical packet.
retrans_pkt: Cancelling retransmit of OPTIONs
```

The SIP protocol is based on requests and replies. Both sides send requests and wait for replies. Some of these requests are important. In a TCP/IP network many things can happen with IP packets. Firewalls, NAT devices, Session Border Controllers and SIP Proxys are in the signalling path and they will affect the call.

SIP Call setup - INVITE-200 OK - ACK

To set up a SIP call, there's an INVITE transaction. The SIP software that initiates the call sends an INVITE, then wait to get a reply. When a reply arrives, the caller sends an ACK. This is a three-way handshake that is in place since a phone can ring for a very long time and the protocol needs to make sure that all devices are still on line when call setup is done and media starts to flow.

- The first reply we're waiting for is often a "100 trying". This message means that some type of SIP server has received our request and makes sure that we will get a reply. It could be the other endpoint, but it could also be a SIP proxy or SBC that handles the request on our behalf.
- After that, you often see a response in the 18x class, like "180 ringing" or "183 Session Progress". This typically means that our request has reached at least one endpoint and something is alerting the other end that there's a call coming in.
- Finally, the other side answers and we get a positive reply, "200 OK". This is a positive answer. In that message, we get an address that goes directly to the device that answers. Remember, there could be multiple phones ringing. The address is specified by the Contact: header.
- To confirm that we can reach the phone that answered our call, we now send an ACK to the Contact: address. If this ACK doesn't reach the phone, the call fails. If we can't send an ACK, we can't send anything else, not even a proper hangup. Call signaling will simply fail for the rest of the call and there's no point in keeping it alive.
- If we get an error response to our INVITE, like "Busy" or "Rejected", we send the ACK to the same address as we sent the INVITE, to confirm that we got the response.

In order to make sure that the whole call setup sequence works and that we have a call, a SIP client retransmits messages if there's too much delay between request and expected response. We retransmit a number of times while waiting for the first response. We retransmit the answer to an incoming INVITE while waiting for an ACK. If we get multiple answers,
we send an ACK to each of them.

If we don't get the ACK or don't get an answer to our INVITE, even after retransmissions, we will hangup the call with the first error message you see above.

Other SIP requests

Other SIP requests are only based on request - reply. There's no ACK, no three-way handshake. In Asterisk we mark some of these as CRITICAL - they need to go through for the call to work as expected. Some are non-critical, we don't really care what happens with them, the call will go on happily regardless.

The qualification process - OPTIONS

If you turn on qualify= in sip.conf for a device, Asterisk will send an OPTIONS request every minute to the device and check if it replies. Each OPTIONS request is retransmitted a number of times (to handle packet loss) and if we get no reply, the device is considered unreachable. From that moment, we will send a new OPTIONS request (with retransmits) every tenth second.

Why does this happen?

For some reason signalling doesn't work as expected between your Asterisk server and the other device. There could be many reasons why this happens.

- A NAT device in the signalling path. A misconfigured NAT device is in the signalling path and stops SIP messages.
- A firewall that blocks messages or reroutes them wrongly in an attempt to assist in a too clever way.
- A SIP middlebox (SBC) that rewrites contact: headers so that we can't reach the other side with our reply or the ACK.
- A badly configured SIP proxy that forgets to add record-route headers to make sure that signalling works.
- Packet loss. IP and UDP are unreliable transports. If you loose too many packets the retransmits doesn't help and communication is impossible. If this happens with signaling, media would be unusable anyway.

What can I do?

Turn on SIP debug, try to understand the signalling that happens and see if you're missing the reply to the INVITE or if the ACK gets lost. When you know what happens, you've taken the first step to track down the problem. See the list above and investigate your network.

For NAT and Firewall problems, there are many documents to help you. Start with reading sip.conf.sample that is part of your Asterisk distribution.

The SIP signalling standard, including retransmissions and timers for these, is well documented in the IETF RFC 3261.

Good luck sorting out your SIP issues!

/Olle E. Johansson

– oej (at) edvina.net, Sweden, 2008-07-22
– http://www.voip-forum.com

Troubleshooting Asterisk Module Loading

Symptoms

- Specific Asterisk functionality is no longer available or completely non-functioning, but other Asterisk features and modules continue to function.
- Specific Asterisk CLI commands are no longer available.

Example:

```
No such command 'sip show peers'
```

We can presume that something is wrong with **chan_sip** module since we know it provides the 'sip' commands and sub-commands.

Problem

Asterisk has started successfully and the module providing the missing functionality either didn't load at all, or it loaded but isn't running.

The reason for the failure to load or run is typically invalid configuration or a failure to parse the configuration for the module.

On this Page

Solution

Identify the state of the module. If the module is loaded but not running, or not loaded at all, then resolve file format, configuration syntax issues or unwanted modules.conf configuration for the specific module. Restart Asterisk.

Troubleshooting

Check Module Loaded and Running States

From the Asterisk CLI you can use the 'module show' commands to identify the state of a module.

Previous to Asterisk 12, you could only see if the module is loaded. However it may not actually be running (usable).

```
*CLI> module show like chan_sip.so
Module                         Description                              Use Count
chan_sip.so                    Session Initiation Protocol (SIP)        0
1 modules loaded
```

In Asterisk 12 and beyond you can quickly see if a module is loaded and whether it is running or not.

```
*CLI> module show like chan_sip.so
Module                         Description                              Use Count  Status
chan_sip.so                    Session Initiation Protocol (SIP)        0          Not Running
1 modules loaded
```

Make sure Asterisk is configured to load the module

Modules.conf is a core configuration file that includes parameters affecting module loading and loading order. There are a few items to check.

Verify that **autoload=yes** is enabled if you are intending to load modules from the Asterisk modules directory automatically.

Verify that there is **not** a '**noload'** line for the module that is failing to load. That is, if we had a line as follows:

```
noload => chan_sip.so
```

That would tell Asterisk to not load chan_sip.so.

If you are not using **autoload**, then be sure you have a **load** line for the module you desire to load.

```
load => chan_sip.so
```

Check For Module Loading Issues on Asterisk Startup

Now that we know the suspect module should be loading, we can look at some logs that may tell us what is happening.

Stop Asterisk

Be sure Asterisk is stopped to avoid issues with making the logs confusing.

```
asterisk -rx "core stop now"
```

or

```
service asterisk stop
```

Enable logging channels

You can read in detail about Logging facilities on the wiki. In short, for this example, make sure you have the following lines uncommented in your logger.conf file.

```
[logfiles]
full => notice,warning,error,debug,verbose
```

Clear out old logs

You don't want to mistakenly look at an older log where Asterisk was loading appropriately.

Remove the most recent log file, or else move it somewhere you want to keep it.

```
# rm /var/log/asterisk/full
```

Start Asterisk with appropriate log levels

It is important to start Asterisk with log levels that will provide us enough information.

```
# asterisk -cvvvvvddd
```

You'll see a lot of information output in the terminal as Asterisk loads.

Stop Asterisk after it has finished loading

After the output calms down and Asterisk has finished loading, go ahead and stop Asterisk. The logs should have already been recorded.

```
*CLI> core stop now
```

Search logs for lines related to suspect module

Search the log file using keywords based on the specific module that appeared to be failing to load or run.

```
/var/log/asterisk# grep -i chan_sip full
[Oct  9 14:54:43] VERBOSE[21809] chan_sip.c: SIP channel loading...
[Oct  9 14:54:43] ERROR[21809] chan_sip.c: Contents of sip.conf are invalid and cannot be parsed

/var/log/asterisk# grep -i sip.conf full
[Oct  9 14:54:43] DEBUG[21809] config.c: Parsing /etc/asterisk/sip.conf
[Oct  9 14:54:43] VERBOSE[21809] config.c:   == Parsing '/etc/asterisk/sip.conf': Found
[Oct  9 14:54:43] WARNING[21809] config.c: parse error: No category context for line 1 of /etc/asterisk/sip.conf
[Oct  9 14:54:43] ERROR[21809] chan_sip.c: Contents of sip.conf are invalid and cannot be parsed
[Oct  9 14:54:55] DEBUG[21809] config.c: Parsing /etc/asterisk/sip.conf
[Oct  9 14:54:55] VERBOSE[21809] config.c:   == Parsing '/etc/asterisk/sip.conf': Found
[Oct  9 14:54:55] WARNING[21809] config.c: parse error: No category context for line 1 of /etc/asterisk/sip.conf
```

Based on the lines found, you can then use an editor like VIM to view the full log and jump to where the relevant messages are.

```
[Oct  9 14:54:43] VERBOSE[21809] chan_sip.c: SIP channel loading...
[Oct  9 14:54:43] DEBUG[21809] config.c: Parsing /etc/asterisk/sip.conf
[Oct  9 14:54:43] VERBOSE[21809] config.c:   == Parsing '/etc/asterisk/sip.conf': Found
[Oct  9 14:54:43] WARNING[21809] config.c: parse error: No category context for line 1 of /etc/asterisk/sip.conf
[Oct  9 14:54:43] ERROR[21809] chan_sip.c: Contents of sip.conf are invalid and cannot be parsed
```

In this case, not much more is revealed past what we saw with grep. You can see that Asterisk tries to load and run chan_sip, it fails because the contents of sip.conf are invalid and cannot be parsed. The most specific clue is the WARNING:

```
WARNING[21809] config.c: parse error: No category context for line 1 of /etc/asterisk/sip.conf
```

Edit the related config file to resolve the issue

If we look at line 1 of sip.conf we'll spot the root problem.

```
general]
context=public
allowoverlap=no
```

For our example, a square bracket is missing from the context definition! Fix this issue, restart Asterisk and things should work assuming I don't have any other syntax errors.

Unable to connect to remote Asterisk

Overview

When first learning Asterisk some users will find they are unable to connect to the Asterisk service.

You may see the below message after running some variation of `asterisk -r`

```
Unable to connect to remote asterisk (does /var/run/asterisk/asterisk.ctl exist?)
```

First, let's break down the message.

- "Unable to connect to remote asterisk"

This means you are attempting to connect to Asterisk with a remote console. That is, you are using "asterisk -r" as opposed to "asterisk -c".

- "does /var/run/asterisk/asterisk.ctl exist?"

This is letting you know specifically what file is potentially missing or unable to be accessed.

Now let's find out what asterisk.ctl is and investigate potential causes of this error.

The asterisk.ctl file

asterisk.ctl is a UNIX Domain Socket file. It is used to pass commands to an Asterisk process and it is how the Asterisk console ("asterisk r" or any variation) communicates with the back-end Asterisk process.

The /var/run/asterisk/ directory

This directory is the default Asterisk run directory. Asterisk will create it the first time it is run. This location is defined in the Asterisk Main Configuration File. As is explained in the Directory and File Structure section; when Asterisk is running, you'll see two files here, **asterisk.ctl** and **asterisk.pid**. These are the control socket and the PID(Process ID) files for Asterisk.

Potential Causes and Solutions

Cause 1: asterisk.ctl does exist. Your user does not have write permission for this file.

Solution:

1. To verify, check the permissions of the asterisk.ctl file and also check what user you are currently running as.
2. Switch to the correct user that has access to the /var/run/asterisk/ directory and asterisk.ctl file, or provide your current user with the correct permissions.

Cause 2: Permissions issue. Asterisk does not have write access to /var/run/asterisk or your user doesn't have write access to asterisk.ctl.

Solution:

1. If /var/run/asterisk does not exist then create it and assign permission to it such that the user that runs Asterisk will have write and read access.
2. If it already exists, simply verify and assign the correct permissions to the directory. You probably want to double-check what user runs Asterisk.

Cause 3: asterisk.ctl does not exist because Asterisk isn't running. When Asterisk is started it may run briefly and then quickly halt due to an error.

Solution:

You need to find out what error is causing Asterisk to halt and resolve it.

- The quick way is to run Asterisk in root/core console mode and watch for the last messages Asterisk prints out before halting.

  -
    ```
    asterisk -cvvvvv
    ```

    This will start Asterisk in console mode with level 5 verbosity. That should give you plenty to look at.
- Another way is to setup Asterisk Logging to log what you want to see to a file. You'll need to read up on Asterisk's Logging Configuration
- Asterisk could halt for a variety of reasons. The last messages you see before Asterisk halts will give you a clue. Then you can Google from there or ask the user community.

Cause 4: SELinux enabled and not properly configured for Asterisk. SELinux not allowing asterisk.ctl to be created.

Solution:

Configuring SELinux is outside the scope of this article.

- Consult an experienced SELinux user or SELinux documentation on how to configure it properly.
- Disable SELinux if you don't require it (Not recommended)

IPv6 Support

Overview

Since Asterisk 12, IPv6 is supported by the most commonly used components of Asterisk which support IP based communication. This includes the latest SIP channel driver chan_pjsip as well as the older chan_sip. IPv6 support may be spotty before Asterisk 12. For sufficient IPv6 support it is recommended that you upgrade to Asterisk 13 or greater.

Configuration

Fortunately, the configuration is easy and most things will simply work. For channel technologies such as chan_pjsip, chan_sip or chan_iax2 the IPv6 support must be configured in the channel's configuration file. (i.e. pjsip.conf, sip.conf or iax.conf). In each configuration file there are typically options referring to a general bind address or specific TCP or UDP bind addresses. Other than configuring those options to bind to IPv6 interfaces there should be few other options needed. The documentation or configuration samples for each driver should make this clear.

Links to specific instructions:

- Configuring res_pjsip for IPv6
- Configuring chan_sip for IPv6
- Configuring IAX for IPv6
- Configuring ACLs for IPv6

At the time of writing, DUNDi does not support IPv6.

PSTN Connectivity

Advice of Charge

Written by: David Vossel
Initial version: 04-19-2010
Email: dvossel@digium.com

This document is designed to give an overview of how to configure and generate Advice of Charge along with a detailed explanation of how each option works.

Read This First

PLEASE REPORT ANY ISSUES ENCOUNTERED WHILE USING AOC. This feature has had very little community feedback so far. If you are using this feature please share with us any problems you are having and any improvements that could make this feature more useful. Thank you!

Terminology

AOC: Advice of Charge

AOC-S: Advice of Charge message sent at the beginning of a call during call setup. This message contains a list of rates associated with the call.

AOC-D: Advice of Charge message sent during the call. This message is typically used to update the endpoint with the current call charge.

AOC-E: Advice of Charge message sent at the end of a call. This message is used to indicate to the endpoint the final call charge.

AMI: Asterisk Manager Interface. This interface is used to generate AOC messages and listen for AOC events.

AOC in chan_dahdi

LibPRI Support:
ETSI, or euroisdn, is the only switchtype that LibPRI currently supports for AOC.

Enable AOC Pass-through in chan_dahdi
To enable AOC pass-through between the ISDN and Asterisk use the 'aoc_enable' config option. This option allows for any combination of AOC-S, AOC-D, and AOC-E to be enabled or disabled.

For example:

```
aoc_enable=s,d,e ; enables pass-through of AOC-S, AOC-D, and AOC-E

aoc_enable=s,d   ; enables pass-through of AOC-S and AOC-D. Rejects
; AOC-E and AOC-E request messages
```

Since AOC messages are often transported on facility messages, the 'facilityenable' option must be enabled as well to fully support AOC pass-through.

Handling AOC-E in chan_dahdi
Whenever a dahdi channel receives an AOC-E message from Asterisk, it stores that message to deliver it at the appropriate time during call termination. This means that if two AOC-E messages are received on the same call, the last one will override the first one and only one AOC-E message will be sent during call termination.

There are some tricky situations involving the final AOC-E message. During a bridged call, if the endpoint receiving the AOC messages terminates the call before the endpoint delivering the AOC does, the final AOC-E message sent by the sending side during termination will never make it to the receiving end because Asterisk will have already torn down that channel.
This is where the chan_dahdi.conf 'aoce_delayhangup' option comes into play.

By enabling 'aoce_delayhangup', anytime a hangup is initiated by the ISDN side of an Asterisk channel, instead of hanging up the channel, the channel sends a unique internal AOC-E termination request to its bridge channel. This indicates it is about to hangup and wishes to receive the final AOC-E message from the bridged channel before completely tearing down. If the bridged channel knows what to do with this AOC-E termination request, it will do whatever is necessary to indicate to its endpoint that the call is being terminated without actually hanging up the Asterisk channel. This allows the final AOC-E message to come in and be sent across the bridge while both channels are still up. If the channel delaying its hangup for the final AOC-E message times out, the call will be torn down just as it normally would. In chan_dahdi the timeout period is 1/2 the T305 timer which by default is 15 seconds.

'aoce_delayhangup' currently only works when both bridged channels are dahdi_channels. If a SIP channel receives an AOC-E termination request, it just responds by immediately hanging up the channel. Using this option when bridged to any channel technology besides SIP or DAHDI will result in the 15 second timeout period before tearing down the call completely.

Requesting AOC services

AOC can be requested on a call by call basis using the DAHDI dialstring option, A(). The A() option takes in 's', 'd', and 'e' parameters which represent the three types of AOC messages, AOC-S, AOC-D, and AOC-E. By using this option Asterisk will indicate to the endpoint during call setup that it wishes to receive the specified forms of AOC during the call.

Example Usage in extensions.conf

```
exten => 1111,1,Dial(DAHDI/g1/1112/A(s,d,e) ; requests AOC-S, AOC-D, and AOC-E on
; call setup
exten => 1111,1,Dial(DAHDI/g1/1112/A(d,e)  ; requests only AOC-D, and AOC-E on
; call setup
```

AOC in chan_sip

Asterisk supports a very basic way of sending AOC on a SIP channel to Snom phones using an AOC specification designed by Snom. This support is limited to the sending of AOC-D and AOC-E pass-through messages. No support for AOC-E on call termination is present, so if the Snom endpoint receiving the AOC messages from Asterisk terminates the call, the channel will be torn down before the phone can receive the final AOC-E message.

To enable passthrough of AOC messages via the snom specification, use the 'snom_aoc_enabled' option in sip.conf.

Generate AOC Messages via AMI

Asterisk supports a way to generate AOC messages on a channel via the AMI action AOCMessage. At the moment the AOCMessage action is limited to AOC-D and AOC-E message generation. There are some limitations involved with delivering the final AOC-E message as well. The AOCMessage action has its own detailed parameter documentation so this discussion will focus on higher level use. When generating AOC messages on a Dahdi channel first make sure the appropriate chan_dahdi.conf options are enabled. Without enabling 'aoc_enable' correctly for pass-through the AOC messages will never make it out the pri. The same goes with SIP, the 'snom_aoc_enabled' option must be configured before messages can successfully be set to the endpoint.

AOC-D Message Generation
AOC-D message generation can happen anytime throughout the call. This message type is very straight forward.

Example: AOCMessage action generating AOC-D currency message with Success response.

```
Action: AOCMessage
Channel: DAHDI/i1/1111-1
MsgType: d
ChargeType: Currency
CurrencyAmount: 16
CurrencyName: USD
CurrencyMultiplier: OneThousandth
AOCBillingId: Normal
ActionID: 1234

Response: Success
ActionID: 1234
Message: AOC Message successfully queued on channel
```

AOC-E Message Generation

AOC-E messages are sent during call termination and represent the final charge total for the call. Since Asterisk call termination results in the channel being destroyed, it is currently not possible for the AOCMessage AMI action to be used to send the final AOC-E message on call hangup. There is however a work around for this issue that can be used for Dahdi channels. By default chan_dahdi saves any AOC-E message it receives from Asterisk during a call and waits to deliver that message during call termination. If multiple AOC-E messages are received from Asterisk on the same Dahdi channel, only the last message received is stored for delivery. This means that each new AOC-E message received on the channel overrides the previous one. Knowing this the final AOC-E message can be continually updated on a Dahdi channel until call termination occurs allowing the last update to be sent on hangup. This method is only as accurate as the intervals in which it is updated, but allows some form of AOC-E to be generated.

Example: AOCMessage action generating AOC-E unit message with Success response.

```
Action: AOCMessage
Channel: DAHDI/i1/1111-1
MsgType: e
ChargeType: Unit
UnitAmount(0): 111
UnitType(0): 6
UnitAmount(1): 222
UnitType(1): 5
UnitAmount(2): 333
UnitType(3): 4
UnitAmount(4): 444
AOCBillingId: Normal
ActionID: 1234

Response: Success
ActionID: 1234
Message: AOC Message successfully queued on channel
```

Call Completion Supplementary Services (CCSS)

Introduction

A new feature for Asterisk 1.8 is Call Completion Supplementary Services. This document aims to explain the system and how to use it. In addition, this document examines some potential troublesome points which administrators may come across during their deployment of the feature.

What is CCSS?

Call Completion Supplementary Services (often abbreviated "CCSS" or simply "CC") allow for a caller to let Asterisk automatically alert him when a called party has become available, given that a previous call to that party failed for some reason. The two services offered are Call Completion on Busy Subscriber (CCBS) and Call Completion on No Response (CCNR). To illustrate, let's say that Alice attempts to call Bob. Bob is currently on a phone call with Carol, though, so Alice hears a busy signal. In this situation, assuming that Asterisk has been configured to allow for such activity, Alice would be able to request CCBS. Once Bob has finished his phone call, Alice will be alerted. Alice can then attempt to call Bob again.

CCSS Glossary

In this document, we will use some terms which may require clarification. Most of these terms are specific to Asterisk, and are by no means standard.

- CCBS: Call Completion on Busy Subscriber. When a call fails because the recipient's phone is busy, the caller will have the opportunity to request CCBS. When the recipient's phone is no longer busy, the caller will be alerted. The means by which the caller is alerted is dependent upon the type of agent used by the caller.
- CCNR: Call Completion on No Response. When a call fails because the recipient does not answer the phone, the caller will have the opportun- ity to request CCNR. When the recipient's phone becomes busy and then is no longer busy, the caller will be alerted. The means by which the caller is alerted is dependent upon the type of the agent used by the caller.
- Agent: The agent is the entity within Asterisk that communicates with and acts on behalf of the calling party.
- Monitor: The monitor is the entity within Asterisk that communicates with and monitors the status of the called party.
- Generic Agent: A generic agent is an agent that uses protocol-agnostic methods to communicate with the caller. Generic agents should only be used for phones, and never should be used for "trunks."
- Generic Monitor: A generic monitor is a monitor that uses protocol- agnostic methods to monitor the status of the called party. Like with generic agents, generic monitors should only be used for phones.
- Native Agent: The opposite of a generic agent. A native agent uses protocol-specific messages to communicate with the calling party. Native agents may be used for both phones and trunks, but it must be known ahead of time that the device with which Asterisk is communica- ting supports the necessary signaling.
- Native Monitor: The opposite of a generic monitor. A native monitor uses protocol-specific messages to subscribe to and receive notification of the status of the called party. Native monitors may be used for both phones and trunks, but it must be known ahead of time that the device with which Asterisk is communicating supports the necessary signaling.
- Offer: An offer of CC refers to the notification received by the caller that he may request CC.
- Request: When the caller decides that he would like to subscribe to CC, he will make a request for CC. Furthermore, the term may refer to any outstanding requests made by callers.
- Recall: When the caller attempts to call the recipient after being alerted that the recipient is available, this action is referred to as a "recall."

The Call Completion Process

The Initial Call

The only requirement for the use of CC is to configure an agent for the caller and a monitor for at least one recipient of the call. This is controlled using the cc_agent_policy for the caller and the cc_monitor_policy for the recipient. For more information about these configuration settings, see configs/samples/ccss.conf.sample. If the agent for the caller is set to something other than "never" and at least one recipient has his monitor set to something other than "never," then CC will be offered to the caller at the end of the call.

Once the initial call has been hung up, the configured cc_offer_timer for the caller will be started. If the caller wishes to request CC for the previous call, he must do so before the timer expires.

Requesting CC

Requesting CC is done differently depending on the type of agent the caller is using.

With generic agents, the CallCompletionRequest application must be called in order to request CC. There are two different ways in which this may be called. It may either be called before the caller hangs up during the initial call, or the caller may hang up from the initial call and dial an extension which calls the CallCompletionRequest application. If the second method is used, then the caller will have until the cc_offer_timer expires to request CC.

With native agents, the method for requesting CC is dependent upon the technology being used, coupled with the make of equipment. It may be possible to request CC using a programmable key on a phone or by clicking a button on a console. If you are using equipment which can natively support CC but do not know the means by which to request it, then contact the equipment manufacturer for more information.

Cancelling CC

CC may be canceled after it has been requested. The method by which this is accomplished differs based on the type of agent the calling party uses.

When using a generic agent, the dialplan application CallRequestCancel is used to cancel CC. When using a native monitor, the method by which CC is cancelled depends on the protocol used. Likely, this will be done using a button on a phone.

Keep in mind that if CC is cancelled, it cannot be un-cancelled.

Monitoring the Called Party

Once the caller has requested CC, then Asterisk's job is to monitor the progress of the called parties. It is at this point that Asterisk allocates the necessary resources to monitor the called parties.

A generic monitor uses Asterisk's device state subsystem in order to determine when the called party has become available. For both CCBS and CCNR, Asterisk simply waits for the phone's state to change to a "not in use" state from a different state. Once this happens, then Asterisk will consider the called party to be available and will alert the caller.

A native monitor relies on the network to send a protocol-specific message when the called party has become available. When Asterisk receives such a message, it will consider the called party to be available and will alert the caller.

Note that since a single caller may dial multiple parties, a monitor is used for each called party. It is within reason that different called parties will use different types of monitors for the same CC request.

Alerting the Caller

Once Asterisk has determined that the called party has become available the time comes for Asterisk to alert the caller that the called party has become available. The method by which this is done differs based on the type of agent in use.

If a generic agent is used, then Asterisk will originate a call to the calling party. Upon answering the call, if a callback macro has been configured, then that macro will be executed on the calling party's channel. After the macro has completed, an outbound call will be issued to the parties involved in the original call.

If a native agent is used, then Asterisk will send an appropriate notification message to the calling party to alert it that it may now attempt its recall. How this is presented to the caller is dependent upon the protocol and equipment that the caller is using. It is possible that the calling party's phone will ring and a recall will be triggered upon answering the phone, or it may be that the user has a specific button that he may press to initiate a recall.

If the Caller is unavailable

When the called party has become available, it is possible that when Asterisk attempts to alert the calling party of the called party's availability, the calling party itself will have become unavailable. If this is the case, then Asterisk will suspend monitoring of the called party and will instead monitor the availability of the calling party. The monitoring procedure for the calling party is the same as is used in the section "Monitoring the Called Party." In other words, the method by which the calling party is monitored is dependent upon the type of agent used by the caller.

Once Asterisk has determined that the calling party has become available again, Asterisk will then move back to the process used in the section "Monitoring the Called Party."

The CC recall

The calling party will make its recall to the same extension that was dialed. Asterisk will provide a channel variable, CC_INTERFACES, to be used as an argument to the Dial application for CC recalls. It is strongly recommended that you use this channel variable during a CC recall. Listed are two reasons:

1. The dialplan may be written in such a way that the dialed destintations are dynamically generated. With such a dialplan, it cannot be guaranteed that the same interfaces will be recalled.
2. For calling destinations with native CC monitors, it may be necessary to dial a special string in order to notify the channel driver that the number being dialed is actually part of a CC recall.

> ⚠ Even if your call gets routed through local channels, the CC_INTERFACES variable will be populated with the appropriate values for that specific extension.

When the called parties are dialed, it is expected that a called party will answer, since Asterisk had previously determined that the party was available. However, it is possible that the called party may choose not to respond to the call, or he could have become busy again. In such a situation, the calling party must re-request CC if he wishes to still be alerted when the calling party has become available.

Call Completion Info and Tips

- Be aware when using a generic agent that the max_cc_agents configuration parameter is ignored. The main driving reason for this is that the mechanism for cancelling CC when using a generic agent would become much more potentially confusing to execute. By limiting a calling party to having a single request, there is only ever a single request to be cancelled, making the process simple.

- Keep in mind that no matter what CC agent type is being used, a CC request can only be made for the latest call issued.

- If available timers are running on multiple called parties, it is possible that one of the timers may expire before the others do. If such a situation occurs, then the interface on which the timer expired will cease to be monitored. If, though, one of the other called parties becomes available before his available timer expires, the called party whose available timer had previously expired will still be included in the CC_INTERFACES channel variable on the recall.
- It is strongly recommended that lots of thought is placed into the settings of the CC timers. Our general recommendation is that timers for phones should be set shorter than those for trunks. The reason for this is that it makes it less likely for a link in the middle of a network to cause CC to fail.

- CC can potentially be a memory hog if used irresponsibly. The following are recommendations to help curb the amount of resources required by the CC engine. First, limit the maximum number of CC requests in the system using the cc_max_requests option in ccss.conf. Second, set the cc_offer_timer low for your callers. Since it is likely that most calls will not result in a CC request, it is a good idea to set this value to something low so that information for calls does not stick around in memory for long. The final thing that can be done is to conditionally set the cc_agent_policy to "never" using the CALLCOMPLETION dialplan function. By doing this, no CC information will be kept around after the call completes.

- It is possible to request CCNR on answered calls. The reason for this is that it is impossible to know whether a call that is answered has actually been answered by a person or by something such as voicemail or some other IVR.

- Not all channel drivers have had the ability to set CC config parameters in their configuration files added yet. At the time of this writing (2009 Oct), only chan_sip has had this ability added, with short-term plans to add this to chan_dahdi as well. It is possible to set CC configuration parameters for other channel types, though. For these channel types, the setting of the parameters can only be accomplished using the CALLCOMPLETION dialplan function.

- It is documented in many places that generic agents and monitors can only be used for phones. In most cases, however, Asterisk has no way of distinguishing between a phone and a trunk itself. The result is that Asterisk will happily let you violate the advice given and allow you to set up a trunk with a generic monitor or agent. While this will not cause anything catastrophic to occur, the behavior will most definitely not be what you want.

- At the time of this writing (2009 Oct), Asterisk is the only known SIP stack to write an implementation of draft-ietf-bliss-call-completion-04. As a result, it is recommended that for your SIP phones, use a generic agent and monitor. For SIP trunks, you will only be able to use CC if the other end is terminated by another Asterisk server running version 1.8 or later.

- Native SIP CC will only work if the xml2 development library is installed. This is because we use libxml2 in order to parse PIDF bodies of PUBLISH messages received. If, at configure time, Asterisk cannot detect that the necessary library is installed, then native CC in SIP will be disabled. Attempts to set a channel or SIP peer to use native CC will be changed to having CC being disabled instead.

- If the Dial application is called multiple times by a single extension, CC will only be offered to the caller for the parties called by the first instantiation of Dial.

- If a phone forwards a call, then CC may only be requested for the phone that executed the call forward. CC may not be requested for the phone to which the call was forwarded.
- CC is currently only supported by the Dial application. Queue, Followme, and Page do not support CC because it is not particularly useful for those applications.

> ✓
>
> - Generic CC relies heavily on accurate device state reporting. In particular, when using SIP phones it is vital to be sure that device state is updated properly when using them. In order to facilitate proper device state handling, be sure to set callcounter=yes for all peers and to set limitonpeers=yes in the general section of sip.conf

- When using SIP CC (i.e. native CC over SIP), it is important that your minexpiry and maxexpiry values allow for available timers to run as little or as long as they are configured. When an Asterisk server requests call completion over SIP, it sends a SUBSCRIBE message with an Expires header set to the number of seconds that the available timer should run. If the Asterisk server that receives this SUBSCRIBE has a maxexpiry set lower than what is in the received Expires header, then the available timer will only run for maxexpiry seconds.

- CC support for ETSI PTP and Q.SIG requires CallerID information to match CC requests with CC offers. For Q.SIG, depending upon the options negotiated when CC is requested, the CallerID information needs to be callable as well.

- As with all Asterisk components, CC is not perfect. If you should find a bug or wish to enhance the feature, please open an issue on https://issues.asterisk.org. If writing an enhancement, please be sure to include a patch for the enhancement, or else the issue will be closed.

Generic Call Completion Example

The following is an incredibly bare-bones example sip.conf and dialplan to show basic usage of generic call completion. It is likely that if you have a more complex setup, you will need to make use of items like the CALLCOMPLETION dialplan function or the CC_INTERFACES channel variable.
First, let's establish a very simple sip.conf to use for this

sip.conf

```
[Mark]
context=phone_calls
cc_agent_policy=generic
cc_monitor_policy=generic ;We will accept defaults for the rest of the cc parameters
;We also are not concerned with other SIP details for this
;example

[Richard]
context=phone_calls
cc_agent_policy=generic
cc_monitor_policy=generic
```

Now, let's write a simple dialplan

extensions.conf

```
[phone_calls]
exten => 1000,1,Dial(SIP/Mark,20)
exten => 1000,n,Hangup
exten => 2000,1,Dial(SIP/Richard,20)
exten => 2000,n,Hangup
exten => 30,1,CallCompletionRequest
exten => 30,n,Hangup
exten => 31,1,CallCompletionCancel
exten => 31,n,Hangup
```

Scenario 1: Mark picks up his phone and dials Richard by dialing 2000. Richard is currently on a call, so Mark hears a busy signal. Mark then hangs up, picks up the phone and dials 30 to call the CallCompletionRequest application. After some time, Richard finishes his call and hangs up. Mark is automatically called back by Asterisk. When Mark picks up his phone, Asterisk will dial extension 2000 for him.

Scenario 2: Richard picks up his phone and dials Mark by dialing 1000. Mark has stepped away from his desk, and so he is unable to answer the phone within the 20 second dial timeout. Richard hangs up, picks the phone back up and then dials 30 to request call completion. Mark gets back to his desk and dials somebody's number. When Mark finishes the call, Asterisk detects that Mark's phone has had some activity and has become available again and rings Richard's phone. Once Richard picks up, Asterisk automatically dials exteision 1000 for him.

Scenario 3: Much like scenario 1, Mark calls Richard and Richard is busy. Mark hangs up, picks the phone back up and then dials 30 to request call completion. After a little while, Mark realizes he doesn't actually need to talk to Richard, so he dials 31 to cancel call completion. When Richard becomes free, Mark will not automatically be redialed by Asterisk.

Scenario 4: Richard calls Mark, but Mark is busy. About thirty seconds later, Richard decides that he should perhaps request call completion. However, since Richard's phone has the default cc_offer_timer of 20 seconds, he has run out of time to request call completion. He instead must attempt to dial Mark again manually. If Mark is still busy, Richard can attempt to request call completion on this second call instead.

Scenario 5: Mark calls Richard, and Richard is busy. Mark requests call completion. Richard does not finish his current call for another 2 hours (7200 seconds). Since Mark has the default ccbs_available_timer of 4800 seconds set, Mark will not be automatically recalled by Asterisk when Richard finishes his call.

Scenario 6: Mark calls Richard, and Richard does not respond within the 20 second dial timeout. Mark requests call completion. Richard does not use his phone again for another 4 hours (144000 seconds). Since Mark has the default ccnr_available_timer of 7200 seconds set, Mark will not be automatically recalled by Asterisk when Richard finishes his call.

Caller ID in India

India finds itself in a unique situation (hopefully). It has several telephone line providers, and they are not all using the same CID signaling; and the CID signalling is not like other countries.

In order to help those in India quickly find to the CID signaling system that their carrier uses (or range of them), and get the configs right with a minimal amount of experimentation, this file is provided. Not all carriers are covered, and not all mentioned below are complete. Those with updates to this table should post the new information on bug 6683 of the asterisk bug tracker.

Provider: Bharti (is this BSNL?)
Config:

```
cidstart=polarity_in
cidsignalling=dtmf
```

Results: ? (this should work), but needs to be tested?

Tested by: ?

Provider: VSNL
Config:

```
null
```

Results: ?

Tested by: ?

Provider: BSNL
Config:

```
cid_start=ring
cid_signalling=dtmf
```

Results: ?

Tested by: (abhi)

Provider: MTNL, old BSNL
Config:

```
cidsignalling = v23
cidstart=ring
```

Results: works

Tested by: (enterux)

Provider: MTNL (Delhi)
Config:

```
cidsignalling = v23
cidstart = ring
```

or:

```
cidsignalling = dtmf
cidstart = polarity_IN
```

or:

```
cidsignalling = dtmf
cidstart = polarity
```

Results: fails

Tested by: brealer

Provider: TATA

Config:

```
cidsignalling = dtmf
cidstart=polarity_IN
```

Results: works

Tested by: brealer

Asterisk still doesn't work with some of the CID scenarios in India. If you are in India, and not able to make CID work with any of the permutations of cidsignalling and cidstart, it could be that this particular situation is not covered by Asterisk. A good course of action would be to get in touch with the provider, and find out from them exactly how their CID signalling works. Describe this to us, and perhaps someone will be able to extend the code to cover their signaling.

Signaling System Number 7

The LibSS7 project is not actively developed or maintained.

Where to get LibSS7?

Currently, all released branches of Asterisk including trunk use the released versions of libss7.

The latest compatible libss7 code can be obtained from the 1.0 SVN branch

As of 7/2013, the libss7 trunk (http://svn.asterisk.org/svn/libss7/trunk) is currently only usable with this Asterisk branch (http://svn.asterisk.org/svn/asterisk/team/rmudgett/ss7_27_knk) based off of Asterisk trunk.

Tested Switches:

- Siemens EWSD - (ITU style) MTP2 and MTP3 comes up, ISUP inbound and outbound calls work as well.
- DTI DXC 4K - (ANSI style) 56kbps link, MTP2 and MTP3 come up, ISUP inbound and outbound calls work as well.
- Huawei M800 - (ITU style) MTP2 and MTP3 comes up, ISUP National, International inbound and outbound calls work as well, CallerID presentation&screening work.

and MORE~!

Thanks:

Mark Spencer, for writing Asterisk and libpri and being such a great friend and boss.

Luciano Ramos, for donating a link in getting the first "real" ITU switch working.

Collin Rose and John Lodden, John for introducing me to Collin, and Collin for the first "real" ANSI link and for holding my hand through the remaining changes that had to be done for ANSI switches.

To Use:

In order to use libss7, you must get at least the following versions of DAHDI and Asterisk:
DAHDI: 2.0.x
libss7: trunk (currently, there **only** is a trunk release).
Asterisk: 1.6.x

You must then do a `make; make install` in each of the directories that you installed in the given order (DAHDI first, libss7 second, and Asterisk last).

In order to check out the code, you must have the subversion client installed. This is how to check them out from the public subversion server.

These are the commands you would type to install them:

```
`svn co http://svn.digium.com/svn/dahdi/linux/trunk dahdi-trunk`
`cd dahdi-trunk`
`make; make install`

`svn co http://svn.digium.com/svn/dahdi/tools/trunk dahdi-tools`
`cd dahdi-tools`
`./configure; make; make install`

`svn co http://svn.digium.com/svn/libss7/trunk libss7-trunk`
`cd libss7-trunk`
`make; make install`

`svn co http://svn.digium.com/svn/asterisk/trunk asterisk-trunk`
`cd asterisk-trunk`
`./configure; make; make install;`
```

This should build DAHDI, libss7, and Asterisk with SS7 support.

In the past, there was a special asterisk-ss7 branch to use which contained the SS7 code. That code has been merged back into the trunk version of Asterisk, and the old asterisk-ss7 branch has been deprecated and removed. If you are still using the asterisk-ss7 branch, it will not work against the current version of libss7, and you should switch to asterisk-trunk instead.

Configuration:

In /etc/dahdi/system.conf, your signalling channel(s) should be a "dchan" and your bearers should be set as "bchan".

The sample chan_dahdi.conf contains sample configuration for setting up an E1 link.

In brief, here is a simple ss7 linkset setup:

```
signalling = ss7
ss7type = itu   ; or ansi if you are using an ANSI link

linkset = 1  ; Pick a number for your linkset identifier in chan_dahdi.conf

pointcode = 28  ; The decimal form of your point code.  If you are using an
   ; ANSI linkset, you can use the xxx-xxx-xxx notation for
   ; specifying your link
adjpointcode = 2 ; The point code of the switch adjacent to your linkset

defaultdpc = 3  ; The point code of the switch you want to send your ISUP
   ; traffic to.  A lot of the time, this is the same as your
   ; adjpointcode.

; Now we configure our Bearer channels (CICs)

cicbeginswith = 1 ; Number to start counting the CICs from.  So if DAHDI/1 to
   ; DAHDI/15 are CICs 1-15, you would set this to 1 before you
   ; declare channel=1-15

channel=1-15  ; Use DAHDI/1-15 and assign them to CICs 1-15

cicbeginswith = 17 ; Now for DAHDI/17 to DAHDI/31, they are CICs 17-31 so we initialize
   ; cicbeginswith to 17 before we declare those channels

channel = 17-31  ; This assigns CICs 17-31 to channels 17-31

sigchan = 16  ; This is where you declare which DAHDI channel is your signalling
   ; channel.  In our case it is DAHDI/16.  You can add redundant
   ; signalling channels by adding additional sigchan= lines.

; If we want an alternate redundant signalling channel add this

sigchan = 48  ; This would put two signalling channels in our linkset, one at
   ; DAHDI/16 and one at DAHDI/48 which both would be used to send/receive
   ; ISUP traffic.

; End of chan_dahdi.conf
```

This is how a basic linkset is setup. For more detailed chan_dahdi.conf SS7 config information as well as other options available for that file, see the default chan_dahdi.conf that comes with the samples in asterisk. If you would like, you can do a `make samples` in your asterisk-trunk directory and it will install a sample chan_dahdi.conf for you that contains
more information about SS7 setup.

For more information, you can ask questions of the community on the asterisk-ss7 or asterisk-dev mailing lists.

Secure Calling

Top-level page for articles about securing VoIP calls using encryption.

Secure Calling Specifics

Asterisk supports a channel-agnostic method for handling secure call requirements. Since there is no single meaning of what constitutes a "secure call," Asterisk allows the administrator the control to define "secure" for themselves via the dialplan and channel-specific configuration files.

Channel-specific configuration

Currently the IAX2 and SIP channels support the call security features in Asterisk. Both channel-specific configuration files (iax2.conf and sip.conf) support the encryption=yes setting. For IAX2, this setting causes Asterisk to offer encryption when placing or receiving a call. To force encryption with IAX2, the forceencrypt=yes option is required. Due to limitations of SDP, encryption=yes in sip.conf results in a call with only a secure media offer, therefor forceencrypt=yes would be redundant in sip.conf.

If a peer is defined as requiring encryption but the endpoint does not support it, the call will fail with a HANGUPCAUSE of 58 (bearer capability does not exist).

Security-based dialplan branching

Each channel that supports secure signaling or media can implement a CHANNEL read callback function that specifies whether or not that channel meets the specified criteria. Currently, chan_iax2 and chan_sip implement these callbacks. Channels that do not support secure media or signaling will return an empty string when queried. For example, to only allow an inbound call that has both secure signaling and media, see the following example.

```
exten => 123,1,GotoIf($["${CHANNEL(secure_signaling)}" = "1"]?:fail)
exten => 123,n,GotoIf($["${CHANNEL(secure_media)}" = "1"]?:fail)
exten => 123,n,Dial(SIP/123)
exten => 123,n,Hangup
exten => 123,n(fail),Playback(vm-goodbye)
exten => 123,n,Hangup
```

Forcing bridged channels to be secure

Administrators can force outbound channels that are to be bridged to a calling channel to conform to secure media and signaling policies. For example, to first make a call attempt that has both secure signaling and media, but gracefully fall back to non-secure signaling and media see the following example:

```
exten => 123,1,NoOp(We got a call)
exten => 123,n,Set(CHANNEL(secure_bridge_signaling)=1)
exten => 123,n,Set(CHANNEL(secure_bridge_media)=1)
exten => 123,n,Dial(SIP/somebody)
exten => 123,n,NoOp(HANGUPCAUSE=${HANGUPCAUSE})
exten => 123,n,GotoIf($["${HANGUPCAUSE}"="58"]?encrypt_fail)
exten => 123,n,Hangup ; notify user that retrying via insecure channel (user-provided
prompt)
exten => 123,n(encrypt_fail),Playback(secure-call-fail-retry)
exten => 123,n,Set(CHANNEL(secure_bridge_signaling)=0)
exten => 123,n,Set(CHANNEL(secure_bridge_media)=0)
exten => 123,n,Dial(SIP/somebody)
exten => 123,n,Hangup
```

Secure Calling Tutorial

Overview

⚠ This tutorial makes use of SRTP and TLS. SRTP support was added in Asterisk 1.8, TLS was added in 1.6.

So you'd like to make some secure calls.

Here's how to do it, using Blink, a SIP soft client for Mac OS X, Windows, and Linux. You can find some brief instructions for installing Blink on Ubuntu on the wiki.

These instructions assume that you're running as the root user (sudo su -).

Part 1 (TLS)

Transport Layer Security (TLS) provides encryption for call signaling. It's a practical way to prevent people who aren't Asterisk from knowing who you're calling. Setting up TLS between Asterisk and a SIP client involves creating key files, modifying Asterisk's SIP configuration to enable TLS, creating a SIP peer that's capable of TLS, and modifying the SIP client to connect to Asterisk over TLS.

Keys

First, let's make a place for our keys.

```
mkdir /etc/asterisk/keys
```

Next, use the "ast_tls_cert" script in the "contrib/scripts" Asterisk source directory to make a self-signed certificate authority and an Asterisk certificate.

```
./ast_tls_cert -C pbx.mycompany.com -O "My Super Company" -d /etc/asterisk/keys
```

- The "-C" option is used to define our host - DNS name or our IP address.
- The "-O" option defines our organizational name.
- The "-d" option is the output directory of the keys.

1. You'll be asked to enter a pass phrase for /etc/asterisk/keys/ca.key, put in something that you'll remember for later.
2. This will create the /etc/asterisk/keys/ca.crt file.
3. You'll be asked to enter the pass phrase again, and then the /etc/asterisk/keys/asterisk.key file will be created.
4. The /etc/asterisk/keys/asterisk.crt file will be automatically generated.
5. You'll be asked to enter the pass phrase a third time, and the /etc/asterisk/keys/asterisk.pem will be created, a combination of the asterisk.key and asterisk.crt files.

Next, we generate a client certificate for our SIP device.

```
./ast_tls_cert -m client -c /etc/asterisk/keys/ca.crt -k /etc/asterisk/keys/ca.key -C phone1.mycompany.com -O "My Super Company"
-d /etc/asterisk/keys -o malcolm
```

- The "-m client" option tells the script that we want a client certificate, not a server certificate.
- The "-c /etc/asterisk/keys/ca.crt" option specifies which Certificate Authority (ourselves) that we're using.
- The "-k /etc/asterisk/keys/ca.key" provides the key for the above-defined Certificate Authority.
- The "-C" option, since we're defining a client this time, is used to define the hostname or IP address of our SIP phone
- The "-O" option defines our organizational name.
- The "-d" option is the output directory of the keys."
- The "-o" option is the name of the key we're outputting.

1. You'll be asked to enter the pass phrase from before to unlock /etc/asterisk/keys/ca.key.

Now, let's check the keys directory to see if all of the files we've built are there. You should have:

```
asterisk.crt
asterisk.csr
asterisk.key
asterisk.pem
malcolm.crt
malcolm.csr
malcolm.key
malcolm.pem
ca.cfg
ca.crt
ca.key
tmp.cfg
```

Next, copy the malcolm.pem and ca.crt files to the computer running the Blink soft client.

.p12 Client Certificates
If your client requires a .p12 certificate file instead, you can generate that using openssl like:

```
# openssl pkcs12 -export -out MySuperClientCert.p12 -inkey ca.key -in ca.crt
-certfile asterisk.crt
```

Asterisk chan_pjsip configuration

Now, let's configure Asterisk's PJSIP channel driver to use TLS.

In the **pjsip.conf** configuration file, you'll need to enable a TLS-capable transport. An example of one would resemble:

```
[transport-tls]
type=transport
protocol=tls
bind=0.0.0.0:5061
cert_file=/etc/asterisk/keys/asterisk.crt
priv_key_file=/etc/asterisk/keys/asterisk.key
method=tlsv1
```

Note the **protocol**, **cert_file**, **priv_key_file**, and **method** options. Here, we're using the TLS protocol, we're specifying the keys that we generated earlier for **cert_file** and **priv_key_file** and we're setting the **method** to TLSv1.

Next, you'll need to configure a TLS-capable endpoint. An example of one would resemble:

```
[malcolm]
type=aor
max_contacts=1
remove_existing=yes

[malcolm]
type=auth
auth_type=userpass
username=malcolm
password=useabetterpasswordplease

[malcolm]
type=endpoint
aors=malcolm
auth=malcolm
context=local
disallow=all
allow=g722
dtmf_mode=rfc4733
media_encryption=sdes
```

Note the **media_encryption** option for the endpoint. In this case, we've configured an endpoint that will be using SDES encryption for RTP.

You might be tempted to add a **transport=transport-tls** to the endpoint but in pjproject versions at least as late as 2.4.5, this will cause issues like **Connection refused** in a few situations. Let pjproject do the transport selection on its own. If you still see issues, set **rewrite_contact = yes** in the endpoint configuration.

Asterisk chan_sip configuration

Or, if you are using chan_sip, you can use the following to assist.

In the **sip.conf** configuration file, set the following:

```
tlsenable=yes
tlsbindaddr=0.0.0.0
tlscertfile=/etc/asterisk/keys/asterisk.pem
tlscafile=/etc/asterisk/keys/ca.crt
tlscipher=ALL
tlsclientmethod=tlsv1 ;none of the others seem to work with Blink as the client
```

Here, we're enabling TLS support.
We're binding it to our local IPv4 wildcard (the port defaults to 5061 for TLS).
We've set the TLS certificate file to the one we created above.
We've set the Certificate Authority to the one we created above.
TLS Ciphers have been set to ALL, since it's the most permissive.
And we've set the TLS client method to TLSv1, since that's the preferred one for RFCs and for most clients.

Next, you'll need to configure a SIP peer within Asterisk to use TLS as a transport type. Here's an example:

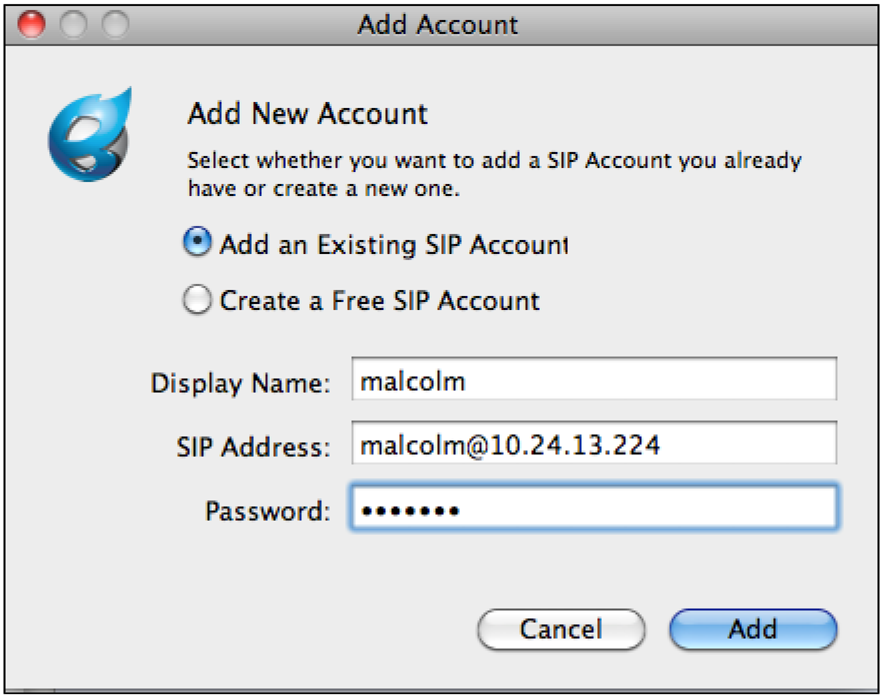
```
[malcolm]
type=peer
secret=malcolm ;note that this is NOT a secure password
host=dynamic
context=local
dtmfmode=rfc2833
disallow=all
allow=g722
transport=tls
```

Notice the **transport** option. The Asterisk SIP channel driver supports three types: udp, tcp and tls. Since we're configuring for TLS, we'll set that. It's also possible to list several supported transport types for the peer by separating them with commas.

Configuring a TLS-enabled SIP client to talk to Asterisk

Next, we'll configure Blink.

First, let's add a new account.

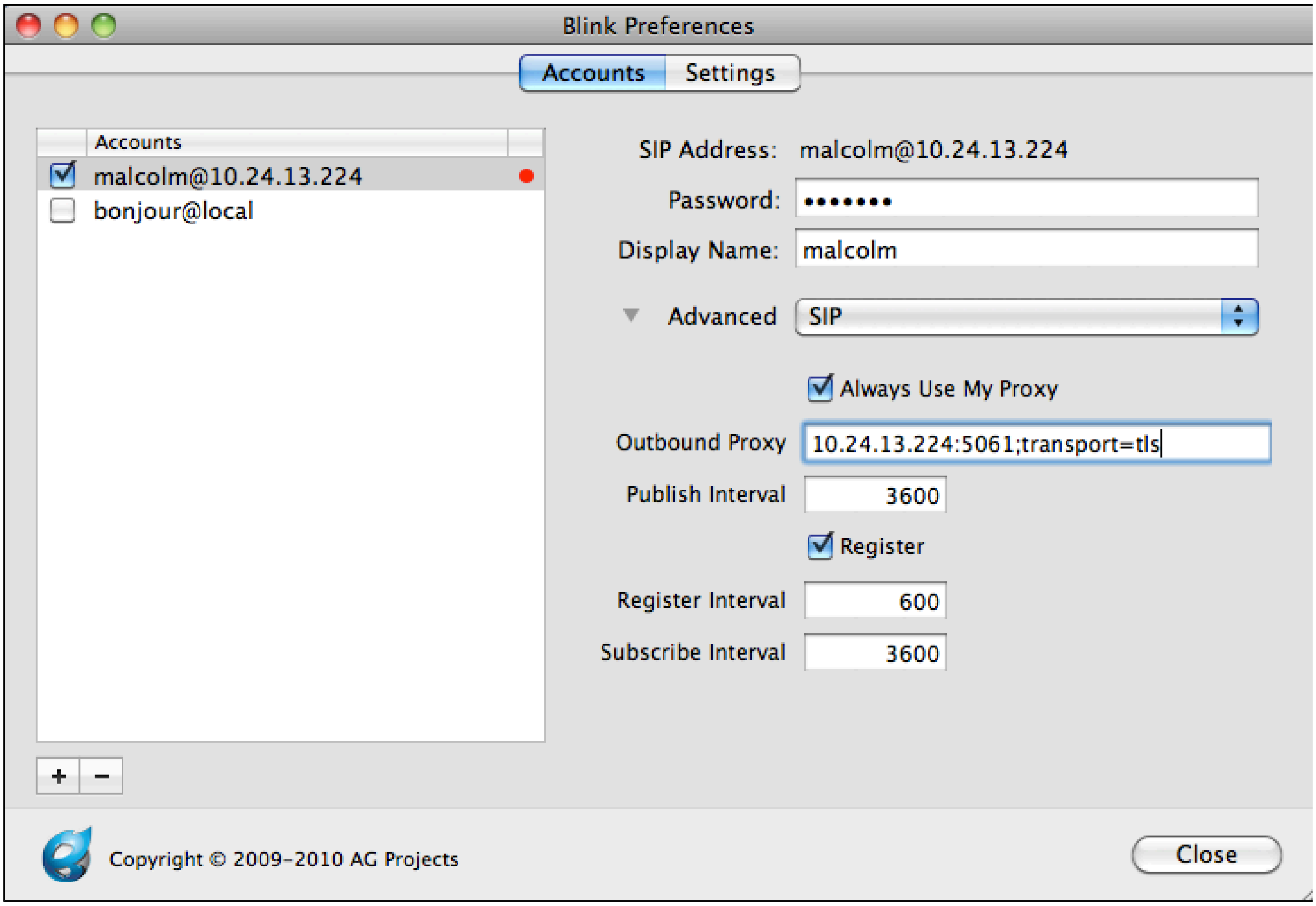


Select whether you want to add a SIP Account you already have or create a new one.

● Add an Existing SIP Account
○ Create a Free SIP Account

Display Name: malcolm
SIP Address: malcolm@10.24.13.224
Password: •••••••

Then, we need to modify the Account Preferences, and under the SIP Settings, we need to set the outbound proxy to connect to the TLS port and transport type on our Asterisk server. In this case, there's an Asterisk server running on port 5061 on host 10.24.13.224.



☑ malcolm@10.24.13.224 ●
□ bonjour@local

SIP Address: malcolm@10.24.13.224
Password: •••••••
Display Name: malcolm
▼ Advanced SIP

☑ Always Use My Proxy
Outbound Proxy 10.24.13.224:5061;transport=tls
Publish Interval 3600
☑ Register
Register Interval 600
Subscribe Interval 3600

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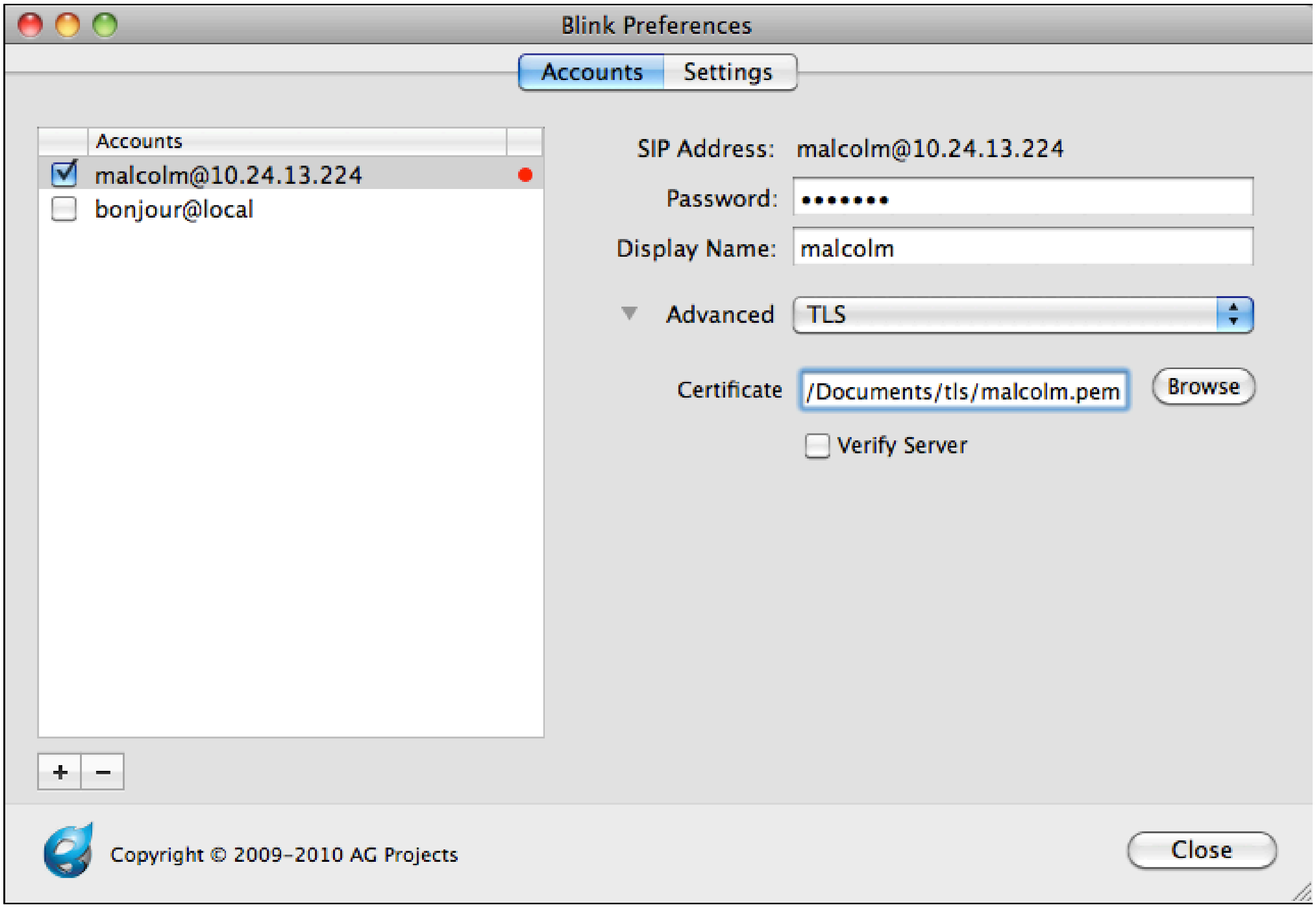
Now, we need to point the TLS account settings to the client certificate (malcolm.pem) that we copied to our computer.



Accounts

☑ malcolm@10.24.13.224 ●

□ bonjour@local

SIP Address: malcolm@10.24.13.224

Password: •••••••

Display Name: malcolm

Advanced: TLS

Certificate: /Documents/tls/malcolm.pem

□ Verify Server

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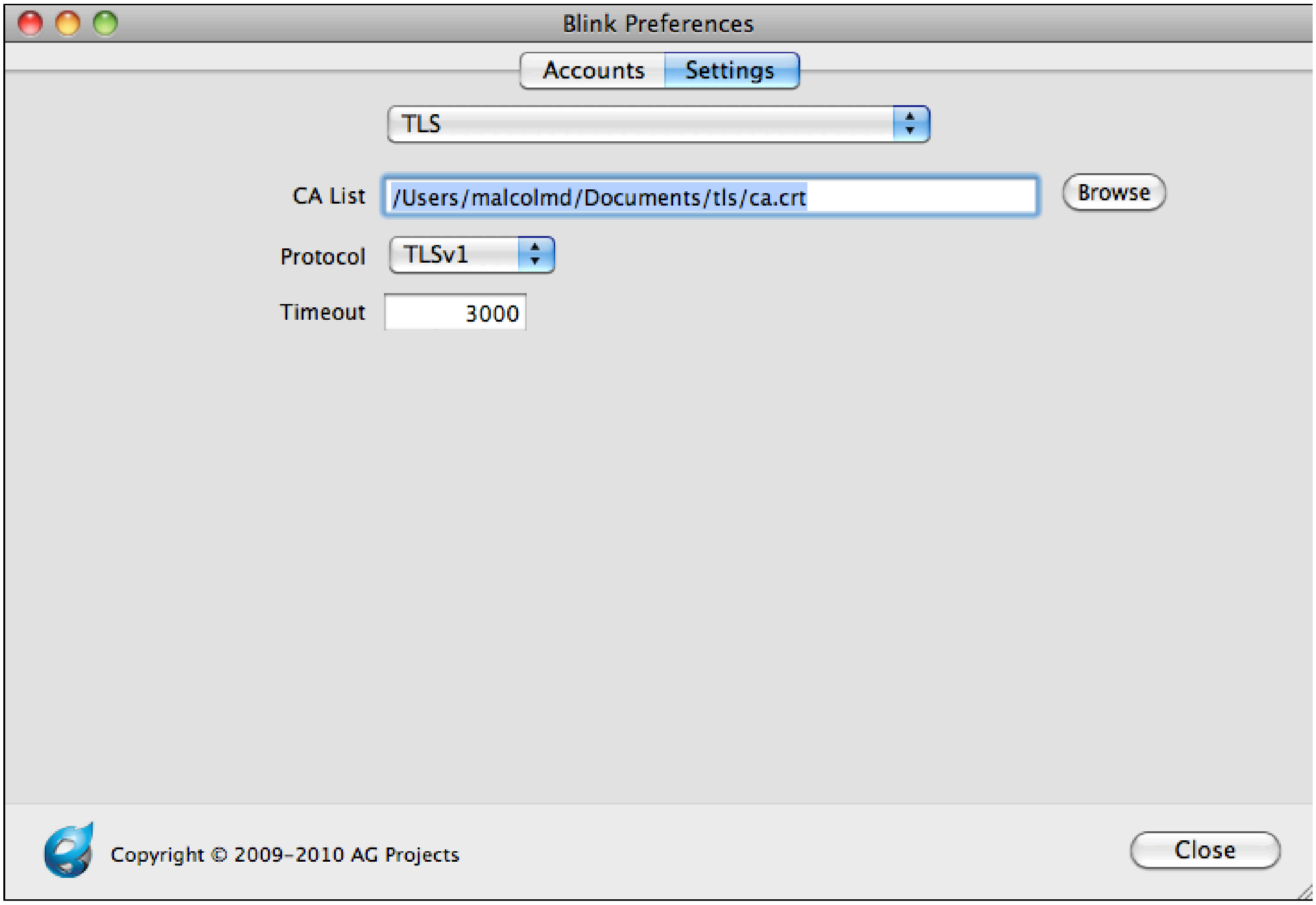
Then, we'll point the TLS server settings to the ca.crt file that we copied to our computer.

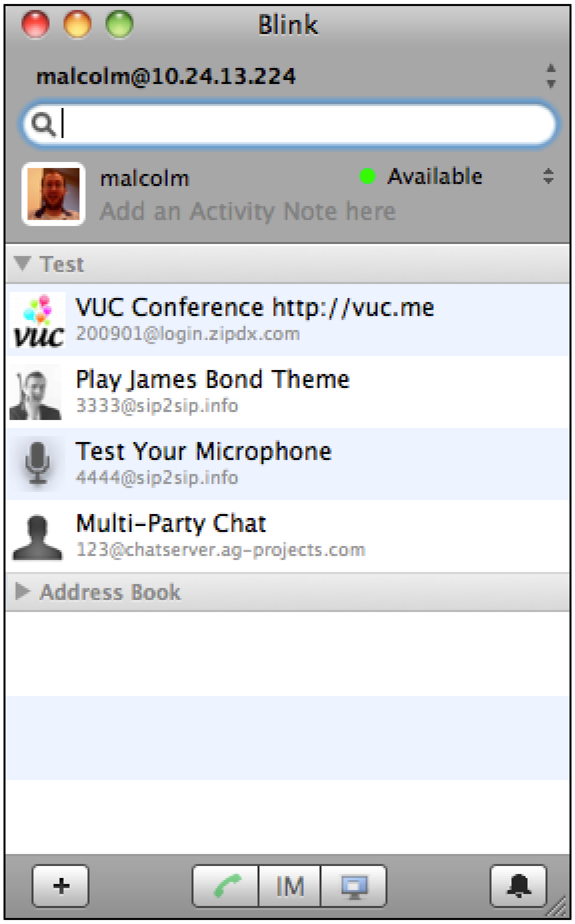


CA List /Users/malcolmd/Documents/tls/ca.crt

Protocol TLSv1

Timeout 3000

Press "close," and you should see Blink having successfully registered to Asterisk.

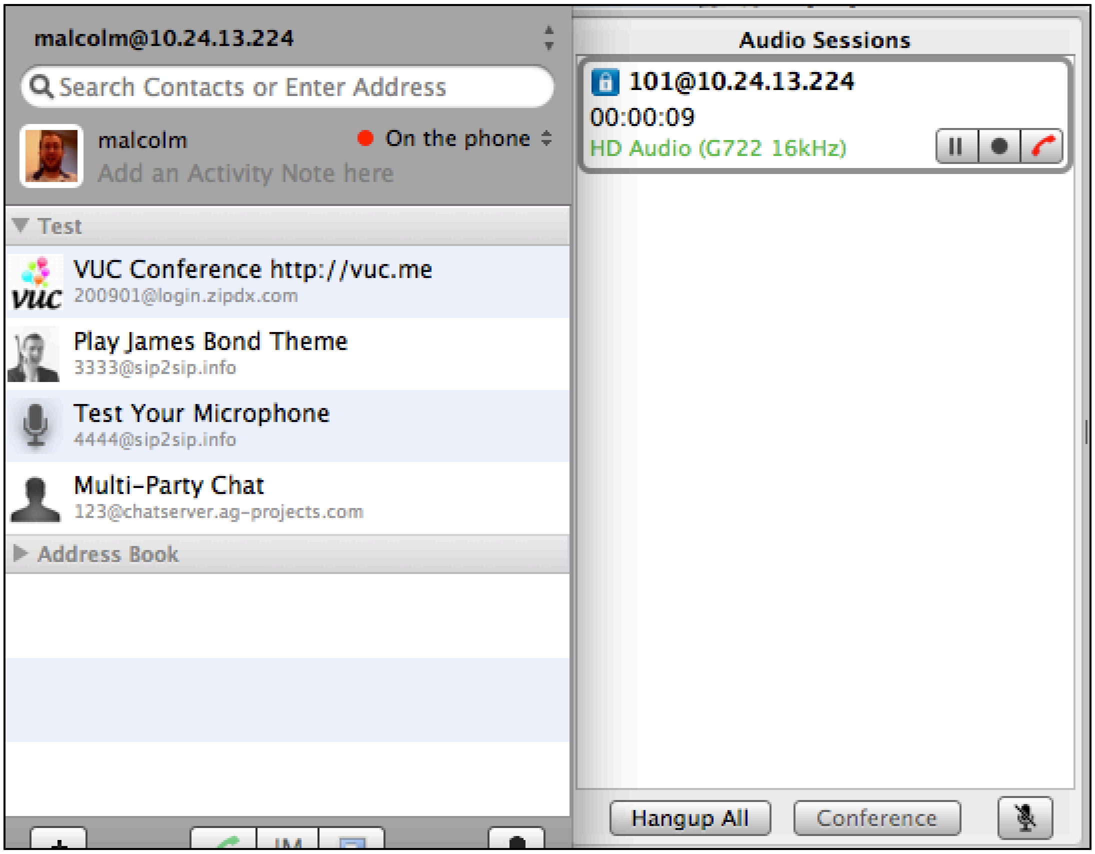


Depending on your Asterisk CLI logging levels, you should see something like:

```
  -- Registered SIP 'malcolm' at 10.24.250.178:5061
       > Saved useragent "Blink 0.22.2 (MacOSX)" for peer malcolm
```

Notice that we registered on port 5061, the TLS port.

Now, make a call. You should see a small secure lockbox in your Blink calling window to indicate that the call was made using secure (TLS) signaling:



Problems with server verification

If the host or IP you used for the common name on your cert doesn't match up with your server then you may run into problems when your client is calling Asterisk. Make sure the client is configured to **not** verify the server against the cert.

When calling **from** Asterisk to Blink or another client, you might run into an ERROR on the Asterisk CLI similar to this:

```
[Jan 29 16:04:11] DEBUG[11217]: tcptls.c:248 handle_tcptls_connection:  SSL Common Name compare s1='10.24.18.124'
s2='phone1.mycompany.com'
[Jan 29 16:04:11] ERROR[11217]: tcptls.c:256 handle_tcptls_connection: Certificate common name did not match (10.24.18.124)
```

This is the opposite scenario, where Asterisk is acting as the client and by default attempting to verify the destination server against the cert.

You can set **tlsdontverifyserver=yes** in sip.conf to prevent Asterisk from attempting to verify the server.

```
;tlsdontverifyserver=[yes|no]
;        If set to yes, don't verify the servers certificate when acting as
;        a client.  If you don't have the server's CA certificate you can
;        set this and it will connect without requiring tlscafile to be set.
;        Default is no.
```

Part 2 (SRTP)

Now that we've got TLS enabled, our signaling is secure - so no one knows what extensions on the PBX we're dialing. But, our media is still not secure - so someone can snoop our RTP conversations from the wire. Let's fix that.

SRTP support is provided by libsrtp. libsrtp has to be installed on the machine before Asterisk is compiled, otherwise you're going to see something like:

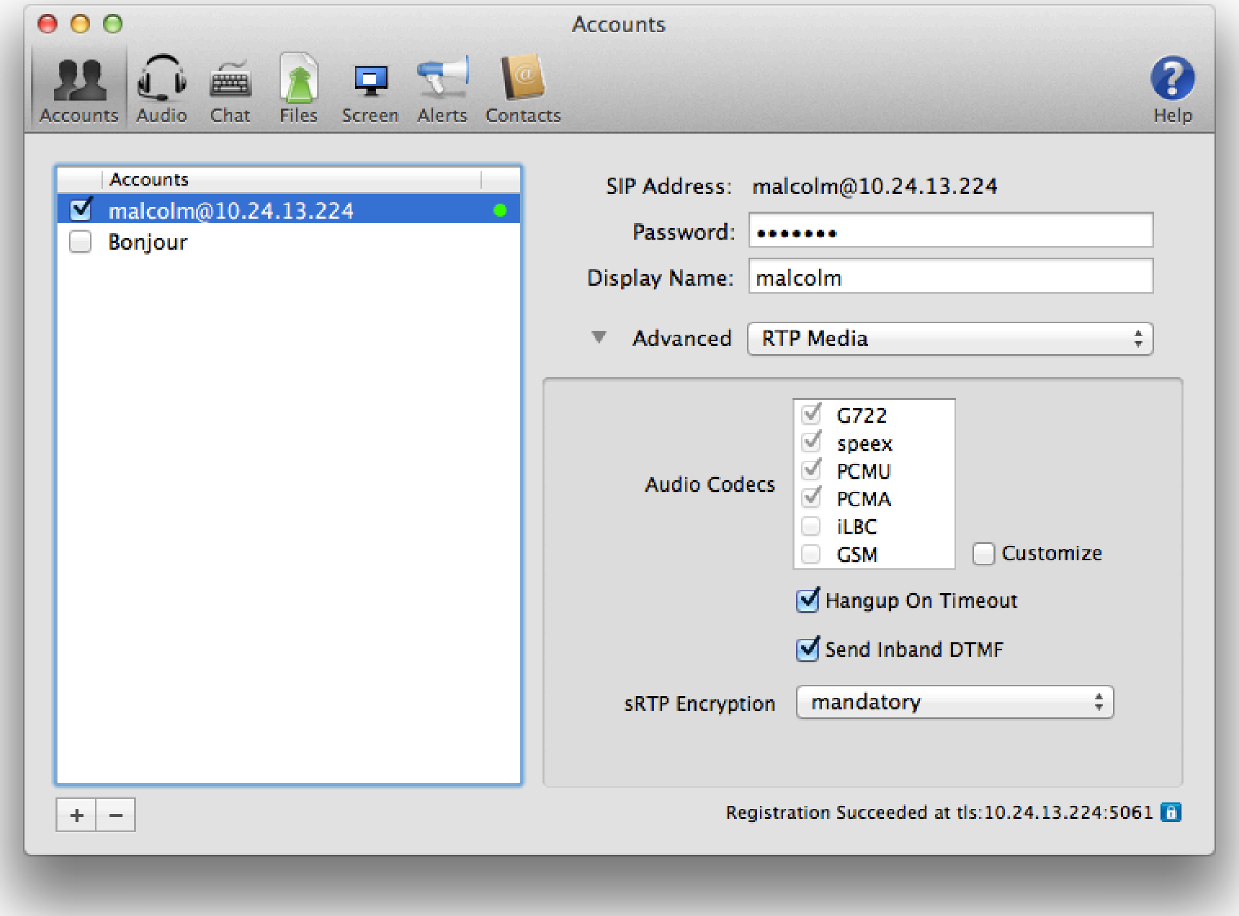
```
[Jan 24 09:29:16] ERROR[10167]: chan_sip.c:27987 setup_srtp: No SRTP module loaded, can't setup SRTP session.
```

on your Asterisk CLI. If you do see that, install libsrtp (and the development headers), and then reinstall Asterisk (./configure; make; make install).

With that complete, let's first go back into our peer definition in **sip.conf.** We're going to add a new encryption line, like:

```
[malcolm]
type=peer
secret=malcolm ;note that this is NOT a secure password
host=dynamic
context=local
dtmfmode=rfc2833
disallow=all
allow=g722
transport=tls
encryption=yes
context=local
```

Next, we'll set Blink to use SRTP:



☑ malcolm@10.24.13.224

□ Bonjour

SIP Address: malcolm@10.24.13.224

Password: •••••••

Display Name: malcolm

Advanced: RTP Media

Audio Codecs:
- ✓ G722
- ✓ speex
- ✓ PCMU
- ✓ PCMA
- □ iLBC
- □ GSM

□ Customize

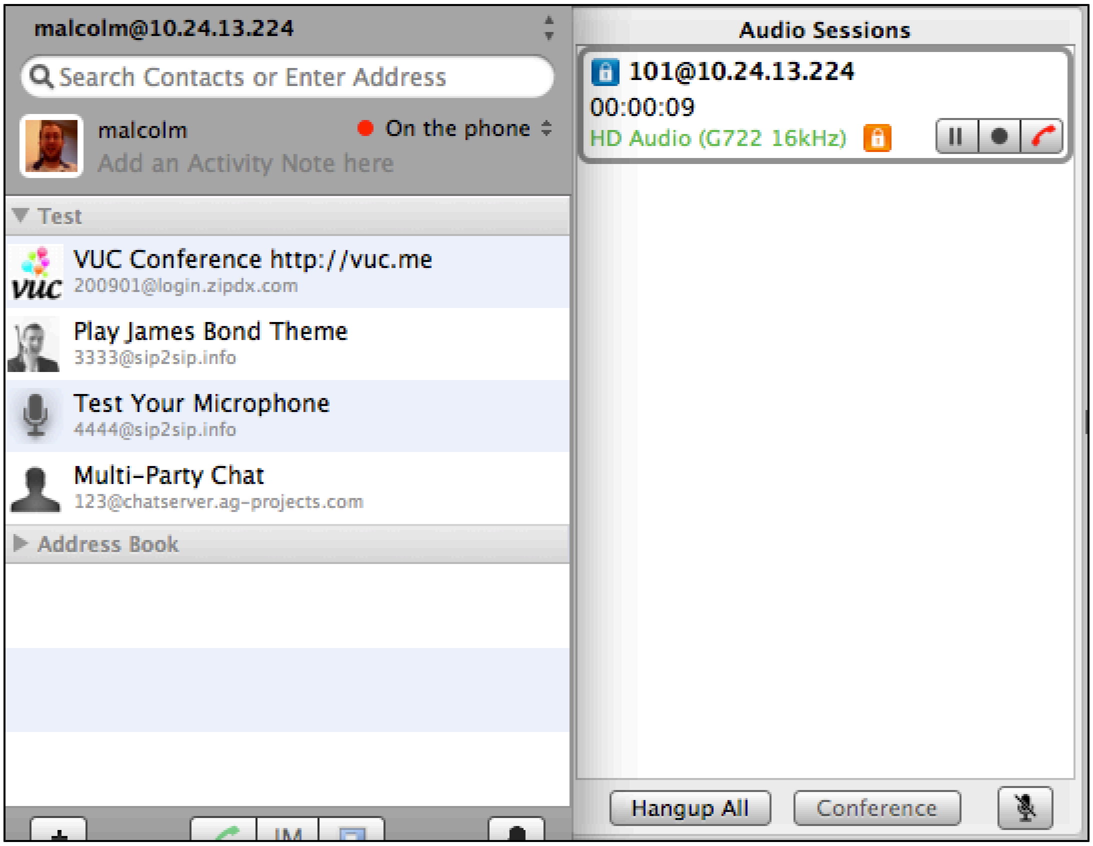
☑ Hangup On Timeout

☑ Send Inband DTMF

sRTP Encryption: mandatory

Registration Succeeded at tls:10.24.13.224:5061

Reload Asterisk's SIP configuration (sip reload), make a call, and voilà:



We're making secure calls with TLS (signaling) and SRTP (media).

Installing Blink SIP client

Overview

I wanted to provide some brief instructions on installing the Blink SIP client on Linux since it is useful for running the Secure Calling Tutorial.

How to install Blink on Ubuntu 12.04 Precise Pangolin

Install some dependencies

1. *Get latest Python 2.X
2. Setup repo from here: http://projects.ag-projects.com/projects/documentation/wiki/Repositories
3. In your terminal, run the following commands

```
sudo apt-get update
sudo apt-get install python-sipsimple python-gnutls python-eventlib python-application python-cjson python-eventlet
python-qt4 python-twisted-core python-zope.interface
```

If you have issues see: http://sipsimpleclient.org/projects/sipsimpleclient/wiki/SipInstallation

Download and install Blink

1. Download Blink source code from http://download.ag-projects.com/BlinkQt
2. In your terminal, extract the tar.gz file contents to a folder and then **cd** to that folder in the terminal
3. Change directory into the Blink folder and run **sudo python setup.py install**

```
root@newtonr-laptop:/usr/src/blink-0.6.0# sudo python setup.py install
running install
running build
running build_py
running build_scripts
running install_lib
running install_scripts
changing mode of /usr/local/bin/blink to 755
running install_data
running install_egg_info
Removing /usr/local/lib/python2.7/dist-packages/blink-0.6.0.egg-info
Writing /usr/local/lib/python2.7/dist-packages/blink-0.6.0.egg-info
```

Run Blink!

1. From the command line, run Blink by executing blink.

```
root@newtonr-laptop:/usr/src/blink-0.6.0# blink
using set_wakeup_fd
"sni-qt/6493" WARN  08:11:15.444 void StatusNotifierItemFactory::connectToSnw() Invalid interface to SNW_SERVICE
```

2. Blink should launch and show up within your graphical desktop.

⚠ Blink doesn't appear to support call forwarding or call transfers, so don't expect to do anything too fancy!

SIP TLS Transport

Asterisk SIP/TLS Transport

When using TLS the client will typically check the validity of the certificate chain. So that means you either need a certificate that is signed by one of the larger CAs, or if you use a self signed certificate you must install a copy of your CA certificate on the client.

So far this code has been tested with:

- Asterisk as client and server (TLS and TCP)
- Polycom Soundpoint IP Phones (TLS and TCP) - Polycom phones require that the host (ip or hostname) that is configured match the 'common name' in the certificate
- Minisip Softphone (TLS and TCP)
- Cisco IOS Gateways (TCP only)
- SNOM 360 (TLS only)
- Zoiper Biz Softphone (TLS and TCP)

sip.conf options

- `tlsenable=yes` - Enable TLS server, default is `no`
- `tlsbindaddr=<ip address>` - Specify IP address to bind TLS server to, default is `0.0.0.0`
- `tlscertfile=</path/to/certificate>` - The server's certificate file. Should include the key and certificate. This is mandatory if you're going to run a TLS server.
- `tlscafile=</path/to/certificate>` - If the server you're connecting to uses a self signed certificate you should have their certificate installed here so the code can verify the authenticity of their certificate.
- `tlscapath=</path/to/ca/dir>` - A directory full of CA certificates. The files must be named with the CA subject name hash value. (see `man SSL_CTX_load_verify_locations` for more info)
- `tlsdontverifyserver=yes` - If set to `yes`, don't verify the servers certificate when acting as a client. If you don't have the server's CA certificate you can set this and it will connect without requiring `tlscafile` to be set. Default is `no`.
- `tlscipher=<SSL cipher string>` - A string specifying which SSL ciphers to use or not use. A list of valid SSL cipher strings can be found at http://www.openssl.org/docs/apps/ciphers.html#CIPHER_STRINGS

Sample config

Here are the relevant bits of config for setting up TLS between 2 Asterisk servers. With server_a registering to server_b

On server_a:

```
[general]
tlsenable=yes
tlscertfile=/etc/asterisk/asterisk.pem
tlscafile=/etc/ssl/ca.pem  ; This is the CA file used to generate both certificates
register => tls://100:test@192.168.0.100:5061

[101]
type=friend
context=internal
host=192.168.0.100 ; The host should be either IP or hostname and should
                   ; match the 'common name' field in the servers certificate
secret=test
dtmfmode=rfc2833
disallow=all
allow=ulaw
transport=tls
port=5061
```

On server_b:

```
[general]
tlsenable=yes
tlscertfile=/etc/asterisk/asterisk.pem

[100]
type=friend
context=internal
host=dynamic
secret=test
dtmfmode=rfc2833
disallow=all
allow=ulaw
;You can specify transport= and port=5061 for TLS, but its not necessary in
;the server configuration, any type of SIP transport will work
;transport=tls
;port=5061
```

Reference Use Cases for Asterisk

This section contains fictitious reference customers and users to serve as a platform for use cases to base all future documentation, tutorials and example files on.

Super Awesome Company

Overview

This page describes the usage of Asterisk within Super Awesome Company (SAC), a fictitious entity for the purpose of example only. All companies, entities and peoples on this page should be assumed fictitious.

Original creator and humourist - the notorious Malcolm Davenport.

This document will be one of a few Reference Use Cases for Asterisk and will be used as a basis for examples and tutorials on this wiki and included with Asterisk.

Background

SAC, founded in 2015 by the noted daredevil Lindsey Freddie, the first person to BASE jump the Burj Khalifa with a paper napkin in place of a parachute, designs and delivers strategic software solutions for the multi-level marketing industry. Humbly beginning in Freddie's garage as a subsidiary of her paper delivery route, SAC has recently moved into modern offices, complete with running water, garbage collection, and access to the Internet and the regular telephone network - a substantial upgrade from the garage-based two cans and a string, water well and composting landfill.

On this Page

Phase 1, "We're a Business"

Upon the advice of the IT staff, SAC has decided to deploy a communications system using Asterisk. For desktop endpoints, SAC has decided to deploy Digium's IP telephones.

SAC requires:

- desk phones for each of its employees
- the ability for each employee to receive direct calls from their loved ones at home as well as kind-hearted telemarketers
- the ability for each employee to place calls to their loved ones at home, their bookies, and also potential customers
- the ability for employees to call one another inside of the office
- a main number that can be dialed by the public and that is answered by a friendly and sarcastic voice that can direct callers about what might be best for them
- a queue for inbound customers inquiring about many of the fine SAC software offerings
- a queue for inbound customers wondering why many of the fine SAC software offerings fail to live up to what customers refer to as "promises"
- an operator to handle callers that spend too much time waiting in queues
- voicemail boxes for each of the employees
- a conference bridge for use by customers and employees
- a conference bridge for use by employees only

Company Organization

SAC employees 15 people with the following corporate structure:

- Lindsey Freddie (lindsey@example.com), President for Life
    - Temple Morgan (temple@ example.com), Life Assistant to the President for Life
    - Terry Jules (terry@ example.com), Director of Sales
        - Garnet Claude (garnet@ example.com), Sales Associate
        - Franny Ocean (franny@ example.com), Sales Associate
    - Maria Berny (maria@ example.com), Director of Customer Experience
        - Dusty Williams (dusty@ example.com), Customer Advocate
        - Tommie Briar (tommie@ example.com), Customer Advocate
    - Penelope Bronte (penelope@ example.com), Director of Engineering

    - Hollis Justy (hollis@ example.com), Software Engineer
    - Richard Casey (richard@ example.com), Software Engineer
    - Sal Smith (sal@ example.com), Software Engineer
    - Laverne Roberts (laverne@ example.com), Software Engineer
    - Colby Hildred (colby@ example.com), IT Systems
- Aaron Courtney (aaron@ example.com), Accounting and Records

By a stroke of luck, Freddie was also the original registrant of the example.com domain name, which, having been lost in years past, she recently managed to win back from ICANN in a game of high-stakes chubby bunny.

Outside Connectivity and Networking

SAC, located in the ghost town of Waldo, Alabama, is fortunate to have access to any type of telecommunications. In spite of Waldo's sole telecom company, WaldoCom, having turned off the lights and locked the doors on their way out, gigabit Internet connectivity over Metro Ethernet is available and operational. It is rumored that WaldoCom left behind a skeleton crew to maintain equipment on an ongoing basis.

WaldoCom, as a traditional communications provider will allow SAC to purchase telephone service. But, because the calling plans across the WaldoCom telephone network must be paid in Confederate dollars, SAC has instead decided to contract with Digium, Inc. for voice services. Digium provides inbound and outbound calling over the Internet as a "SIP Trunk" using SAC's existing Internet connectivity.

⚠ WaldoCom has ceased trading in Bitcoin after having been accused by the FBI of running the Silk Road as the Dread Pirate Roberts.

SAC has purchased a well-loved Linksys WRT54G, aka "Old Unreliable," from the now defunct Waldo Happy Hands Club. They intend to use it to terminate the Ethernet connectivity from WaldoCom. WaldoCom provides a single IPv4 address across the link - 203.0.113.1. The Linksys will provide NAT translation from the Internet to the internal SAC campus network. Within the SAC network, the 10.0.0.0/8 address space will be used.

DIDs / Telephone Numbers

Digium has provided SAC with a block of 300 DIDs (Telephone numbers) beginning with 256-555-1100 and ending with 256-555-1299.

For inbound calls, each SAC employee has a direct DID that rings them.

For outbound calls, each SAC employee's 10-digit DID number is provided as their Caller ID.

Internal Calls

As a shortcut to dialing 10 digits, SAC uses 4-digit dialing for calls between internal employees. The 4 digits assigned to each employee match that employees last 4 digits of their inbound DID. Each employee has been individually assigned a number that can be dialed by any other employee.

SAC has other direct-dial numbers assigned to Queues and menus.

SAC also has several internal extensions that do not map directly to DIDs.

Calls to an Extension

Calls made directly to an extension should ring for 30 sec. After 30 seconds, callers should be directed to the voicemail box of the employee. If the employee was on the phone at the time, callers should hear the voicemail busy greeting; otherwise, callers should hear the voicemail unavailable greeting.

Voicemail

When Asterisk records a voicemail for an SAC employee, it should forward a copy of that voicemail to the employee's e-mail.

Main IVR

SAC has a main IVR. Potential multi-level marketing brand ambassadors or disgruntled bottom-tier customers who find SAC in the WaldoCom Yellow Pages are greeted with a friendly voice and the following helpful message:

"Thank you for calling Super Awesome Company, Waldo's premier provider of perfect products. If you know your party's extension, you may dial it at any time. To establish a sales partnership, press one. To speak with a customer advocate, press two. For accounting and other receivables, press three. For a company directory, press four. For an operator, press zero."

If no DTMF entry is detected, the main IVR repeats twice and then hangs up on the caller.

Employee Number Plan

Temple Morgan, Esquire, Life Assistant to the President for Life, Lindsey Freddie, has randomly assigned the following phone numbers for each of the employees:

Internal Extensions

- Maria Berny - 256-555-1101
- Tommie Briar - 256-555-1102
- Penelope Bronte - 256-555-1103
- Richard Casey - 256-555-1104
- Garnet Claude - 256-555-1105
- Aaron Courtney - 256-555-1106
- Lindsey Freddie - 256-555-1107
- Colby Hildred - 256-555-1108
- Terry Jules - 256-555-1109
- Hollis Justy - 256-555-1110
- Temple Morgan - 256-555-1111
- Franny Ocean - 256-555-1112
- Laverne Roberts - 256-555-1113
- Sal Smith - 256-555-1114
- Dusty Williams - 256-555-1115

Other Numbers

- SAC provides an extension, 8000, that lets internal employees directly dial their voicemail box. When internal SAC employees dial their voicemail box, they’re not prompted for their mailbox number or their PIN code.
- Voicemail may be accessed remotely by employees who dial 256-555-1234. When employees dial voicemail remotely, they must input both their mailbox number and their pin code.
- The Main IVR may be reached externally by dialing 256-555-1100, or internally by dialing 1100.
- The Sales Queue may be reached externally by dialing 256-555-1200, or internally by dialing 1200.
- The Customer Experience Queue may be reached externally by dialing 256-555-1250, or internally by dialing 1250.
- An Operator may be reached from any IVR menu or calling queue by dialing 0.
- The company has a party-line conference room, for use by any employee, on extension 6000.
- The company has a conference room that can be used by customers and employees on extension 6500.

Sales Queue

Calls to the Sales Queue should ring Terry, Garnet and Franny in ring-all fashion

If no one answers a call to the Sales Queue after 5 minutes, the caller should be directed to the Operator so that the Operator can take a message and have a Sales person contact the caller at a later time.

Customer Advocate Queue

Calls to the Customer Advocate Queue should ring Maria, Dusty and Tommie in ring-all fashion.

If no one answers a call to the Customer Advocate queue after 20 minutes, the caller should be directed to the Operator so that the Operator can take a message and have a Customer Advocate contact the caller at a later time.

Operator

Temple Morgan serves as the Operator for SAC.

Deskphones

Each SAC employee is provided with a Digium D70 model telephone.

Outbound calls

Because WaldoCom never upgraded their switches beyond 5-digit dialing, which now covers the entire metropolis of Waldo, SAC employees are not used to dialing 10-digit numbers. Since most calls to potential customers will be made inside the local Waldo area, SAC has managed, through extensive training, to convince their employees to dial 7-digits for local Waldo numbers dialed over their Digium trunk. Because Freddie believes there may be a market for the Broomshakalaka and other products outside of Waldo, perhaps even as far away as Montgomery or Mobile, she has also insisted that 10-digit dialing and 1+10-digit dialing be possible.

Performance Tuning

These are some areas to consider when trying to performance tune your Asterisk installation.

Threadpools

There are two threadpools of interest: pjsip and stasis.

> ⓘ Any changes to threadpool settings require a full Asterisk restart. A reload is insufficient.

PJSIP Threadpool:

The `system` section of pjsip.conf (or the ps_systems table in the database) contains 2 settings that control the threadpool used for the stack:

```
[system]
type=system
;
;  <other settings>
;

; Sets the threadpool size at startup.
; Setting this higher can help Asterisk get through high startup loads
; such as when large numbers of phones are attempting to re-register or
; re-subscribe.
threadpool_initial_size=20

; When more threads are needed, how many should be created?
; Adding 5 at a time is probably safe.
threadpool_auto_increment=5

; Destroy idle threads after this timeout.
; Idle threads do have a memory overhead but it's slight as is the overhead of starting a
new thread.
; However, starting and stopping threads frequently can cause memory fragmentation.  If
the call volume
; is fairly consistent, this parameter is less important since threads will tend to get
continuous
; activity.  In "spikey" situations, setting the timeout higher will decrease the
probability
; of fragmentation.  Don't obsess over this setting.  Setting it to 2 minutes is probably
safe
; for all PBX usage patterns.
threadpool_idle_timeout=120

; Set the maximum size of the pool.
; This is the most important settings.  Setting it too low will slow the transaction rate
possibly
; causing timeouts on clients.  Setting it too high will use more memory, increase the
chances of
; deadlocks and possibly cause other resources such as CPU and I/O to become exhausted.
; For a busy 8 core PBX, 100 is probably safe.  Setting this to 0 will allow the pool to
grow
; as high as the system will allow.  This is probably not what you want. :)  Setting it
to 500
; is also probably not what you want.  With that many threads, Asterisk will be thrashing
and
; attempting to use more memory than can be allocated to a 32-bit process.  If memory
starts
; increasing, lowering this value might actually help.
threadpool_max_size=100
```

Stasis Threadpool

Although the stasis message bus is not used much for simple call processing, it *is* used heavily for ARI and AGI processing, transfers, conference bridges, AMI, CDR and CEL processing, etc. The threadpool is configured in stasis.conf:

```
[threadpool]
;
; For a busy 8 core PBX, these settings are probably safe.
;
initial_size = 10
idle_timeout_sec = 120
;
; The notes about the pjsip max size apply here as well.  Increasing to 100 threads is
probably
; safe, but anything more will probably cause the same thrashing and memory
over-utilization,
max_size = 60
```

If you don't need AMI, CDR, or CEL then disabling those modules will reduce resource usage. The CDR module uses a lot of processing to create the CDR records and can easily get backed up on a busy system.

PJSIP Protocol Tuning

Timers

The `timer_t1` and `timer_b` settings are in the `system` section of pjsip.conf (or the ps_systems table in the database)

```
[system]
type=system
; Timer t1 sets the timeout after which pjsip gives up on waiting for a response from
; the remote party.  The general rule is to set this to slightly higher than the
round-trip
; time to the furthest remote party.  Although the default of 500ms is safe, this timer
; controls other timing aspects of the of the stack so reducing it is in your best
interest.
; Unless you have a provider or remote phones with more than a 100ms RTT, setting this to
; 100ms (the minimum) is probably safe.  If you have outlier phones such as cell phones
; with VoIP clients, setting it to 250ms is probably safe.
timer_t1=100

; Timer B is technically the INVITE transaction timeout but it also controls other
aspects
; of stack timing.  It's default is 32 seconds but its minimum is (64 * timer_t1) which
; would also be 32 seconds if timer_t1 were left at its default of 500ms.  Unfortunately,
; this timer has the side effect of controlling how long completed transactions are kept
in
; memory so on a busy PBX, a setting of 32 seconds will probably result in higher than
; necessary memory utilization.  For most installations, 6400ms is fine.
timer_b=6400
```

Identification Priority

The order in which endpoint identification methods are tried when an incoming request is received directly affects transaction rate. The default order is set in the `global` of pjsip.conf (or the ps_globals table in the database).

```
[global]
type=global
; The default identifier order is ip,username,anonymous but for a PBX environment
; with lots of phones that register, identifying by ip address first is a waste of time.
; The order should be from the most likely to be used, to the least likely to be used
; which in this case would put username first for the phones, and ip second for
providers.
endpoint_identifier_order=username,ip,anonymous
```

Sorcery/Database

While storing pjsip objects in the pjsip.conf results in the fastest access time during call processing, a config change requires the entire file to be re-written and the res_pjsip module to be reloaded. Using backend database for storage is most convenient for configuration but will be slowest for access time during call processing. The solution is to use the database for storage and use sorcery to cache the objects. This will result in the same access times as using pjsip.conf.

Setting up caching

The sorcery caches are defined in sorcery.conf.

```
[res_pjsip]

; maximum_objects:  How many object to allow in the cache at 1 time.
; expire_on_reload:  If res_pjsip is reloaded, should the cache be flushed?
; object_lifetime_maximum; How long should an object remain in the cache before it's
flushed.

; There is only ever 1 row in the ps_globals table but it's referenced heavily and rarely
; changes.  You may choose to leave this in pjsip.conf and comment out these 2 lines.
; On recent versions of Asterisk, the global section is only read on a pjsip reload which
; effectively caches the settings without an expiration time.
;global/cache=memory_cache,maximum_objects=2,expire_on_reload=yes,object_lifetime_maximum
=3600
;global=realtime,ps_globals

; There is only ever 1 row in the ps_systems table and it's not referenced after startup.
; You may chose to leave this in pjsip.conf and comment out these 2 lines.
system/cache=memory_cache,maximum_objects=2,expire_on_reload=yes,object_lifetime_maximum=
3600
system=realtime,ps_systems

; endpoints, aors, and auths are heavily read objects but are only written to when their
; configuration is changed.  Set the maximum_objects to the number of extensions, plus
the
; number of peered PBXes, plus the number of hosts defined for all providers (a provider
; with 10 hosts defined will use 10 endpoints, 10 aors and 1 auth).  Add a few to spare.
; When a configuration change is made to an object, the specific object is flushed from
the
; cache so the object_lifetime_maximum of 15 minutes is just a fail-safe.
endpoint/cache=memory_cache,maximum_objects=3000,expire_on_reload=yes,object_lifetime_max
imum=900
endpoint=realtime,ps_endpoints

aor/cache=memory_cache,maximum_objects=3000,expire_on_reload=yes,object_lifetime_maximum=
900
aor=realtime,ps_aors
```

```
auth/cache=memory_cache,maximum_objects=3000,expire_on_reload=yes,object_lifetime_maximum=900
auth=realtime,ps_auths

; contacts are both read from and written to regularly by Asterisk.
contact/cache=memory_cache,maximum_objects=3000,expire_on_reload=yes,object_lifetime_maximum=900
contact=realtime,ps_contacts


[res_pjsip_endpoint_identifier_ip]
; There will be 1 ip identifier for each host across all providers plus 1 for each peered PBX.
identify/cache=memory_cache,maximum_objects=150,expire_on_reload=yes,object_lifetime_maximum=900
identify=realtime,ps_endpoint_id_ips


;[res_pjsip_outbound_registration]
; There could be 1 outbound registration for each host across all providers depending on whether
; the provider requires them.
registration/cache=memory_cache,maximum_objects=150,expire_on_reload=yes,object_lifetime_maximum=900
```

```
registration=realtime,ps_registrations
```

Flushing the caches:

The `sorcery memory cache` Asterisk CLI commands will allow flushing caches and individual objects from a specific cache. There are also equivalent AMI commands (SorcerymemoryCache*) that do the same. After you make all pjsip configuration changes, call the appropriate AMI commands to flush objects and caches where appropriate. This is necessary for Asterisk to see the changes made in the database immediately.

Phone Provisioning in Asterisk

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.

Configuration of phoneprov.conf

The configuration file, phoneprov.conf, is used to set up the built-in variables SERVER and SERVER_PORT, to define a default phone profile to use, and to define different phone profiles available for provisioning.

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.

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 mime-type. 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.

⚠ 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.

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.

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.

Individual 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
directmedia = 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.

Phone Provisioning 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).

Phone Provisioning, 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 Asterisk. 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

Asterisk Security Framework

Attacks on Voice over IP networks are becoming increasingly more common. It has become clear that we must do something within Asterisk to help mitigate these attacks.

Through a number of discussions with groups of developers in the Asterisk community, the general consensus is that the best thing that we can do within Asterisk is to build a framework which recognizes and reports events that could potentially have security implications. Each channel driver has a different concept of what is an "event", and then each administrator has different thresholds of what is a "bad" event and what is a restorative event. The process of acting upon this information is left to an external program to correlate and then take action - block traffic, modify dialing rules, etc. It was decided that embedding actions inside of Asterisk was inappropriate, as the complexity of construction of such rule sets is difficult and there was no agreement on where rules should be enabled or how they should be processed. The addition of a major section of code to handle rule expiration and severity interpretation was significant. As a final determining factor, there are external programs and services which already parse log files and act in concert with packet filters or external devices to protect or alter network security models for IP connected hosts.

Security Framework Overview

This section discusses the architecture of the Asterisk modifications being proposed.
There are two main components that we propose for the initial implementation of the security framework:

- Security Event Generation
- Security Event Logger

Security Event Generation

The ast_event API is used for the generation of security events. That way, the events are in an easily interpretable format within Asterisk to make it easy to write modules that do things with them. There are also some helper data structures and functions to aid Asterisk modules in reporting these security events with the proper contents.

The next section of this document contains the current list of security events being proposed. Each security event type has some required pieces of information and some other optional pieces of information.

Subscribing to security events from within Asterisk can be done by subscribing to events of type AST_EVENT_SECURITY. These events have an information element, AST_EVENT_IE_SECURITY_EVENT, which identifies the security event sub-type (from the list described in the next section). The result of the information elements in the events contain the required and optional meta data associated with the event sub-type.

Asterisk Security Event Logger

In addition to the infrastructure for generating the security events, an event logger module (that can consume these events) is also available. Asterisk, through its Logging Configuration supports multiple types of dynamic logging levels. The security logging module takes advantage of this and creates a custom "security" logging level when loaded. To enable logging of security events simply add a file, specifying the "security" logging level, to the logger.conf. For example adding the following will log security events to a file named "security_log":

```
security_log => security
```

The content of the log file is well defined and is in an easily interpretable format allowing for external scripts to read and act upon.

Security Events to Log

```
(-) required
(+) optional

Invalid Account ID
  (-) Local address family/IP address/port/transport
  (-) Remote address family/IP address/port/transport
  (-) Service (SIP, AMI, IAX2, ...)
  (-) System Name
  (+) Module
  (+) Account ID (username, etc)
  (+) Session ID (CallID, etc)
  (+) Session timestamp (required if Session ID present)
  (-) Event timestamp (sub-second precision)
Failed ACL match
  -> everything from invalid account ID
  (+) Name of ACL (when we have named ACLs)

Invalid Challenge/Response
  -> everything from invalid account ID
  (-) Challenge
  (-) Response
  (-) Expected Response
Invalid Password
  -> everything from invalid account ID

Successful Authentication
  -> informational event
  -> everything from invalid account ID

Invalid formatting of Request
  -> everything from invalid account ID
  -> account ID optional
  (-) Request Type
  (+) Request parameters
Session Limit Reached (such as a call limit)
  -> everything from invalid account ID

Memory Limit Reached
  -> everything from invalid account ID
Maximum Load Average Reached
  -> everything from invalid account ID

Request Not Allowed
  -> everything from invalid account ID
  (-) Request Type
  (+) Request parameters
Request Not Supported
  -> everything from invalid account ID
  (-) Request Type

Authentication Method Not Allowed
  -> everything from invalid account ID
  (-) Authentication Method attempted
In dialog message from unexpected host
  -> everything from invalid account ID
  (-) expected host
```

Security Log File Format

The beginning of each line in the log file is the same as it is for other logger levels within Asterisk.

```
[Feb 11 07:57:03] SECURITY[23736] res_security_log.c: <...>
```

The part of the log entry identified by \<...\> is where the security event content resides. The security event content is a comma separated list of key value pairs. The key is the information element type, and the value is a quoted string that contains the associated meta data for that information element. Any embedded quotes within the content are escaped with a backslash.

INFORMATION_ELEMENT_1="IE1 content",INFORMATION_ELEMENT_2="IE2 content"

The following table includes potential information elements and what the associated content looks like:

- IE: SecurityEvent
  Content: This is the security event sub-type.
  Values: FailedACL, InvalidAccountID, SessionLimit, MemoryLimit, LoadAverageLimit, RequestNotSupported, RequestNotAllowed, AuthMethodNotAllowed, ReqBadFormat, UnexpectedAddress, ChallengeSent, ChallengeResponseFailed, InvalidPassword

- IE: Severity
  Content: This is the relatively severity of the security event.
  Values: Informational, Error

- IE: EventVersion
  Content: This is a numeric value that indicates when updates are made to the content of the event.
  Values: Monotonically increasing integer, starting at 1

- IE: Service
  Content: This is the Asterisk service that generated the event.
  Values: TEST, SIP, PJSIP, AMI

- IE: Module
  Content: This is the Asterisk module that generated the event.
  Values: chan_sip

- IE: AccountID
  Content: This is a string used to identify the account associated with the event. In most cases, this would be a username.

- IE: SessionID
  Content: This is a string used to identify the session associated with the event. The format of the session identifier is specific to the service. In the case of SIP, this would be the Call-ID.

- IE: SessionTV
  Content: The time that the session associated with the SessionID started.
  Values: <seconds><microseconds> since epoch

- IE: ACLName
  Content: This is a string that identifies which named ACL is associated with this event.

- IE: LocalAddress
  Content: This is the local address that was contacted for the related event.
  Values: <Address Family>/<Transport>/<Address>/<Port>
  Examples: -> IPV4/UDP/192.168.1.1/5060 -> IPV4/TCP/192.168.1.1/5038

- IE: RemoteAddress
  Content: This is the remote address associated with the event.
  Examples: -> IPV4/UDP/192.168.1.2/5060 -> IPV4/TCP/192.168.1.2/5038

- IE: ExpectedAddress
  Content: This is the address that was expected to be the remote address.
  Examples: -> IPV4/UDP/192.168.1.2/5060 -> IPV4/TCP/192.168.1.2/5038

- IE: EventTV
  Content: This is the timestamp of when the event occurred.
  Values: <seconds><microseconds> since epoch

- IE: RequestType
  Content: This is a service specific string that represents the invalid request

- IE: RequestParams
  Content: This is a service specific string that represents relevant parameters given with a request that was considered invalid.

- IE: AuthMethod
  Content: This is a service specific string that represents an authentication method that was used or requested.

- IE: Challenge
  Content: This is a service specific string that represents the challenge provided to a user attempting challenge/response authentication.

- IE: Response
  Content: This is a service specific string that represents the response received from a user attempting challenge/response authentication.

- IE: ExpectedResponse
  Content: This is a service specific string that represents the response that was expected to be received from a user attempting challenge/response authentication.

Distributed Universal Number Discovery (DUNDi)

Top-level page for all things DUNDi

Introduction to DUNDi

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

DUNDi Dialplan Functions

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
```

Digium General Peering Agreement

```
DIGIUM GENERAL PEERING AGREEMENT (TM)
                      Version 1.0.0, September 2004
 Copyright (C) 2004 Digium, Inc.
                445 Jan Davis Drive, Huntsville, AL 35806 USA

 Everyone is permitted to copy and distribute complete verbatim copies
 of this General Peering Agreement provided it is not modified in any
 manner.

        ------------------------------------------------------

                    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.
```

Packet Loss Concealment (PLC)

What is PLC?

PLC stands for Packet Loss Concealment. PLC describes any method of generating new audio data when packet loss is detected. In Asterisk, there are two main flavors of PLC, generic and native. Generic PLC is a method of generating audio data on signed linear audio streams. Signed linear audio, often abbreviated "slin," is required since it is a raw format that has no companding, compression, or other transformations applied. Native PLC is used by specific codec implementations, such as iLBC and Speex, which generates the new audio in the codec's native format. Native PLC happens automatically when using a codec that supports native PLC. Generic PLC requires specific configuration options to be used and will be the focus of this document.

How does Asterisk detect packet loss?

Oddly, Asterisk does not detect packet loss when reading audio in. In order to detect packet loss, one must have a jitter buffer in use on the channel on which Asterisk is going to write missing audio using PLC. When a jitter buffer is in use, audio that is to be written to the channel is fed into the jitterbuffer. When the time comes to write audio to the channel, a bridge will request that the jitter buffer gives a frame of audio to the bridge so that the audio may be written. If audio is requested from the jitter buffer but the jitter buffer is unable to give enough audio to the bridge, then the jitter buffer will return an interpolation frame. This frame contains no actual audio data and indicates the number of samples of audio that should be inserted into the frame.

PLC Background on Translation

As stated in the introduction, generic PLC can only be used on slin audio. The majority of audio communication is not done in slin, but rather using lower bandwidth codecs. This means that for PLC to be used, there must be a translation step involving slin on the write path of a channel. One item of note is that slin must be present on the write path of a channel since that is the path where PLC is applied. Consider that Asterisk is bridging channels A and B. A uses ulaw for audio and B uses GSM. This translation involves slin, so things are shaping up well for PLC. Consider, however if Asterisk sets up the translation paths like so:

Fig. 1

```
A                      +------------+       B
<---ulaw<---slin<---GSM|            |GSM--->
                       |  Asterisk  |
ulaw--->slin--->GSM--->|            |<---GSM
                       +------------+
```

The arrows indicate the direction of audio flow. Each channel has a write path (the top arrow) and a read path (the bottom arrow). In this setup, PLC can be used when sending audio to A, but it cannot be used when sending audio to B. The reason is simple, the write path to A's channel contains a slin step, but the write path to B contains no slin step. Such a translation setup is perfectly valid, and Asterisk can potentially set up such a path depending on circumstances. When we use PLC, however, we want slin audio to be present on the write paths of both A and B. A visual representation of what we want is the following:

Fig. 2

```
A               +------------+               B
<---ulaw<---slin|            |slin--->GSM--->
                |  Asterisk  |
ulaw--->slin--->|            |<---slin<---GSM
                +------------+
```

In this scenario, the write paths for both A and B begin with slin, and so PLC may be applied to either channel. This translation behavior has, in the past been doable with the transcode_via_sln option in asterisk.conf. Recent changes to the PLC code have also made the `genericplc` option in codecs.conf imply the `transcode_via_sln` option. The result is that by enabling `genericplc` in codecs.conf, the translation path set up in Fig. 2 should automatically be used as long as the two codecs required transcoding in the first place.

If the codecs on the inbound and outbound channels are the same or do not require transcoding, PLC won't normally be used for the reasons stated above. You can however, force transcoding and PLC in this situation by setting the `genericplc_on_equal_codecs parameter` in the `plc` section of codecs.conf to true. This feature was introduced in Asterisk 13.21 and 15.4.

PLC Restrictions and Caveats

One restriction that has not been spelled out so far but that has been hinted at is the presence of a bridge. The term bridge in this sense means two channels exchanging audio with one another. A bridge is required because use of a jitter buffer is a prerequisite for using PLC, and a jitter buffer is only used when bridging two channels. This means that one-legged calls, (e.g. calls to voicemail, to an IVR, to an extension that just plays back audio) will not use PLC. In addition, MeetMe and ConfBridge calls will not use PLC. It should be obvious, but it bears mentioning, that PLC cannot be used when using a technology's native bridging functionality. For instance, if two SIP channels can exchange RTP directly, then Asterisk will never be able to process the audio in the first place. Since translation of audio is a requirement for using PLC, and translation will not allow for a native bridge to be created, this is something that is not likely to be an issue, though. Since a jitter buffer is a requirement in order to use PLC, it should be noted that simply enabling the jitter buffer via the jbenable option may not be enough. For instance, if bridging two SIP channels together, the default behavior will not be to enable jitter buffers on either channel. The rationale is that the jitter will be handled at the endpoints to which Asterisk is sending the audio. In order to ensure that a jitter buffer is used in all cases, one must enable the jbforce option for channel types on which PLC is desired.

Requirements for PLC Use

The following are all required for PLC to be used:

1. Enable genericplc in the plc section of codecs.conf
2. If PLC is desired when two channels use the same codec, enable genericplc_on_equal_codecs in the plc section of codecs.conf. This forces transcoding via slin.
3. Enable (and potentially force) jitter buffers on channels
4. Two channels must be bridged together for PLC to be used (no Meetme or one-legged calls)
5. The audio must be translated between the two channels and must have slin as a step in the translation process.

PLC Tips

One of the restrictions mentioned is that PLC will only be used when two audio channels are bridged together. Through the use of Local channels, you can create a bridge even if the call is, for all intents and purposes, one-legged. By using a combination of the /n and /j suffixes for a Local channel, one can ensure that the Local channel is not optimized out of the talk path and that a jitter buffer is applied to the Local channel as well. Consider the following simple dialplan:

```
[example]
exten => 1,1,Playback(tt-weasels)
exten => 2,1,Dial(Local/1@example/nj)
```

When dialing extension 1, PLC cannot be used because there will be only a single channel involved. When dialing extension 2, however, Asterisk will create a bridge between the incoming channel and the Local channel, thus allowing PLC to be used.

Enhanced Messaging

Conference Participant Messaging

Overview

Since Asterisk 13.22.0 and 15.5.0, in-dialog SIP MESSAGE support in the chan_pjsip channel driver is enhanced and conference bridges added support for relaying messages. The chan_pjsip channel driver now allows exchanging enhanced messages with Asterisk's core that have additional metadata indicating the sender and the mime-type of the message contents. The conference bridges now allow relaying text and enhanced messages from one participant to all other participants.

How it works

It sounds simple enough but this required some restructuring of the bridging core to preserve the original sender's information and add support for text content types other than text/plain.

- The participant creates a SIP MESSAGE request with a specific content type, message body, and optionally a "From" display name.
- The participant then sends that message in-dialog to the conference bridge.
- Normally when a channel driver receives a text message, it passes only the text body to the bridging core, but this causes the sender and content type to be lost. Now, when the chan_pjsip res_pjsip_messaging module receives an in-dialog SIP MESSAGE, it captures the From header's display name, the content type, and the body to pass on to the bridging core. Other than the From display name, no other sender information is exposed.
- When bridge_softmix (the bridging module used by ConfBridge) sees the message, it relays it to all other bridge participants.
- Any other participants connected via chan_pjsip will get the From display name, content-type, and body. Those not connected via chan_pjsip will get whatever the channel driver supports.

Using Enhanced Messaging

While Enhanced Messaging is interesting for Asterisk 13, it is very interesting for Asterisk 15 and later. Imagine a video conference using Asterisk 15's Selective Forwarding Unit (SFU) capability in ConfBridge. Enhanced Messaging allows the conference participants to chat while participating in the conference. CyberMegaPhone is an example for such a video conference application.

Configuring Asterisk

There is no additional configuration needed. Enhanced Messaging is built-in and always available.

In the browser...

How you design the user interface portion is totally up to you but here is a sample of how CyberMegaPhone could be extended to send a message using JsSIP. In this example, this._ua is a JsSIP.UA instance and this.rtc is a JsSIP.RTCSession instance. Refer to the CyberMegaPhone code to see where this might fit.

```
CyberMegaPhone.prototype.sendMessage = function (string_msg, options = {} ) {
    /*
     * You could allow the user to set a nickname
     * for themselves which JsSIP can send as the
     * display name in the SIP From header.  In the code
     * that receives the message, you can then grab the
     * display name from the packet.
     */
    if (options.from) {
        from = options.from;
        this._ua.set("display_name", from);
    }
    /*
     * The message payload can be any UTF-8 string but you are not
     * limited to plain text.  The Content-Type must be set to one
     * of the text/ or application/ types but as long as the sender
     * and receiver agree on the payload format, it can contain
     * whatever you want. In this example, we are going to send
     * a JSON blob.
     *
     * If you do not want to alter the display name on the actual
     * SIP MESSAGE From header, you could include the user's
     * nickname in the payload.
     */
    let msg = {
        'From': from,
        'Body': string_msg
    };
    let body = JSON.stringify(msg);
    let extraHeaders = [ 'Content-Type: application/x-myphone-confbridge-chat+json' ];
    this.rtc.sendRequest(JsSIP.C.MESSAGE, {
        extraHeaders,
        body: body,
        eventHandlers: options.handlers
    });
};
/*
 * Now here is how you would call sendMessage
 */
    phone.sendMessage("Hello!", {from: "My Name", handlers: {
        onSuccessResponse(response) {
            // You may want to show an indicator that the message was sent
successfully.
            console.log("Message Sent: " + response);
        },
        onErrorResponse(response) {
            console.log("Message ERROR: " + response);
        },
        onTransportError() {
            console.log("Could not send message");
        },
        onRequestTimeout() {
            console.log("Timeout sending message");
        },
        onDialogError() {
            console.log("Dialog Error sending message");
        },
    }});
```

Congratulations, you have just sent a text message! Assuming the user called a conference bridge in the first place, all the other participants should receive it. The code to retrieve the message is even simpler than the code to send it. Once again, in this CyberMegaPhone example, this._ua is the JsSIP.UA instance.

```
this._ua.on('newMessage', function (data) {
        /* We do not care about messages we send. */
        if (data.originator === 'local') {
            return;
        }
        /* Grab the Content-Type header from the packet */
        let ct = data.request.headers['Content-Type'][0].raw;
        /* Make sure the message is one we care about */
        if (ct === 'application/x-myphone-confbridge-chat+json') {
            /* Parse the body back into an object */
            let msg = JSON.parse(data.request.body);
            /* Tell the UI that we got a chat message */
            that.raise('messageReceived', msg);
        }
    });
```

Conference Bridge Messaging

Overview

Since Asterisk 13.22.0 and 15.5.0, the ConfBridge application is optionally able to send enhanced messages to participants about the bridge itself and the participants in the bridge.

What data is available?

The AMI events generated by the ConfBridge application give you an idea of what data is available. You can take a look at the ConfBridge manager events pages of your version of Asterisk. However, AMI events are typically sent only to trusted parties so not all of the information in each event is available via Enhanced Messaging. Conversely, there is data available via Enhanced Messaging that is useful for a conference application but not included in the AMI events. One of the most useful of these is a ConfbridgeWelcome event that participants receive when they join a conference. It contains a list of all the other participants currently in the conference.

What could you do with the data?

These are just a few examples.

- For your chat application, show a list of all the current conference participants.
- Overlay participant nicknames on their video windows.
- Add mute/unmute indicators.
- Highlight the video element of the current speaker.

Message examples

When Bob joins the ConfBridge he receives the ConfbridgeWelcome event:

```
{
  "type": "ConfbridgeWelcome",
  "timestamp": "2018-08-17T08:33:30.806-0600",
  "bridge": {
    "id": "75539107-a8fb-47fd-9b6a-7a9391ce011a",
    "name": "MYCONF",
    "video_mode": "sfu"
  },
  "channels": [
    {
      "id": "confserver-1534516369.15",
      "name": "PJSIP/trunk1-00000003",
      "state": "Up",
      "caller": {
        "name": "Alice",
        "number": "alicewonderland"
      },
      "creationtime": "2018-08-17T08:32:49.741-0600",
      "language": "en",
      "admin": true,
      "muted": false
    },
    {
      "id": "confserver-1534516410.21",
      "name": "PJSIP/trunk1-00000000",
      "state": "Up",
      "caller": {
        "name": "Bob",
        "number": "robert"
      },
      "creationtime": "2018-08-17T08:17:36.353-0600",
      "language": "en",
      "admin": true,
      "muted": false
    }
  ]
}
```

In the ConfbridgeWelcome event, the "channels" arrays contains all of the channels currently in the bridge, including Bob himself. In this case, Alice was already in the conference when Bob joined.

The ConfbridgeJoin event is sent to all other participants when someone joins the conference. In this case Bob is joining the conference. If the "echo_events" option is enabled, Bob can also receive his join message when he joins.

```
{
  "type": "ConfbridgeJoin",
  "timestamp": "2018-08-17T08:17:39.578-0600",
  "bridge": {
    "id": "ea65ab81-179a-47eb-b55e-be716a2c7c80",
    "name": "MYCONF",
    "video_mode": "sfu"
  },
  "channels": [
    {
      "id": "confserver-1534516410.21",
      "name": "PJSIP/trunk1-00000000",
      "state": "Up",
      "caller": {
        "name": "Bob",
        "number": "robert"
      },
      "creationtime": "2018-08-17T08:17:36.353-0600",
      "language": "en",
      "admin": true,
      "muted": false
    }
  ]
}
```

Since this is an event related to a specific participant, the "channels" array just has Bob's channel in it.

Most of the other ConfBridge events have the same contents. They are ConfbridgeJoin, ConfbridgeLeave, ConfbridgeRecord, ConfbridgeStopRecord, ConfbridgeMute, ConfbridgeUnmute, ConfbridgeWelcome. If your bridge has the "talk_detection_events" option set to "yes", participants will also receive ConfbridgeTalking events containing a "talking_status" indicator in place of the "muted" parameter appearing in most of the other events.

If you are familiar with the ConfBridge AMI events, you will notice that the ConfbridgeStart and ConfbridgeEnd events are missing. That is because they do not make much sense in this context. At the time they are generated, there are no participants to receive them.

Getting events sent to your web application

In Asterisk

Start by adding a few parameters to your user and bridge profiles in confbridge.conf:

```
[default_user]
type=user
send_events=yes ; If events are enabled for this bridge and this option is
                ; set, users will receive events like join, leave, talking,
                ; etc. via text messages.  For users accessing the bridge
                ; via chan_pjsip, this means in-dialog MESSAGE requests.
                ; This is most useful for WebRTC participants where the
                ; browser application can use the messages to alter the user
                ; interface.
echo_events=yes ; If events are enabled for this user and this option is set,
                ; the user will receive events they trigger, talking, mute, etc.
                ; If not set, they will not receive their own events.
[default_bridge]
type=bridge
enable_events=yes  ; If enabled, recipients who joined the bridge via a channel driver
                   ; that supports Enhanced Messaging (currently only chan_pjsip) will
                   ; receive in-dialog messages containing a JSON body describing the
                   ; event.  The Content-Type header will be
                   ; "application/x-asterisk-confbridge-event+json".
                   ; This feature must also be enabled in user profiles.
```

Of course, your configuration will be different but those are the parameters that need to be set.

> If a user connects to the bridge via a DAHDI channel or some other non-SIP based channel, they may receive messages in another format, like SMS, which is probably not a good idea. To prevent this, you may want to use two different user profiles, one with events enabled and one without. You could then do some simple dialplan logic to look at the incoming channel technology and call ConfBridge() with the appropriate user profile.

In the browser

You use the same mechanism as shown in Conference Participant Messaging. The only difference will be that the message Content-Type will be "application/x-asterisk-confbridge-event+json". The message bodies will be JSON events as shown above.

Putting it all together

One of the possible uses for the events mentioned above was overlaying a participant's nickname on their video window. Here is a hint on how you could do that.

If you look at the events you will notice that every channel specifies an "id" field which is the Asterisk channel's unique id. When you receive ConfbridgeJoin or ConfbridgeWelcome events, save the event in a hashmap keyed by that id. When you receive a new video stream, which you can intercept by adding an "sdp" handler to your JsSip RTCSession, look at the sdp and look for the "a:label" and "a:msid" attributes in each video stream. The "a:label" will be the participant's channel id as specified in the events and the "a:msid" will be available in the video element WebRTC creates. Now, just create another hashmap that cross references the two and when you draw your video elements, you can grab the msid from it and then get the corresponding participant from the hashmaps.

Operation

Top-level page for a section for documentation concerning the operation of the Asterisk program and it's environment. Such as: , How to run Asterisk, System requirements, Maintenance, Logging, CLI usage, etc.

System Requirements

In order to compile and install Asterisk, you'll need to install a C compiler and a number of system libraries on your system.

- Compiler
- System Libraries

Compiler

The compiler is a program that takes source code (the code written in the C programming language in the case of Asterisk) and turns it into a program that can be executed. Currently, Asterisk version 1.8 and later depend on extensions offered by the **GCC** compiler for its RAII_VAR macro implementation, so **GCC** must be used to compile Asterisk. There are currently efforts under way to make Asterisk compatible with Clang's equivalent extensions.

If the **GCC** compiler isn't already installed on your machine, simply use appropriate package management system on your machine to install it. You'll also want to install **GCC**'s C++ compiler (g++) as well since certain Asterisk modules require it.

System Libraries

In addition to the C compiler, you'll also need a set of system libraries. Essential libraries are used by Asterisk and must be installed before you can compile Asterisk. Core libraries allow compilation of additional core supported features. On most operating systems, you'll need to install both the library and it's corresponding development package.

> ✓ Development libraries
>
> For most operating systems, the development packages will have -dev or -devel on the end of the name. For example, on a Red Hat Linux system, you'd want to install both the "openssl" and "openssl-devel" packages.

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Essential Libraries

- libjansson
- libsqlite3
- libxml2
- libxslt
- ncurses
- openssl
- uuid

Core Libraries

- DAHDI
- pjproject
- unixodbc
- libspeex
- libspeexdsp
- libresample
- libcurl3
- libvorbis
- libogg
- libsrtp
- libical
- libiksemel
- libneon
- libgmime
- libunbound

We recommend you use the package management system of your operating system to install these libraries before compiling and installing Asterisk.

> ✓ Help Finding the Right Libraries
>
> Asterisk comes with a shell script called install_prereq.sh in the contrib/scripts sub-directory. If you run install_prereq test, it will give you the exact commands to install the necessary system libraries on your operating system. If you run install_prereq install, it will attempt to download and install the prerequisites automatically.

Running Asterisk

Running Asterisk from the Command Line

- By default, starting Asterisk will run it in the background:

```
# asterisk

# ps aux | grep asterisk
my_user     26246  2.0  4.1 2011992 165520 ?      Ssl  16:35   0:16 asterisk
```

- In order to connect to a running Asterisk process, you can attach a **remote console** using the `-r` option:

```
# asterisk -r

Asterisk 11.9.0, Copyright (C) 1999 - 2014 Digium, Inc. and others.
Created by Mark Spencer <markster@digium.com>
Asterisk comes with ABSOLUTELY NO WARRANTY; type 'core show warranty' for details.
This is free software, with components licensed under the GNU General Public
License version 2 and other licenses; you are welcome to redistribute it under
certain conditions. Type 'core show license' for details.
=========================================================================
Connected to Asterisk 11.9.0 currently running on asterisk-server (pid = 26246)
asterisk-server*CLI>
```

> The `-R` option will also attach a remote console - however, it will attempt to automatically reconnect to Asterisk if for some reason the connection is broken. This is particularly useful if your remote console restarts Asterisk.

On this Page

- To disconnect from a connected remote console, simply hit **Ctrl+C**:

```
asterisk-server*CLI>
Disconnected from Asterisk server
Asterisk cleanly ending (0).
Executing last minute cleanups
```

- To shut down Asterisk, issue `core stop gracefully`:

```
asterisk-server*CLI> core stop gracefully
Disconnected from Asterisk server
Asterisk cleanly ending (0).
Executing last minute cleanups
```

> You can stop/restart Asterisk in many ways. See Stopping and Restarting Asterisk From The CLI for more information.

- You can start Asterisk in the foreground, with an attached **root console**, using the `-c` option:

```
# asterisk -c

Asterisk 11.9.0, Copyright (C) 1999 - 2014 Digium, Inc. and others.
Created by Mark Spencer <markster@digium.com>
Asterisk comes with ABSOLUTELY NO WARRANTY; type 'core show warranty' for details.
This is free software, with components licensed under the GNU General Public
License version 2 and other licenses; you are welcome to redistribute it under
certain conditions. Type 'core show license' for details.
=========================================================================
Connected to Asterisk 11.9.0 currently running on asterisk-server (pid = 26246)
[May 16 17:02:50] NOTICE[27035]: loader.c:1323 load_modules: 287 modules will be loaded.
Asterisk Ready.
*CLI>
```

Adding Verbosity

Asterisk provides a number of mechanisms to control the verbosity of its logging. One way in which this can be controlled is through the command line parameter `-v`. For each `-v` specified, Asterisk will increase the level of `VERBOSE` messages by 1. The following will create a console and set the `VERBOSE` message level to 2:

```
# asterisk -c -v -v
```

Command line parameters can be combined. The previous command can also be invoked in the following way:

```
# asterisk -cvv
```

> The `VERBOSE` message level set via the command line is only applicable if the `asterisk.conf` `verbose` setting is not set.

Remote Console Verbosity

> **This feature is only available in Asterisk 11 and later versions.**

The verboseness of a remote console is set independently of the verboseness of other consoles and the core. A root console can be created with no verboseness:

```
# asterisk -c
```

While a remote console can be attached to that Asterisk process with a different verbosity:

```
# asterisk -rvvv
```

Multiple remote consoles can be attached, each with their own verbosity:

```
# asterisk -rv
```

Executing as another User

> **Do not run as root**
> Running Asterisk as `root` or as a user with super user permissions is dangerous and not recommended. There are many ways Asterisk can affect the system on which it operates, and running as `root` can increase the cost of small configuration mistakes.
>
> For more information, see the README-SERIOUSLY.bestpractices.txt file delivered with Asterisk.

Asterisk can be run as another user using the `-U` option:

```
# asterisk -U asteriskuser
```

Often, this option is specified in conjunction with the `-G` option, which specifies the group to run under:

```
# asterisk -U asteriskuser -G asteriskuser
```

When running Asterisk as another user, make sure that user owns the various directories that Asterisk will access:

```
# sudo chown -R asteriskuser:asteriskuser /usr/lib/asterisk
# sudo chown -R asteriskuser:asteriskuser /var/lib/asterisk
# sudo chown -R asteriskuser:asteriskuser /var/spool/asterisk
# sudo chown -R asteriskuser:asteriskuser /var/log/asterisk
# sudo chown -R asteriskuser:asteriskuser /var/run/asterisk
# sudo chown asteriskuser:asteriskuser /usr/sbin/asterisk
```

More Options

There are many more command line options available. For more information, use the `-h` option:

```
# asterisk -h
Asterisk 11.9.0, Copyright (C) 1999 - 2014, Digium, Inc. and others.
Usage: asterisk [OPTIONS]
...
```

Running Asterisk as a Service

The most common way to run Asterisk in a production environment is as a service. Asterisk includes both a `make` target for installing Asterisk as a service, as well as a script - `live_asterisk` - that will manage the service and automatically restart Asterisk in case of errors.

- Asterisk can be installed as a service using the `make config` target:

```
# make config
   /etc/rc0.d/K91asterisk -> ../init.d/asterisk
   /etc/rc1.d/K91asterisk -> ../init.d/asterisk
   /etc/rc6.d/K91asterisk -> ../init.d/asterisk
   /etc/rc2.d/S50asterisk -> ../init.d/asterisk
   /etc/rc3.d/S50asterisk -> ../init.d/asterisk
   /etc/rc4.d/S50asterisk -> ../init.d/asterisk
   /etc/rc5.d/S50asterisk -> ../init.d/asterisk
```

- Asterisk can now be started as a service:

```
# service asterisk start
 * Starting Asterisk PBX: asterisk                                                               [ OK ]
```

- And stopped:

```
# service asterisk stop
 * Stopping Asterisk PBX: asterisk                                                               [ OK ]
```

- And restarted:

```
# service asterisk restart
 * Stopping Asterisk PBX: asterisk                                                               [ OK ]
 * Starting Asterisk PBX: asterisk                                                               [ OK ]
```

Supported Distributions

Not all distributions of Linux/Unix are supported by the `make config` target. The following distributions are supported - if not using one of these distributions, the `make config` target may or may not work for you.

- RedHat/CentOS
- Debian/Ubuntu
- Gentoo
- Mandrake/Mandriva
- SuSE/Novell

Stopping and Restarting Asterisk From The CLI

There are three common commands related to stopping the Asterisk service. They are:

1. **core stop now** - This command stops the Asterisk service immediately, ending any calls in progress.
2. **core stop gracefully** - This command prevents new calls from starting up in Asterisk, but allows calls in progress to continue. When all the calls have finished, Asterisk stops.
3. **core stop when convenient** - This command waits until Asterisk has no calls in progress, and then it stops the service. It does not prevent new calls from entering the system.

There are three related commands for restarting Asterisk as well.

1. **core restart now** - This command restarts the Asterisk service immediately, ending any calls in progress.
2. **core restart gracefully** - This command prevents new calls from starting up in Asterisk, but allows calls in progress to continue. When all the calls have finished, Asterisk restarts.
3. **core restart when convenient** - This command waits until Asterisk has no calls in progress, and then it restarts the service. It does not prevent new calls from entering the system.

There is also a command if you change your mind.

- **core abort shutdown** - This command aborts a shutdown or restart which was previously initiated with the gracefully or when convenient options.

See the Asterisk Command Line Interface section for more on that topic.

Maintenance and Upgrades

So you have an Asterisk system running in production. Now what do you do?
Maintaining an Asterisk system can include tasks such as monitoring the system for issues, performing backups and keeping the system up to date. The topics of Asterisk Backups and Updating or Upgrading Asterisk are discussed on the linked sub-pages. A few miscellaneous maintenance topics are discussed below.

File sizes

Files generated by various Asterisk modules or core features may grow to significant sizes depending on how you use Asterisk and the configuration of those sub-systems.
Systems that may generate large files are:

- Logging
- Reporting
- Audio recording applications
  - There are multiple ways to record audio files, applications such as MixMonitor exist for the purpose of audio recording, other applications, e.g. ConfBridge provide audio recording as a sub-feature. The recordings will either go to your default sounds directory (Specified in asterisk.conf) or a directory specified via the application or a configuration file related to the responsible module.

The key is to know where these components store their output and to have some mechanism in place to prevent the files from growing to a point where you have no storage space remaining.

Managing log files in general is outside the scope of this documentation, however a little Internet searching will get you a long way in that area.

The Directory and File Structure wiki page will tell you where most Asterisk files are stored on the Linux file-system.

Security

It is in the interest of every Asterisk administrator to perform due diligence for security concerns. Most security measures are a matter of configuration and prevention, however for a production system already running there are a few things to consider in the context of maintenance.

- The Asterisk Security Event Logger can generate log output for security events. You may want to monitor these manually or have scripts and applications that take action on these events or log messages.
- Be aware of security vulnerability announcements. There are a few places these are announced:
  - http://www.asterisk.org/downloads/security-advisories
  - Asterisk-security mailing list
  - Asterisk-announce mailing list

Interfaces for Monitoring Asterisk Status

Maintenance can mean keeping an eye on the system and its state. The wiki discusses Asterisk interfaces, such as SNMP or APIs such as AMI. Through these interfaces you can monitor Asterisk in a variety of ways or even affect control over calls.

> **Reporting Issues that You Encounter**
> In the process of monitoring the operation of your system, you might spot an issue. If you believe the issue is a bug with Asterisk rather than a configuration issue, then you should follow the guidelines at the Asterisk Issue Guidelines page to report a bug.

Asterisk Backups

Backing up Asterisk Data

Backing up Asterisk is not a complex task. Mostly, you just need to know where the files are and then employ common tools for archiving and storing those files somewhere.

Files to consider for backup

- Asterisk configuration
- Asterisk internal DB
- Other database used by Asterisk
- Asterisk logs and reports

The Directory and File structure page should direct you to where most of these files reside. Otherwise check the individual wiki pages for information on the location of their output.

Other than just using **tar** to archive and compress the files, you might set up a cron job in Linux to regularly perform that process and send the files off-site. In general, use whatever backup processes you use for any other Linux applications that you manage.

Restoring a Backup

Restoring a backup, in most cases should be as simple as placing the files back in their original locations and starting Asterisk.

When restoring a backup to a new major version of Asterisk you'll need to take the same steps as if you were upgrading Asterisk. That is because a new major version may include changes to the format or syntax of configuration, required database schema, or applications and functions could be deprecated, removed or just have different behavior.

Updating or Upgrading Asterisk

Keeping the System Up-to-date

Definition of Updates and Upgrades

Updates involve installing a new minor version. That is, moving from 11.X.X to 11.X.X vs moving from 11 to 12. New minor versions include bug fixes, some of which may be security related. Typically, there should not be features or new functionality included in a minor version except when the fixing of a bug requires it.

Upgrades involve installing a new major version. For example, moving from 11 to 12. New major versions can include bug fixes, new features and functionality, changes to old features or functionality, deprecation of functionality or change in support status.

Updates and upgrades should only be performed when necessary

- Reason to Update
  - Your install is affected by a bug or security vulnerability and you believe the new minor version will fix your issue.
- Reason to Upgrade
  - You require new features or enhancements only available in a new major version and are ready for the work involved in upgrading.

When considering an update or upgrade you should be familiar with the Asterisk support life-cycle. It is useful to know the support status of the version you may be moving to.

Researching a New Asterisk Version

Included with Asterisk releases are a few files that are useful for researching the version you are considering for update or upgrade. These can be found in the root of the Asterisk source directory.

1. UPGRADE.txt - Documents any changes that you need to know about when moving between major versions. Especially changes that break backwards compatibility.
2. CHANGES - Documents new or enhanced functionality between versions listed in the file.
3. ChangeLog - A log showing all changes (bug fixes, new features, security fixes,etc) between released versions. It can be useful if you are searching for a specific fix or change, but this could be overly verbose for looking at changes between two major versions.

Performing Updates

Process

1. Research the new minor version you intend to update to.
2. Be sure you have a backup of any essential data on the system.
3. If you determine one of those changes will be beneficial for you, only then proceed with an update.
4. Download the new version and install Asterisk.

Performing Upgrades

Process

1. Research the new major version you are considering for an upgrade.
2. Be sure you have a backup of any essential data on the system.
3. If you determine the new functionality or changes will be beneficial then proceed with the upgrade.
4. On a test system, a non-production system, download and install the new version.
5. Migrate backups of configuration, databases and other data to the new Asterisk install.
6. Test this new system, or simulate your production environment before moving this new system into production.
   a. Especially test any areas of Asterisk where behavior changes have been noted in the UPGRADE.txt or CHANGES files. APIs, like AGI, AMI or ARI connecting to custom applications or scripts should be thoroughly tested. You should always try to extensively test your dialplan.

Third Party Modules

When updating or upgrading Asterisk you should also check for updates to any third party modules you use. That is, modules that are not distributed with Asterisk. Those third party modules may require updates to work with your new version of Asterisk.

Update and Upgrade Tips

✓ Updates and upgrades could include changes to configuration samples. Sample files will not be updated unless you run "make samples" again or copy the new sample files from the source directory. Be careful not to overwrite your current configuration.

✓ Keep old menuselect.makeopts files (see Asterisk source directory) and use them when building a new version to avoid customizing menuselect again when building a new version. This may only work for updates and not upgrades.

✓ If you forget to re-build all Asterisk modules currently installed on the system then you may be prompted after compilation with a warning about those modules. That can be resolved by simply re-building those modules or re-installing them if you obtain them in binary form from a third party.

Logging

Logging in Asterisk is a powerful mechanism that can be utilized to extract vital information from a running system. Asterisk currently has the capability to log messages to a variety of places that include files, consoles, and the syslog facility. Logging in Asterisk is configured in the logger.conf file. See Logging Configuration page for more information.

Along with the options defined in the logger configuration file, commands are available at runtime that allow a user to manipulate and even override certain settings via the CLI. Additionally flags are available with the Asterisk binary that allow similar configuration.

Most of the configuration for logging is concerning the activation or direction of various logging channels and their verbosity. It is important to note that the verbosity of logging messages is independent between root (or foreground) consoles and remote consoles. An example is provided in the Verbosity in Core and Remote consoles sub-section.

Dialplan Logging Applications

Logging can also be done in the dialplan utilizing the following applications:

Log(<level>, <message>)

Send arbitrary text to a selected log level, which must be one of the following: ERROR, WARNING, NOTICE, DEBUG, or VERBOSE.

Verbose([<level>,] <message>)

Send arbitrary text to verbose output. "Level" here is an optional integer value (defaults to 0) specifying the verbosity level at which to output the message.

Other Logging Resources

For information about extensive and detailed tracing of queued calls see the queue logs page. For instructions on how to help administrators and support givers to more quickly understand problems that occur during the course of calls see call identifier logging page. Also, if a problem is suspected see collecting debug information for help on how to collect debugging logs from an Asterisk machine (this can greatly help support and bug marshals). For details about logging security specific events see the asterisk security event logger page. Lastly, for advice on logging event data that can be grouped together to form a billing record see the channel event logging (CEL) page.

Basic Logging Commands

Here is a selection of basic logging commands to get you started with manipulating log settings at the Asterisk CLI.

core set verbose

Set the level of verbose messages to be displayed on the console. “0” or "off" means no verbose messages should be displayed. The silent option means the command does not report what happened to the verbose level. Equivalent to -v[v[...]] on start up.

```
Usage: core set verbose [atleast] <level> [silent]
```

core set debug

Set the level of debug messages to be displayed or set a module name to display debug messages from. "0" or "off" means no messages should be displayed. Equivalent to -d[d[...]] on start up.

```
Usage: core set debug [atleast] <level> [module]
```

logger show channels

List configured logger channels.

```
Usage: logger show channels
```

logger rotate

Rotates and Reopens the log files.

```
Usage: logger rotate
```

logger reload

Reloads the logger subsystem state. Use after restarting syslogd(8) if using syslog logging.

```
Usage: logger reload [<alt-conf>]
```

core show settings

Show miscellaneous core system settings. Along with showing other various settings, issuing this command will show the current debug level as well as the root and current console verbosity levels. These log settings can be found under the "PBX Core Settings" section after executing the command.

```
Usage: core show settings
```

Basic Logging Start-up Options

As previously indicated, at start up both the debug and verbose levels can also be set via command line arguments. To set the debug level on start up use the "-d" argument optionally followed by more "d"s. Asterisk will start with a debug level represented by the number of "d"s specified. As an example, the following will start Asterisk with a debug level of "3":

```
asterisk -ddd
```

To set the verbose level on start up use the "-v" argument optionally followed by more "v"s. Asterisk will start with a verbose level represented by the number of "v"s specified. As an example, the following will start Asterisk with a verbose level of "3":

```
asterisk -vvv
```

And of course both of these arguments can be specified at the same time:

```
asterisk -dddvvv
```

Call Identifier Logging

Overview

Call ID Logging (which has nothing to do with caller ID) is a new feature of Asterisk 11 intended to help administrators and support givers to more quickly understand problems that occur during the course of calls. Channels are now bound to call identifiers which can be shared among a number of channels, threads, and other consumers.

Usage

No configuration is needed to take advantage of this feature. Asterisk 11 will simply apply an additional bracketed tag to all log messages generated by a thread with a call ID bound or to any log messages specially written to use call identifiers. For example:

- Asterisk receives a request for a non existent extension from SIP/gold
- The following log message is displayed:

  ```
  [Oct 18 10:26:11] NOTICE[27538][C-00000000]: chan_sip.c:25107 handle_request_invite:
  Call from 'gold' (10.24.22.201:5060) to extension '645613' rejected because
  extension not found in context 'default'.
  ```

C-00000000 is the call identifier associated with this attempted call. All call identifiers are represented as C-XXXXXXXX where XXXXXXXX is an 8 digit hexadecimal value much like what you will see with SIP and local channel names.

Aside from log messages, call identifiers are also shown in the output for the 'core show channel <channel name>' command.

Transfers

Transfers can be a little tricky to follow with the call ID logging feature. As a general rule, an attended transfer will always result in a new call ID being made because a separate call must occur between the party that initiates the transfer and whatever extension is going to receive it. Once the attended transfer is completed, the channel that was transferred will use the Call ID created when the transferrer called the recipient.

Blind transfers are slightly more variable. If a SIP peer 'peer1' calls another SIP peer 'peer2' via the dial application and peer2 blind transfers peer1 elsewhere, the call ID will persist. If on the other hand, peer1 blind transfers peer2 at this point a new call ID will be created. When peer1 transfers peer2, peer2 has a new channel created which enters the PBX for the first time, so it creates a new call ID. When peer1 is transferred, it simply resumes running PBX, so the call is still considered the same call. By setting the debug level to 3 for the channel internal API (channel_internal_api.c), all call ID settings for every channel will be logged and this may be able to help when trying to keep track of calls through multiple transfers.

Collecting Debug Information

Collecting Debug Information for the Asterisk Issue Tracker

This document will provide instructions on how to collect debugging logs from an Asterisk machine, for the purpose of helping bug marshals troubleshoot an issue on https://issues.asterisk.org

If Asterisk has crashed or deadlocked, see Getting a Backtrace.

STEPS

Configure Asterisk logging

1. Edit the logger.conf file to enable specific logger channels to output to your filesystem. The word "debug_log_123456" can be changed to anything you want, as that is the filename the logging will be written to.

Modify the file name "debug_log_123456" to reflect your issues.asterisk.org issue number.

logger.conf

```
[logfiles]
debug_log_123456 => notice,warning,error,debug,verbose,dtmf
```

Asterisk 13+
In Asterisk 13 and later, you can dynamically create log channels from the CLI using the `logger add channel` command. For example, to create the log file above, you would enter:

```
logger add channel debug_log_123456 notice,warning,error,debug,verbose,dtmf
```

The new log channel persists until Asterisk is restarted, the logger module is reloaded, or the log files are rotated. If using this CLI command, do **not** reload/restart/rotate the log files in Step 2.

Configure verbosity levels and rotate logs

2. From the Asterisk CLI, set the verbose and debug levels for logging (this affects CLI and log output) and then restart the logger module:

```
*CLI> core set verbose 5
*CLI> core set debug 5
*CLI> module reload logger
```

Optionally, if you've used this file to record data previously, then rotate the logs:

```
*CLI> logger rotate
```

Enable channel tech or feature specific debug

2.1. Depending on your issue and if a protocol level trace is requested, be sure to enable logging for the channel driver or other module.

Module (version)	CLI Command
New PJSIP driver (12 or higher)	`pjsip set logger on`
SIP (1.6.0 or higher)	`sip set debug on`
SIP (1.4)	`sip set debug`
IAX2 (1.6.0 or higher)	`iax2 set debug on`

IAX2 (1.4)	`iax2 set debug`
CDR engine	`cdr set debug on`

Issue reproduction and clean up

3. Now that logging is configured, enabled and verbosity is turned up you should reproduce your issue.

4. Once finished, be sure to disable the extra debugging:

```
*CLI> core set verbose 0
*CLI> core set debug 0
```

4.1. Again, remember to disable any extra logging for channel drivers or features.

SIP (1.4 or higher)

```
*CLI> sip set debug off
```

IAX2 (1.4 or higher)

```
*CLI> iax2 set debug off
```

5. Disable logging to the filesystem. Edit the logger.conf file and comment out or delete the line you added in step 1. Using a semi-colon as the first character on the line will comment out the line.

logger.conf

```
[logfiles]
;debug_log_123456 => notice,warning,error,debug,verbose,dtmf
```

Then reload the logger module (or restart Asterisk) as you did in step 2:

```
*CLI> module reload logger
```

Provide debug to the developers

6. Upload the file located in /var/log/asterisk/debug_log_123456 to the issue tracker.

1. Do **NOT** post the output of your file as a comment. This clutters the issue and will only result in your comment being deleted.
2. Attach the file with a .txt extension to make it easy for the developers to quickly open the file without downloading. Files are attached on the issue page with following menu items: (More > Attach files)

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.

How do I interpret the lines in the Queue log?

The actual queue_log file will contain lines looking like the following:

```
1366720340|1366720340.303267|MYQUEUE|SIP/8007|RINGNOANSWER|1000
```

The pipe delimited fields from left to right are:

- UNIX timestamp
- Typically a Unique ID for the queue callers channel (based on the UNIX timestamp), also possible "REALTIME" or "NONE"
- Queue name
- Queue member channel
- Event type (see below reference)
- All fields to the right of the event type are event parameters

Queue log event types

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.

- ADDMEMBER - A member was added to the queue. The bridged channel name will be populated with the name of the channel added to the queue.

- 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)

- ATTENDEDTRANSFER(method|method-data|holdtime|calltime|origposition) - (Added in 12) This message will indicate the method by which the attended transfer was completed:`BRIDGE` for a bridge merge, `APP` for running an application on a bridge or channel, or `LINK` for linking two bridges together with local channels.

- BLINDTRANSFER(extension|context|holdtime|calltime|origposition) - (Added in 12) A blind transfer will result in a `BLINDTRANSFER` message with the destination context and extension.

- 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.
- REMOVEMEMBER - A queue member was removed from the queue. The bridge channel field will contain the name of the member removed from the queue.
- 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!
- RINGCANCELED - A caller is ringing a queue member, but that caller hangs up before the member answers or times out.
- 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.

Queue log options

There are one or more options for queue logging in queues.conf, such as "log_membername_as_agent". See the queues.conf sample file for explanations of those options.

Verbosity in Core and Remote Consoles

If one issues the "`core show settings`" command from the Asterisk CLI it will show both a "Root" and "Current" console verbosity levels. This is because each console, core or remote has an independent verbosity setting. For instance, if you start asterisk with the following command:

```
asterisk -cv
```

This starts Asterisk in console mode (will be the root console) with a verbose level set to "1". Now if one issues a "`core show settings`" from this console's CLI the following should be observed (note, not showing all settings):

```
*CLI> core show settings

PBX Core settings
-----------------
...
Root console verbosity:      1
Current console verbosity:   1
...
```

A remote console can now be attached with an initial verbosity level of "3" and a "`core show settings`" on that console should show a root console verbosity of "1" and a current console verbosity of "3":

```
asterisk -rvvv

*CLI> core show settings

PBX Core settings
-----------------
...
Root console verbosity:      1
Current console verbosity:   3
...
```

Changing the root console's verbosity will be reflected on both:

```
*CLI> core set verbose 2
Console verbose was 1 and is now 2.
*CLI> core show settings

PBX Core settings
-----------------
...
Root console verbosity:      2
Current console verbosity:   2
...
```

and then on the remote console:

```
*CLI> core show settings

PBX Core settings
-----------------
...
Root console verbosity:      2
Current console verbosity:   3
...
```

Asterisk Command Line Interface

What is the CLI?

The Command Line Interface, or console for Asterisk, serves a variety of purposes for an Asterisk administrator.

- Obtaining information on Asterisk system components
- Affecting system configuration
- Seeing log output, errors and warnings in real-time
- Generating calls for testing
- Viewing embedded help documentation such as for APIs, applications, functions and module configuration

The sub-sections under this page will discuss how to access and use the CLI. That is, CLI syntax, command line help, and other CLI related topics.

For information on configuration of the CLI, see the Asterisk CLI Configuration section of the wiki.

Connecting to the Asterisk CLI

There are two ways to connect to an Asterisk console, either a **foreground console** when starting the Asterisk process or by connecting to a **remote console** after Asterisk is already running.

Connecting to a foreground console

This is as simple as passing the `-c` flag when starting Asterisk.

```
-c     Provide  a control console on the calling terminal. The console is similar to the remote console provided
              by -r. Specifying this option implies -f and will cause asterisk to no longer fork  or  detach  from  the
              controlling terminal. Equivalent to console = yes in asterisk.conf.
```

When working on a foreground console, you won't be able to `exit`, instead you'll have to stop Asterisk to return to the Linux shell. Most people will use a remote console when performing administration tasks.

After Asterisk has finished loading, you'll see the default CLI prompt. The text "server" will be replaced with your system's hostname.

```
server*CLI>
```

Connecting to a remote console

Connecting to a remote console using the `-r` or `-R` flags.

```
-r     Instead of running a new Asterisk process, attempt to connect to a running Asterisk process and provide a
              console interface for controlling it.
       -R     Much like -r. Instead of running a new Asterisk process, attempt to connect to a running Asterisk process
              and provide a console interface for controlling it. Additionally, if connection to the  Asterisk  process
              is lost, attempt to reconnect for as long as 30 seconds.
```

To **exit a remote console**, simply type 'quit' or 'exit'. Please note that you can differentiate between a remote console and a foreground Asterisk console, as 'quit' or 'exit' will not function on the main console, which prevents an accidental shutdown of the daemon. If you would like to shutdown the Asterisk daemon from a remote console, there are various commands available.

Executing Command Outside Of CLI
You can execute an Asterisk command from outside the CLI:

```
$ asterisk -rx "core reload"
```

```
$ asterisk -rx "core show help" | grep -i "sip"
```

CLI Syntax and Help Commands

Command Syntax and Availability

Commands follow a general syntax of **<module name> <action type> <parameters>**.

For example:

- **sip show peers** - returns a list of chan_sip loaded peers
- **voicemail show users** - returns a list of app_voicemail loaded users
- **core set debug 5** - sets the core debug to level 5 verbosity.

> Commands are provided by the core, or by Asterisk modules. If the component that provides the commands is not loaded, then the commands it provides won't be available.

Asterisk does support command aliases. You can find information in the Asterisk CLI Configuration section.

Finding Help at the CLI

Command-line Completion

The Asterisk CLI supports command-line completion on all commands, including many arguments. To use it, simply press the **<Tab>** key at any time while entering the beginning of any command. If the command can be completed unambiguously, it will do so, otherwise it will complete as much of the command as possible. Additionally, Asterisk will print a list of all possible matches, if possible.

On this Page

Listing commands and showing usage

Once on the console, the 'help' alias (for 'core show help') may be used to see a large list of commands available for use.

```
*CLI> help
!                              -- Execute a shell command
acl show                       -- Show a named ACL or list all named ACLs
ael reload                     -- Reload AEL configuration
...
```

The 'help' alias may also be used to obtain more detailed information on how to use a particular command and listing sub-commands. For example, if you type 'help core show', Asterisk will respond with a list of all commands that start with that string. If you type 'help core show version', specifying a complete command, Asterisk will respond with a usage message which describes how to use that command. As with other commands on the Asterisk console, the help command also responds to tab command line completion.

```
*CLI> help core show
core show applications [like|describing] -- Shows registered dialplan applications
core show application          -- Describe a specific dialplan application
...
```

```
*CLI> help core show version
Usage: core show version
       Shows Asterisk version information.
```

Help for functions, applications and more

A big part of working with Asterisk involves making use of Asterisk applications and functions. Often you'll want to know usage details for these, including their overall behavior or allowed arguments and parameters.

The command **`core show application <application name>`** or similarly **`core show function <function name>`** will show you the usage and arguments.

```
*CLI> core show application Wait
  -= Info about application 'Wait' =-
[Synopsis]
Waits for some time.

[Description]
This application waits for a specified number of <seconds>.

[Syntax]
Wait(seconds)

[Arguments]
seconds
    Can be passed with fractions of a second. For example, '1.5' will
    ask the application to wait for 1.5 seconds.
```

Module Configuration Help

A very useful addition to Asterisk's help and documentation features is the command **`config show help`**. This command provides detailed information about configuration files, option sections in those files, and options within the sections.

```
*CLI> help config show help
Usage: config show help [<module> [<type> [<option>]]]
   Display detailed information about module configuration.
     * If nothing is specified, the modules that have
       configuration information are listed.
     * If <module> is specified, the configuration types
       for that module will be listed, along with brief
       information about that type.
     * If <module> and <type> are specified, detailed
       information about the type is displayed, as well
       as the available options.
     * If <module>, <type>, and <option> are specified,
       detailed information will be displayed about that
       option.
   NOTE: the help documentation is partially generated at run
     time when a module is loaded. If a module is not loaded,
     configuration help for that module may be incomplete.
```

For example maybe we see the 'callerid' option in a pjsip.conf file sent to us from a friend. We want to know what that option configures. If we know that pjsip.conf is provided by the res_pjsip module then we can find help on that configuration option.

```
*CLI> config show help res_pjsip endpoint callerid
[endpoint]
callerid = [(null)] (Default: n/a) (Regex: False)

CallerID information for the endpoint

 Must be in the format 'Name <Number>', or only '<Number>'.
```

Creating and Manipulating Channels from the CLI

Here we'll mention a few commands that allow you to create or manipulate channels at the CLI during runtime.

channel request hangup

Provided by the core, this command simply allows you to request that a specified channel or all channels be hungup.

```
Usage: channel request hangup <channel>|<all>
       Request that a channel be hung up. The hangup takes effect
       the next time the driver reads or writes from the channel.
       If 'all' is specified instead of a channel name, all channels
       will see the hangup request.
```

An example:

```
newtonr-laptop*CLI> core show channels
Channel              Location             State   Application(Data)
SIP/6001-00000001    (None)               Up      Playback(demo-congrats)
1 active channel

newtonr-laptop*CLI> channel request hangup SIP/6001-00000001
Requested Hangup on channel 'SIP/6001-00000001'
[May  2 09:51:19] WARNING[7045][C-00000001]: app_playback.c:493 playback_exec: Playback failed on SIP/6001-00000001 for
demo-congrats
```

Here I made a call to an extension calling Playback, then from the CLI I requested that the established channel be hung up. You can see that it hung up in the middle of playing a sound file, so that sound file fails to continue playing.

channel originate

Provided by res_clioriginate.so, this command allows you to create a new channel and have it connect to either a dialplan extension or a specific application.

```
There are two ways to use this command. A call can be originated between a
channel and a specific application, or between a channel and an extension in
the dialplan. This is similar to call files or the manager originate action.
Calls originated with this command are given a timeout of 30 seconds.
Usage1: channel originate <tech/data> application <appname> [appdata]
  This will originate a call between the specified channel tech/data and the
given application. Arguments to the application are optional. If the given
arguments to the application include spaces, all of the arguments to the
application need to be placed in quotation marks.
Usage2: channel originate <tech/data> extension [exten@][context]
  This will originate a call between the specified channel tech/data and the
given extension. If no context is specified, the 'default' context will be
used. If no extension is given, the 's' extension will be used.
```

An example:

```
newtonr-laptop*CLI> channel originate SIP/6001 extension 9999@somecontext
  == Using SIP RTP CoS mark 5
    -- Called 6001
    -- SIP/6001-00000004 is ringing
       > 0x7f0828067710 -- Probation passed - setting RTP source address to 10.24.18.16:4046
    -- SIP/6001-00000004 answered
    -- Executing [9999@somecontext:1] VoiceMailMain("SIP/6001-00000004", "") in new stack
    -- <SIP/6001-00000004> Playing 'vm-login.gsm' (language 'en')
       > 0x7f0828067710 -- Probation passed - setting RTP source address to 10.24.18.16:4046
```

We originated a call to the chan_sip peer 6001 in this case. The extension parameter tells it what extension to connect that channel to once the channel answers. In this case we connect it to an extension calling VoiceMailMain.

channel redirect

Provided by res_clioriginate.so, this command allows you to redirect an existing channel to a dialplan extension.

```
Usage: channel redirect <channel> <[[context,]exten,]priority>
    Redirect an active channel to a specified extension.
```

An example:

```
-- Executing [100@from-internal:1] Playback("SIP/6001-00000005", "demo-congrats") in new stack
       > 0x7f07ec03e560 -- Probation passed - setting RTP source address to 10.24.18.16:4048
    -- <SIP/6001-00000005> Playing 'demo-congrats.gsm' (language 'en')
newtonr-laptop*CLI> channel redirect SIP/6001-00000005 somecontext,9999,1
Channel 'SIP/6001-00000005' successfully redirected to somecontext,9999,1
[May  2 09:56:28] WARNING[7056][C-00000005]: app_playback.c:493 playback_exec: Playback failed on SIP/6001-00000005 for
demo-congrats
    -- Executing [9999@somecontext:1] VoiceMailMain("SIP/6001-00000005", "") in new stack
    -- <SIP/6001-00000005> Playing 'vm-login.gsm' (language 'en')
```

Here we make a call from SIP/6001 to a 100@from-internal, which results in a call to Playback. After the call is established, we issue a 'channel redirect' to redirect that channel to the extension 9999 in the context 'somecontext'. It is immediately placed into that extension and we hear the VoicemailMain prompt.

Simple CLI Tricks

Quit Typing the Whole Command

There are a couple of tricks that will help you on the Asterisk command-line interface. The most popular is tab completion. If you type the beginning of a command and press the Tab key, Asterisk will attempt to complete the name of the command for you, or show you the possible commands that start with the letters you have typed. For example, type co and then press the Tab key on your keyboard.

```
server*CLI> co[Tab]
config core
server*CLI> co
```

Now press the **r** key, and press tab again. This time Asterisk completes the word for you, as **core** is the only command that begins with **cor**. This trick also works with sub-commands. For example, type **core show** and press tab. (You may have to press tab twice, if you didn't put a space after the word **show**.) Asterisk will show you all the sub-commands that start with **core show**.

```
server*CLI> core show [Tab]
application     applications    calls           channel
channels        channeltype     channeltypes    codec
codecs          config          file            function
functions       help            hint            hints
image           license         profile         settings
switches        sysinfo         taskprocessors  threads
translation     uptime          version         warranty
server*CLI> core show
```

Repeat Previous Commands

Another trick you can use on the CLI is to cycle through your previous commands. Asterisk stores a history of the commands you type and you can press the **up arrow** key to cycle through the history.

Run Linux Shell Commands from The Asterisk CLI

If you type an exclamation mark at the Asterisk CLI, you will get a Linux shell. When you exit the Linux shell (by typing **exit** or pressing **Ctrl+D**), you return to the Asterisk CLI. You can also type an exclamation mark and a Linux command, and the output of that command will be shown to you, and then you'll be returned to the Asterisk CLI.

```
server*CLI> !whoami
root
server*CLI>
```

Asterisk Audio and Video Capabilities

Overview of Media Support

Asterisk supports a variety of audio and video media. Asterisk provides CODEC modules to facilitate encoding and decoding of audio streams. Additionally file format modules are provided to handle writing to and reading from the file-system.

The tables on this page describe what capabilities Asterisk supports and specific details for each format.

On This Page

Enabling specific media support

There are three basic requirements for making use of specific audio or video media with Asterisk.

1. The Asterisk core must support the format or a module may be required to add the support.
2. Asterisk configuration must be modified appropriately.
3. The devices interfacing with Asterisk must support the format and be configured to use it.

Module compilation and loading

For audio or video capabilities that require a module - you should make sure that the module is built and installed on the system.

See the section on Using Menuselect to Select Asterisk Options if you need help figuring out how to get a module built and then section on Configuring the Asterisk Module Loader to verify that a module gets loaded when Asterisk starts up.

Channel driver configuration

Audio or video capabilities for Asterisk are used on a per channel or per feature basis. To tell Asterisk what CODECs or formats to use in a particular scenario you may need to configure your channel driver, or modify configuration for the feature itself.

We'll provide two examples, but you should look at the documentation for the channel driver or feature to better understand how to configure media in that context.

Configuring allowed media for a PJSIP endpoint

pjsip.conf

```
[CATHY]
type=endpoint
context=from-internal
allow=!all,ulaw
auth=CATHY
aors=CATHY
```

We set the option "allow" to a string of values "!all,ulaw".

- The value "**!all**" means "Disallow all" and is identical to "disallow=all". This tells Asterisk to disallow all codecs except what we further define in the allow option.
- The value "**ulaw**" instructs Asterisk to allow ulaw audio during media negotiation for this endpoint.

See the section Configuring res_pjsip for more information on the PJSIP channel driver.

Configuring app_voicemail file formats for recordings

voicemail.conf

```
 [general]
format=wav49,wav,gsm
```

In the general section of voicemail.conf you can set the formats used when writing voicemail to the file-system. We set the option "format" to a string of file format names.

- The value "wav49" represents GSM in a WAV|wav49 container.
- The value "wav" represents SLIN in a wav container.
- The value "gsm" represents GSM in straight gsm format.

Endpoint device configuration

Configuring your particular device is outside the scope of the Asterisk documentation.

Consult your devices user/admin manual to find out where you define codecs or media to be used.

For VoIP desk phones there are typically two places to look for media configuration.

1. The web GUI for the phone.
2. The provisioning files that are pulled down by the phones on your network.

Audio Support

A variety of audio capabilities are supported by Asterisk.

Name	Config Value	Capability: (T)ranscoding (P)assthrough	CODEC Module	Format Module	Distributed w/ Asterisk?	Commercial License Required?
ADPCM	adpcm	T	codec_adpcm	format_vox	YES	NO
G.711 A-law	alaw	T	codec_alaw	format_pcm	YES	NO
G.711 µ-law	ulaw	T	codec_ulaw	format_pcm	YES	NO
G.719	g719	P	n/a	format_g719	YES	NO
G.722	g722	T	codec_g722	format_pcm	YES	NO

G.722.1 Siren7	siren7	T	codec_siren7	format_siren7	Codec(NO)
					Format(YES)
						NO

G.722.1C Siren14	siren14	T	codec_siren14	format_siren14	Codec(NO)
					Format(YES)
						NO

G.723.1	g723	T	codec_g723	format_g723	Codec(NO)
					Format(YES)
						YES(hardware required)

G.726	g726	T	codec_g726	format_g726	YES	NO
G.726 AAL2	g726aal2	T	codec_g726	format_g726	YES	NO

G.729A	g729	T	codec_g729a	format_g729	Codec(NO)
					Format(YES)
						YES

GSM	gsm	T	codec_gsm	format_gsm	YES	NO
ILBC	ilbc	T	codec_ilbc	format_ilbc	YES	NO
LPC-10	lpc10	T	codec_lpc10	n/a	YES	NO

SILK	silk	T	codec_silk	n/a	Codec(NO)
					Format(YES)
						NO

Speex	speex	T	codec_speex	n/a	YES	NO

Signed Linear PCM	slin	T	codec_resample	format_sln	YES	NO

Ogg Vorbis	n/a	n/a	n/a	format_ogg_vorbis	Codec(NO)
					Format(YES)
						NO

Opus	opus	T	codec_opus	n/a	Codec(NO)
					Format(YES)
						NO

wav (SLIN)	wav	T	n/a	format_wav	YES	NO
WAV (GSM)	wav49	T	n/a	format_wav_gsm	YES	NO

Speex Support

Asterisk supports 8, 16, and 32kHz Speex. Use of the 32kHz Speex mode is, like the other modes, controlled in the respective channel driver's configuration file, e.g. chan_sip's sip.conf or PJSIP's pjsip.conf.

Signed Linear PCM

Asterisk can resample between several different sampling rates and can read/write raw 16-bit signed linear audio files from/to disk. The complete list of supported sampling rates and file format is found in the expansion link below:

Click here to expand...

Sampling Rate	Asterisk File format
8kHz	.sln
12kHz	.sln12
16kHz	.sln16
24kHz	.sln24
32kHz	.sln32
44.1kHz	.sln44
48kHz	.sln48
96kHz	.sln96
192kHz	.sln192

Users can create 16-bit Signed Linear files of varying sampling rates from WAV files using the sox command-line audio utility.

```
sox input.wav -t raw -b 16 -r 32000 output.sln
mv output.sln output.sln32
```

In this example, an input WAV file has been converted to Signed Linear at a depth of 16-bits and at a rate of 32kHz. The resulting output.sln file is then renamed output.sln32 so that it can be processed correctly by Asterisk.

Video and Image Support

You'll notice the CODEC module column is missing. Video transcoding or image transcoding is not currently supported.

Name	Config Value	Capability: (T)ranscoding (P)assthrough	Format Module	Distributed w/ Asterisk
JPEG	jpeg	P	format_jpeg	YES
H.261	h261	P	n/a	YES
H.263	h263	P	format_h263	YES

H.263+	h263p	P	format_h263	YES
H.264	h264	P	format_h264	YES
VP8	vp8	P	n/a	YES
VP9	vp9	P	n/a	YES

Asterisk Community

- Asterisk Community Code of Conduct
- Asterisk Community Services
- Asterisk Issue Guidelines
- Asterisk Module Support States
- Asterisk Project Working Group Guidelines
- Community Services Signup
- Digium and Asterisk Community Export Compliance Notice
- IRC
- Mailing Lists
- Wiki Organization and Style Guide

Asterisk Community Code of Conduct

The Asterisk community is a large group of individuals, representing many nations, ethnicities, ages, technical professions and specialities. Working together on Asterisk can be a challenge with so many differing perspectives and backgrounds. Therefore to ensure the community is healthy, happy, and stress-free, participants in the Asterisk project agree to adhere to the following Community Code of Conduct.

Note that by joining and/or participating in the Asterisk community, you are agreeing that you accept and will adhere to the rules listed below, even if you do not explicitly state so.

Acceptable Behaviour

Be considerate

- Experience levels vary. Don't assume that someone can understand your particular explanation.
- Keep in mind that English is a second language for many users.

Be respectful

- It is possible to strongly disagree without using harsh language or resorting to derogatory comments. If you disagree with someone, disagree with the argument, not the character of the person.
- Remember that everyone is entitled to an opinion.

Ask for help

- If you don't know how to proceed with an aspect of development or documentation, ask for help! Always read the documentation, but don't be afraid to ask "silly" questions.
- When asking for help, take advantage of the resources that are available to you, including the wiki, mailing list archives, and Asterisk: The Definitive Guide.

Take responsibility

- If you did something wrong, apologize to the affected. Do your best to fix the issue and, if you can't, ask for help!
- If someone does take responsibility, be considerate.

Give credit

- Give proper credit to everyone involved In any contribution to the project, be it documentation, tests, code, or anything in between.
- If someone fails to give adequate contribution, gently remind them while being considerate. Assume that the omission was accidental, not malicious.

Unacceptable Behaviour

The Asterisk project reserves the right to take action in safe-guarding the community from those that participate in unacceptable behaviour. Unacceptable behaviour involves:

- Flaming - Arguing in a disrespectful way, attacking the character of others, rabidly ranting about things you dislike and refusing to drop the topic.
- Trolling - Intentionally baiting others into flaming or heated arguments for the sake of argument or drama itself.
- Mean-spirited or offensive talk - This could be combinations of the above, being rude, vulgar, and generally offensive to others.

In general, if community moderators and administrators are receiving many complaints about your behaviour, then you are likely doing something wrong. If you don't have anything nice to say, don't say anything.

Consequences to bad behaviour may involve bans from communication forums or channels and restrictions on privileges within the community.

Complaints about members behaviour or appeals in regards to bans or loss of privilege can be sent to asteriskteam@digium.com.

Open Community

We invite anybody, from any company, locale, or even other projects to participate in the Asterisk project. Our community is open, and contribution is welcome. Diversity makes a project strong, and we are proud to include anyone who wants to collaborate with others in a respectful fashion.

Project Leadership

The role of project leadership is handled by the founders of the Asterisk project, Digium Inc. As a member of the Asterisk community, Digium develops the project in co-operation with the overall Asterisk community. Community members are always welcome to take positions of leadership as module maintainers within the Asterisk project, particularly when they are the author of the module.

In addition to providing development resources for Asterisk itself, Digium provides community resources including the bug tracker, mailing lists, wiki, version control, continuous integration services, and other necessary project infrastructure. Asterisk goals and objectives are decided upon along with the community at the annual AstriDevCon held at AstriCon. Development discussions take place on the public asterisk-dev mailing list and the #asterisk-dev IR C channel. More information on the development of Asterisk can be found on the Asterisk wiki.

Asterisk Community Services

There are several services provided for the development of Asterisk.

- Issue Tracking
- Wiki
- Code Review
- Version Control
- Source Browsing
- Mailing Lists

There is a regular maintenance window every Monday from 9:00 PM to 10:00 PM Central Time, during which services may have intermittent availability while we apply patches, upgrades, etc. If maintenance is required outside of this window, notice will be sent to the asterisk-announce mailing list. If any of the community services are unavailable outside of a scheduled maintenance time, please notify the Asterisk development team.

Asterisk Issue Guidelines

Purpose of the Asterisk issue tracker

The Asterisk Issue Tracker is used to track bugs and miscellaneous (documentation) elements within the Asterisk project. The issue tracker is designed to manage reports on both core and extended components of the Asterisk project.

The primary use of the issue tracker is to track bugs, where "bug" means anything that causes unexpected or detrimental results in the Asterisk system. The secondary purpose is to track some of the miscellaneous issues surrounding Asterisk, such as documentation, commentary and feature requests or improvements with associated patches.

Feature requests without patches are generally not accepted through the issue tracker. Instead, they are discussed openly on the mailing lists and Asterisk IRC channels. Please read the "How to request a feature" section of this article.

What the issue tracker is not used for:

- **Information Requests**: (How does X parameter work?)
  See the forums, mailing lists, IRC channels, or this wiki. For even more information, see http://www.asterisk.org/community
- **Support requests**: (My phone doesn't register! My database connectivity doesn't work! How do I get it to work?)
  Search and ask on the forums, mailing lists, and IRC. Again, see http://www.asterisk.org/community for more information.
- **Random wishes and feature requests with no patch:** (I want Asterisk to support <insert obscure protocol or gadget>, but I don't know how to code!)
  See the How to request a feature section for more information on requesting a feature.
- **Business development requests** (I will pay you to make Asterisk support fancy unicorn protocol!)
  Please head to the asterisk-biz mailing list at http://lists.digium.com. If what you want is a specific feature or bug fixed, you may want to consider requesting a bug bounty.

Why should you read this?

The steps here will help you provide all the information the Asterisk team needs to solve your issue. Issues with clear, accurate, and complete information are often solved much faster than issues lacking the necessary information.

Bug Reporting Check List

Before filing a bug report...
Your issue may not be a bug or could have been fixed already. Run through the check list below to verify you have done your due diligence.

- **Are you reporting a suspected security vulnerability?**
  Please stop! The issue tracker is public, and once a security vulnerability is made public on the issue tracker, all users of Asterisk are made vulnerable. Please see Asterisk Security Vulnerabilities for more information on reporting a suspected security vulnerability.
- **Are you are on a supported version of Asterisk?**
  See the Asterisk Versions page for the currently supported versions of Asterisk. Bugs reported against versions of Asterisk that are no longer supported will be closed, and you will be asked to reproduce the issue against a supported version.
- **Are you using the latest version of your Asterisk branch?**
  Please check the release notes for newer versions of Asterisk to see if there is a potential fix for your issue. Even if you can't identify a fix in a newer version, it is preferable that you upgrade when reasonable to do so. Release notes are available in the UPGRADE.txt, CHANGES and ChangeLog files within the root directory of your particular release, and are also available at http://downloads.asterisk.org/pub/telephony/asterisk/
- **Are you using the latest third party software, firmware, model, etc?**
  If the error scenario involves phones, third party databases or other software, be sure it is all up to date and check their documentation.
- **Have you asked for help in the community? (mailing lists, IRC, forums)**
  You can locate all these services here: http://www.asterisk.org/community
- **Have you searched the Asterisk documentation in case this behavior is expected?**
  Search the Asterisk wiki for the problem or messages you are experiencing.
- **Have you searched the Asterisk bug tracker to see if an issue is already filed for this potential bug?**
  Search the Asterisk issue tracker for the issue you are seeing. You can search for issues by selecting **Issues -> Search for Issues** in the top menu bar.
- **Can you reproduce the problem?**
  Problems that can't be reproduced can often be difficult to solve, and your issue may be closed if you or the bug marshals cannot

reproduce the problem. If you can't find a way to simulate or reproduce the issue, then it is advisable to configure your system such that it is capturing relevant debug during the times failure occurs. Yes, that could mean running debug for days or weeks if necessary.

Submitting the bug report

You'll report the issue through the tracker (https://issues.asterisk.org/jira), under the ASTERISK project.

1. **Sign up for an asterisk.org account**
   Account signup for asterisk.org community services (including JIRA, Confluence and FishEye/Crucible) can be found at https://signup.asterisk.org.
2. **Create a new issue in the tracker**
   Select the **Create Issue** button in the top right hand corner of the page. This will prompt you for the project and issue type. Pick the ASTERISK project, and choose an appropriate issue type. The following are the supported issue types:
   - Bug
   - New feature
   - Improvement

   > ✓ Please do not select the Information request issue type. This type is not currently used, and your issue is likely to be closed or redefined.
3. **Fill out the issue form**
   For a bug you must include the following information:
   - **Concise and descriptive summary**
     - Accurate and descriptive, not prescriptive. Provide the facts of what is happening and leave out assumptions as to what the issue might be.
     - Good example: "Crash occurs when exactly twelve SIP channels hang up at the same time inside of a queue"
     - Bad Examples: "asterisk crashes" , "problem with queue", "asterisk doesn't work", "channel hangups cause crash"
   - **Operating System detail** (Linux distribution, kernel version, architecture etc)
   - **Asterisk version** (exact branch and point release, such as 1.8.12.0)
   - **Information on any third party software involved in the scenario** (database software, libraries, etc)
   - **Frequency and timing of the issue** (does this occur constantly, is there a trigger? Every 5 minutes? seemingly random?)
   - **Symptoms** described in specific detail ("No audio in one direction on only inbound calls", "choppy noise on calls where trans-coding takes place")
   - **Steps required to reproduce the issue** (tell the developer exactly how to reproduce the issue, just imagine you are making steps for a manual)
   - **Workarounds in detail with specific steps** (if you found a workaround for a serious issue, please include it for others who may be affected)
   - **Debugging output** - You'll almost always want to include extensions.conf, and config files for any involved component of Asterisk. Depending on the issue you may also need the following:
     - For crashes, provide a backtrace generated from an Asterisk core dump. See Getting a Backtrace for more information.
     - For apparent deadlocks, you may need to enable the compile time option `DEBUG_THREADS`. A backtrace may also be necessary. See Getting a Backtrace for more information.
     - For memory leaks or memory corruptions, Valgrind may be necessary. Valgrind can detect memory leaks and memory corruptions, although it does result in a substantial performance impact.
     - For debugging most problems, a properly generated debug log file will be needed. See CLI commands useful for debugging and Collecting Debug Information for more information. Note that for issues involving SIP, IAX2, or other channel drivers, you should enable that driver's enhanced debug mode through the CLI before collecting information. A pcap demonstrating the problem may also be needed.

     > ✓ Be courteous. Do not paste debug output in the description or a comment, instead please **attach** any debugging output as text files and reference them by file name.
4. **Submit the Issue**
   Once you have created the issue, you can now attach debug as files. You are ready to wait for a response from a bug marshal or post additional information as comments. If you are responding to a request for feedback, be sure to use the workflow actions to "send back" the status to the developers.

How you can speed up bug resolution

Follow the checklist and include all information that bug marshals require. Watch for emails where a bug marshal may ask for additional data and help the developers by testing any patches or possible fixes for the issue. A developer **cannot** fix your issue until they have sufficient data to reproduce - or at the very least understand - the problem.

Reasons your report may be closed without resolution

If your bug:

- Is solely the result of a configuration error
- Is a request for Asterisk support
- Has incomplete information
- Cannot be reproduced

it may be closed immediately by a bug marshal. If you believe this to be in error, please comment on the issue with the reason why you feel the issue is still a bug. You may also contact bug marshals directly in the #asterisk-bugs IRC channel. If that fails, bring it up on the mailing list(s) and it will be sorted out by the community.

Working with Bug Marshals

If insufficient commentary or debug information was given in the ticket then bug marshals will request additional information from the reporter. If there are questions posted as follow-ups to your bug or patch, please try to answer them - the system automatically sends email to you for every update on a bug you reported. If the original reporter of the patch/bug does not reply within some period of time (usually 14 days) and there are outstanding questions, the bug/patch may get closed out, which would be unfortunate. The developers have a lot on their plate and only so much time to spend managing inactive issues with insufficient information.

If your bug was closed, but you get additional debug or data later on, you can always contact a bug marshal in #asterisk-bugs on irc.freenode.net to have them re-open the issue.

How to request a feature

Feature requests without patches are not typically monitored or kept on the tracker. Even a feature request with a patch will still have to be approved by the core maintainers and community developers. Many people want Asterisk to do many different things; however, unless the feature has some extremely obvious (to many people) benefit, then it's best to have a community discussion and consensus on a feature before it goes into Asterisk. Feature requests are openly discussed on the mailing lists and IRC channels. Most bug marshals and Asterisk developers actively participate and will note the request if it makes sense.

New features and major architecture changes are often discussed at the AstriDevCon event held before Astricon.

If you really need the feature, but are not a programmer, you'll need to find someone else to write the code for you. Luckily, there are many talented Asterisk developers out there that can be contacted on the asterisk-biz mailing list. You can also offer a bounty for a bug or new feature - see Asterisk Bug Bounties for more information.

Patch and Code Submission

Patches for all issues are always welcome. Issues with patches are typically resolved much faster than those without. Please see the Patch Contribution Process for information on submitting a patch to the Asterisk project.

For information on writing a new feature for Asterisk, please see the New Feature Guidelines.

Issue Lifetime

The status of all issues in the tracker are reflected in the tracker. As issues are worked, they will move through several statuses. Issues that are "Closed" or "Complete" will have a disposition that determines their resolution. Issues with a disposition of "Fixed" have been committed to the project and will be released in the next bug fix release for all supported releases of Asterisk.

For any particular issue you'll want to look at the "Status" field and the comments tab under "Activity".

You can read more about the issue workflow at Issue Tracker Workflow.

Status Field Value	Meaning
Triage	The issue has not been acknowledged yet. First a bug marshal needs to verify it's a valid report. Check the comments!
Open\Reopened	Issue has been acknowledged and is waiting for a developer to take it on. Typically we can't provide an ETA for development as priorities are complex and change constantly.
In Progress	A developer is working on this issue. Check the comments!
Waiting for Feedback	Check the comments. This issue is waiting on the "Assignee" to provide feedback needed for it to move forward. Once feedback is provided you need to click "Send Back".
Closed	Issue has been closed. Check the "Resolution" field for further information.
Complete	The intended resolution was reached, but additional tasks may remain before the issue can be completely closed out.

Frequently Asked Questions

What makes a useful bug report?

There are some key qualities to keep in mind. These should be reflected in your bug report and will increase the chance of your bug being fixed!

- Specific: Pertaining to a certain, clearly defined issue with data to provide evidence of said issue
- Reproducible: Not random - you have some idea when this issue occurs and how to make it happen again

- Concise: Brief but comprehensive. Don't provide an essay on what you think is wrong. Provide only the facts and debug output that supports them.

How can I be notified when an issue's status changes?

Watch an issue: You can receive E-mails whenever an issue is updated. You'll see a "watch" link on the actual JIRA issue, click that!

What's the deal with the clone of my issue (SWP-1234)?

You can safely ignore the issues starting with SWP. Digium has a whole group of developers dedicated to working on Asterisk open source issues. They use a separate project for internal time tracking purposes only to avoid cluttering up the main project. The clone being created does not indicate any particular status, and any updates to the issue will be made in the ASTERISK project.

How can I speed up my issue resolution?

1. Follow the guidelines on this patch! Having good, accurate information that helps bug marshals reproduce the issue typically leads to much faster issue resolution.
2. Provide a patch! Issues with patches are also generally resolved much faster. If you can't write a patch, there are many smart, talented developers in the Asterisk community who may be worth helping you. You can contract with them on the asterisk-biz mailing list, or offer a bug bounty.

Asterisk Module Support States

Introduction

In Asterisk, modules can take one of three supported states. These states include:

- Core
- Extended
- Deprecated

The definition of the various states is described in the following sections.

Core

Most modules in Asterisk are in the Core state, which means issues found with these modules can freely be reported to the Asterisk issue tracker, where the issue will be triaged and placed into a queue for resolution.

Extended

This module is supported by the Asterisk community, and may or may not have an active developer. Some extended modules have active community developers; others do not. Issues reported against these modules may have a low level of support. For more information about which extended support modules have an active developer supporting them, see Asterisk Open Source Maintainers.

Deprecated

The module will remain in the tree, but there is a better way to do it. After two release cycles issues that have been deprecated for some time will be listed in an email to the Asterisk-Dev list where the community will have an opportunity to comment on whether a deprecated module: still compiles, works sufficiently well, and is still being utilized in a system where there is a justification for not using the preferred module.

MODULEINFO Configurations

At the top of modules there is the ability to set meta information about that module. Currently we have the following tags:

- **<defaultenabled>**
- **<use>**
- **<depend>**

The meta data added to **MODULEINFO** in the module causes data to be generated in *menuselect*. For example, when you use **<defaultenabled>no</defaultenabled>** the module will not be selected by default. We would use the **<defaultenabled>** tag for *deprecated* modules so that they are not built unless explicitly selected by an Asterisk administrator.

Adding New Metadata

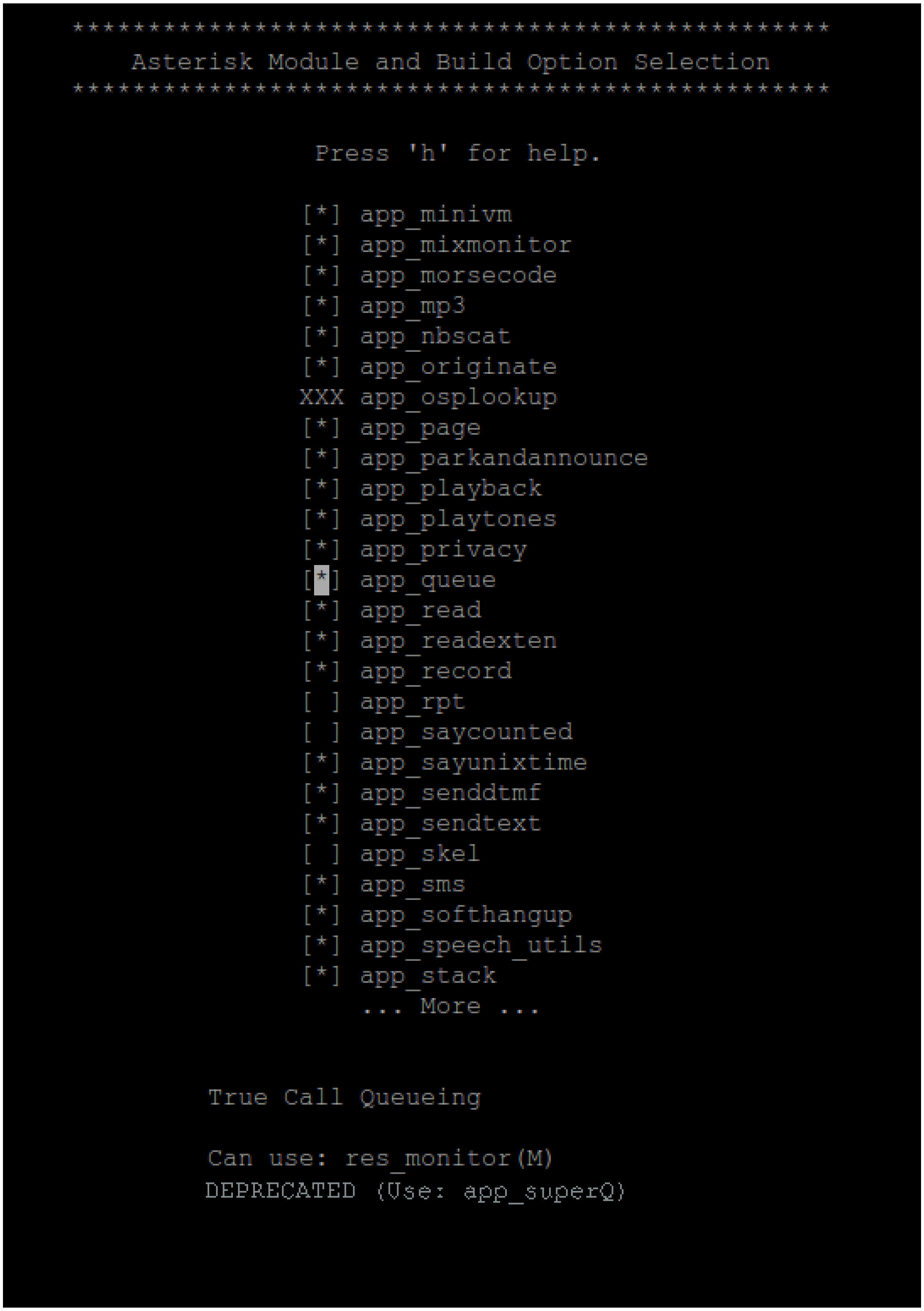
On top of the existing tags, we would add two additional tags to communicate via menuselect that a module was extended or deprecated (and what module supersedes the deprecated module). These tags include:

- **<support_level>**
  - *Example: <support_level>deprecated</support_level>* -- This module would be deprecated. Maintenance of this module may not exist. It is possible this module could eventually be tagged as deprecated, or removed from the tree entirely.
- **<replacement>**
  - *Example: <replacement>func_odbc</replacement>* -- The replacement for this module is the func_odbc module. This is used when the *<support_level>* is set to **deprecated**.

Menuselect Display

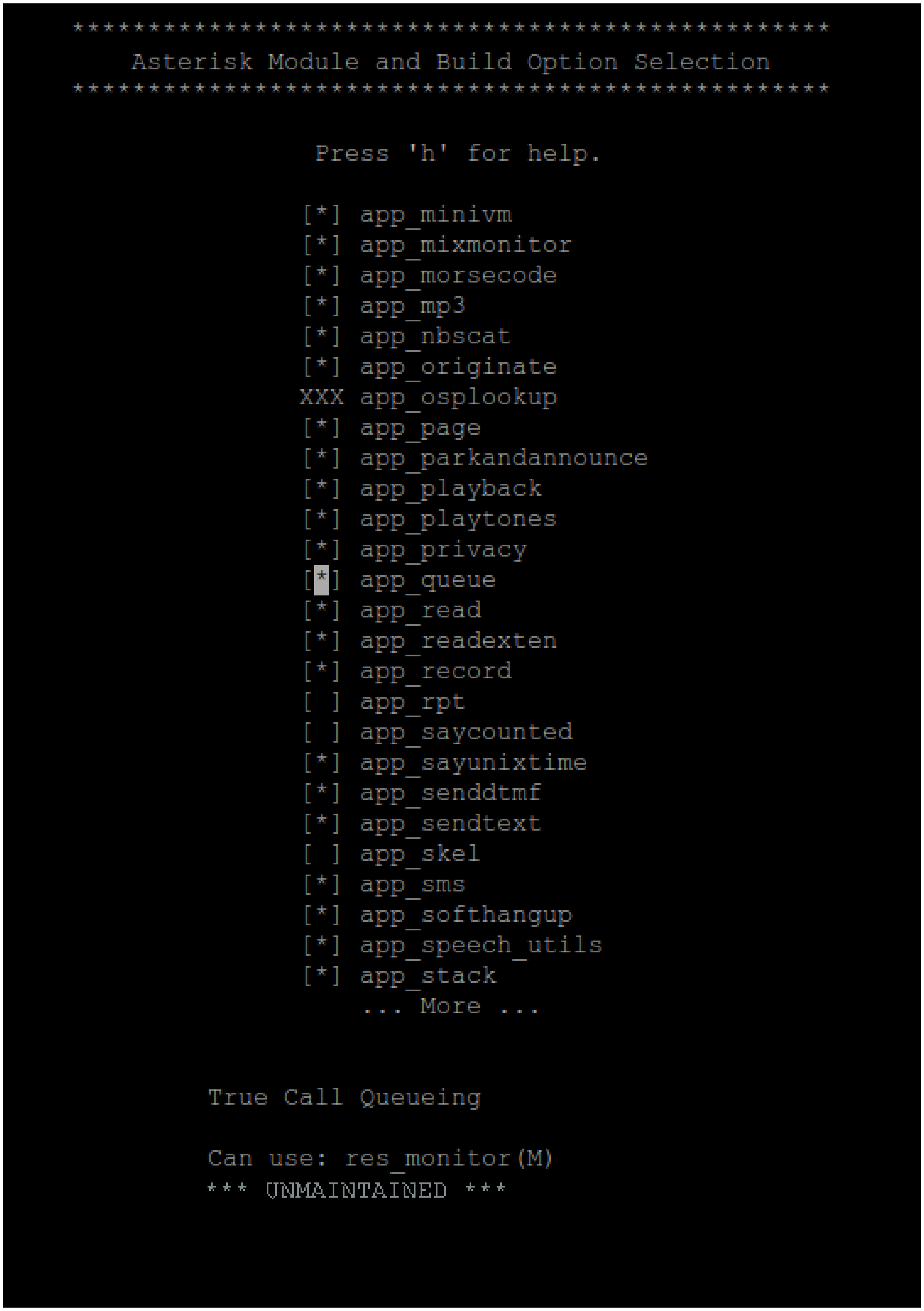
The following two images show the suggested menuselect output based on the addition of the *<support_level>* and *<replacement>* tags. This would be a new line that has not been used before, and therefore would be added to menuselect as that new line.

If the **deprecated** value is used, then the value between the **<replacement>** tags will replace the value of *app_superQ* as shown in the image below. The text surrounding *app_superQ* would be static (same for all modules that used **deprecated**).



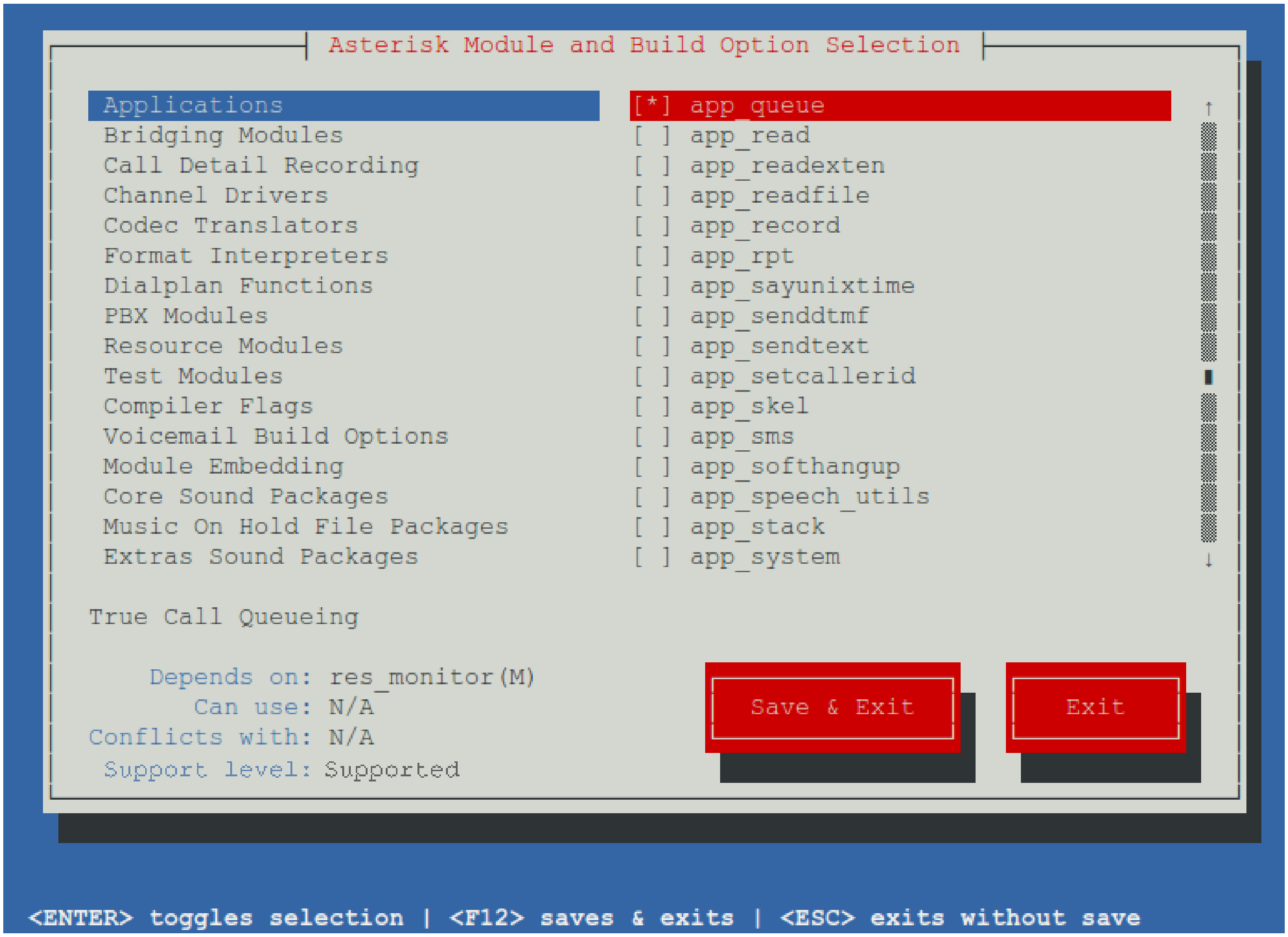
```
    ************************************************
        Asterisk Module and Build Option Selection
    ************************************************

                    Press 'h' for help.

                   [*] app_minivm
                   [*] app_mixmonitor
                   [*] app_morsecode
                   [*] app_mp3
                   [*] app_nbscat
                   [*] app_originate
                   XXX app_osplookup
                   [*] app_page
                   [*] app_parkandannounce
                   [*] app_playback
                   [*] app_playtones
                   [*] app_privacy
                   [*] app_queue
                   [*] app_read
                   [*] app_readexten
                   [*] app_record
                   [ ] app_rpt
                   [ ] app_saycounted
                   [*] app_sayunixtime
                   [*] app_senddtmf
                   [*] app_sendtext
                   [ ] app_skel
                   [*] app_sms
                   [*] app_softhangup
                   [*] app_speech_utils
                   [*] app_stack
                       ... More ...


             True Call Queueing

             Can use: res_monitor(M)
             DEPRECATED (Use: app_superQ)
```

If the **<support_level>** tag is used, then the value of *extended* would cause the additional text of ** EXTENDED ** to be displayed.



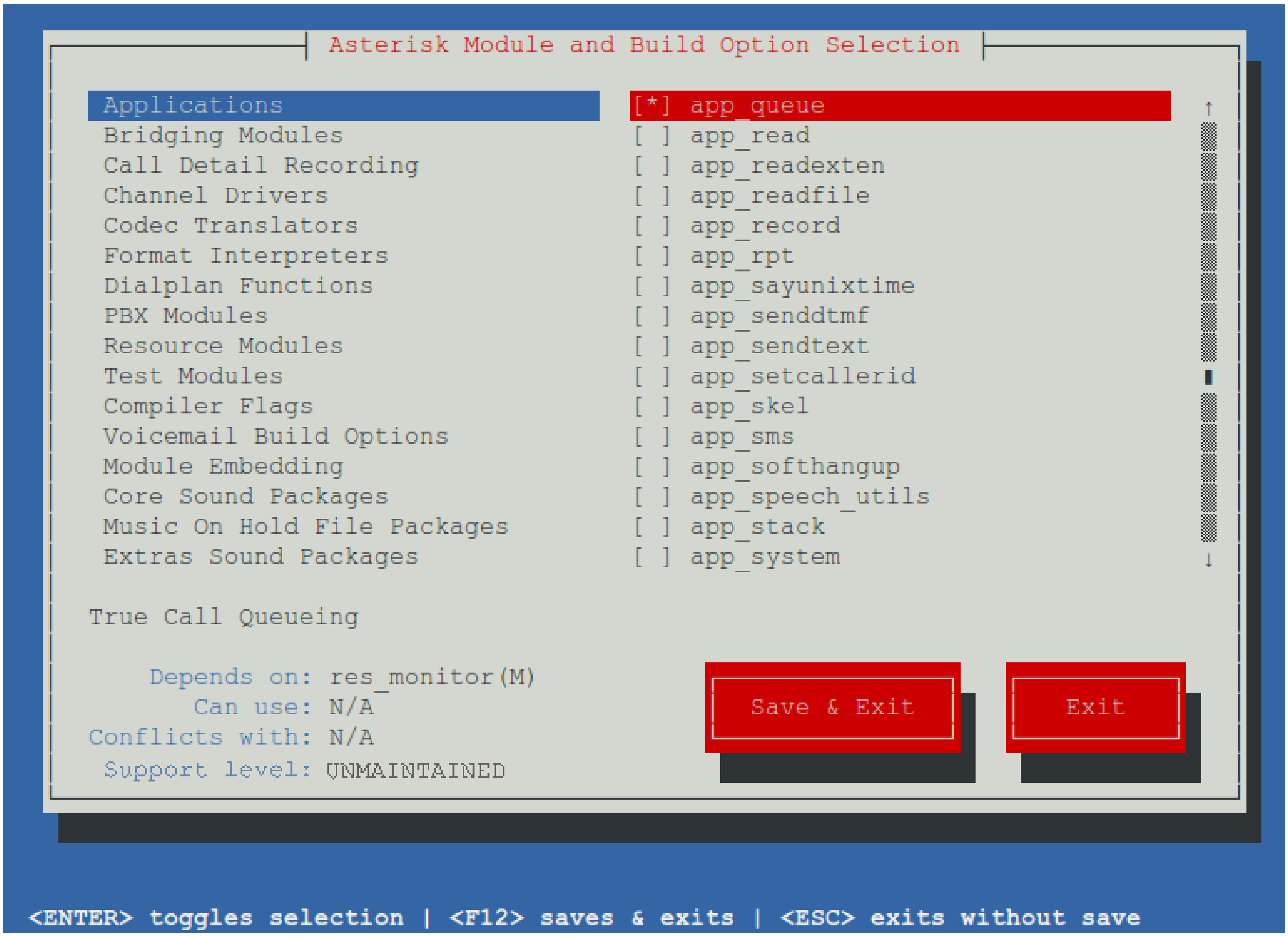
```
    **************************************************
        Asterisk Module and Build Option Selection
    **************************************************

                     Press 'h' for help.

                    [*] app_minivm
                    [*] app_mixmonitor
                    [*] app_morsecode
                    [*] app_mp3
                    [*] app_nbscat
                    [*] app_originate
                    XXX app_osplookup
                    [*] app_page
                    [*] app_parkandannounce
                    [*] app_playback
                    [*] app_playtones
                    [*] app_privacy
                    [*] app_queue
                    [*] app_read
                    [*] app_readexten
                    [*] app_record
                    [ ] app_rpt
                    [ ] app_saycounted
                    [*] app_sayunixtime
                    [*] app_senddtmf
                    [*] app_sendtext
                    [ ] app_skel
                    [*] app_sms
                    [*] app_softhangup
                    [*] app_speech_utils
                    [*] app_stack
                        ... More ...


             True Call Queueing

             Can use: res_monitor(M)
             *** UNMAINTAINED ***
```

Display in menuselect-newt for supported modules. If no <support_level> is specified, then it is assumed the module is supported:



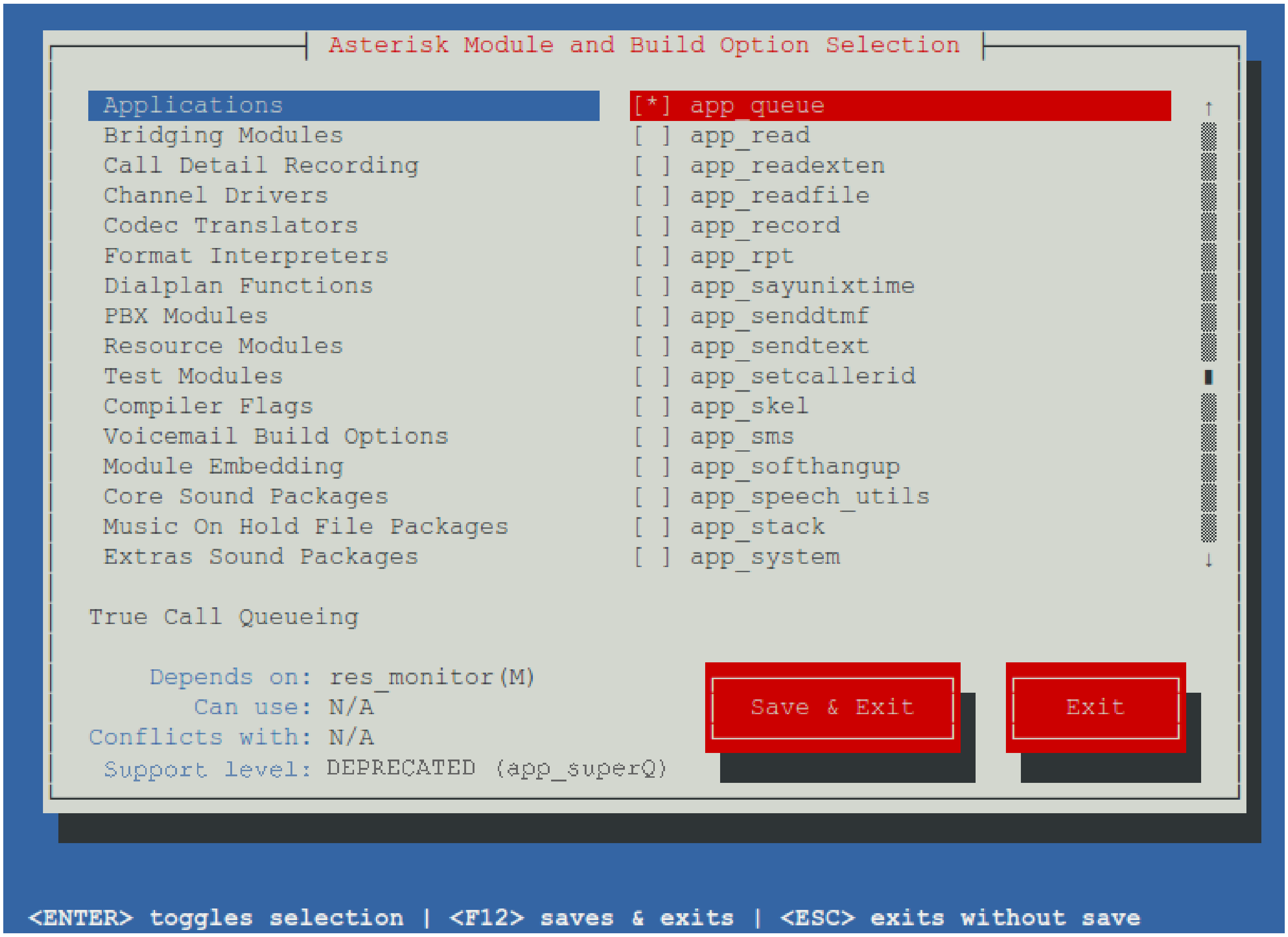
```
                 Asterisk Module and Build Option Selection

  Applications                        [*] app_queue
  Bridging Modules                    [ ] app_read
  Call Detail Recording               [ ] app_readexten
  Channel Drivers                     [ ] app_readfile
  Codec Translators                   [ ] app_record
  Format Interpreters                 [ ] app_rpt
  Dialplan Functions                  [ ] app_sayunixtime
  PBX Modules                         [ ] app_senddtmf
  Resource Modules                    [ ] app_sendtext
  Test Modules                        [ ] app_setcallerid
  Compiler Flags                      [ ] app_skel
  Voicemail Build Options             [ ] app_sms
  Module Embedding                    [ ] app_softhangup
  Core Sound Packages                 [ ] app_speech_utils
  Music On Hold File Packages         [ ] app_stack
  Extras Sound Packages               [ ] app_system

 True Call Queueing

     Depends on: res_monitor(M)
        Can use: N/A
 Conflicts with: N/A
  Support level: Supported

                                            Save & Exit        Exit

<ENTER> toggles selection | <F12> saves & exits | <ESC> exits without save
```

Display in menuselect-newt for extended modules:



```
           Asterisk Module and Build Option Selection

  Applications                        [*] app_queue              ↑
  Bridging Modules                    [ ] app_read
  Call Detail Recording               [ ] app_readexten
  Channel Drivers                     [ ] app_readfile
  Codec Translators                   [ ] app_record
  Format Interpreters                 [ ] app_rpt
  Dialplan Functions                  [ ] app_sayunixtime
  PBX Modules                         [ ] app_senddtmf
  Resource Modules                    [ ] app_sendtext
  Test Modules                        [ ] app_setcallerid
  Compiler Flags                      [ ] app_skel
  Voicemail Build Options             [ ] app_sms
  Module Embedding                    [ ] app_softhangup
  Core Sound Packages                 [ ] app_speech_utils
  Music On Hold File Packages         [ ] app_stack
  Extras Sound Packages               [ ] app_system             ↓

 True Call Queueing

     Depends on: res_monitor(M)
        Can use: N/A                     Save & Exit       Exit
 Conflicts with: N/A
  Support level: UNMAINTAINED

<ENTER> toggles selection | <F12> saves & exits | <ESC> exits without save
```

Display in menuselect-newt for deprecated modules:



```
              Asterisk Module and Build Option Selection

  Applications                        [*] app_queue
  Bridging Modules                    [ ] app_read
  Call Detail Recording               [ ] app_readexten
  Channel Drivers                     [ ] app_readfile
  Codec Translators                   [ ] app_record
  Format Interpreters                 [ ] app_rpt
  Dialplan Functions                  [ ] app_sayunixtime
  PBX Modules                         [ ] app_senddtmf
  Resource Modules                    [ ] app_sendtext
  Test Modules                        [ ] app_setcallerid
  Compiler Flags                      [ ] app_skel
  Voicemail Build Options             [ ] app_sms
  Module Embedding                    [ ] app_softhangup
  Core Sound Packages                 [ ] app_speech_utils
  Music On Hold File Packages         [ ] app_stack
  Extras Sound Packages               [ ] app_system

 True Call Queueing

     Depends on: res_monitor(M)
        Can use: N/A
 Conflicts with: N/A
  Support level: DEPRECATED (app_superQ)

                                          Save & Exit          Exit

<ENTER> toggles selection | <F12> saves & exits | <ESC> exits without save
```

Asterisk Project Working Group Guidelines

Here are a few guidelines that I have come up with governing working groups. Some of these guidelines come from the Node project, as they have a lot of pre-existing material on doing this. I deliberately avoided comprehensively importing their structure and guidelines, but pulled from some of their more essential core principles:

1. There is no explicit or implicit commitment that a working group’s output will actually be turned into code/patches by Digium or anybody else outside of the working group.

2. Some working group topics might include: documentation, feature request list, benchmarking, debug-ability, bug tracker triaging and replication, migration efforts from SIP to PJSIP (and more?)

3. They need somewhere to “work” - so a section of the asterisk.org wiki, mailing list, or public forum.

4. Need a regular (weekly?) meeting time and “place” (an Asterisk conference call, IRC, a google hangout, etc).

5. Need a charter of some sort. A charter would be a clearly defined mission statement determining the subject matter of the group’s efforts.

6. Need at least three initial members.

7. Need to follow a consensus seeking process for any decisions (https://en.wikipedia.org/wiki/Consensus-seeking_decision-making).

8. Membership cannot be changed (added or removed) without unanimous consensus of the members of the group.

9. In order to create one, talk to Matthew Fredrickson and he can see about getting infrastructure (mailing list, wiki, git access) setup.

10. Any working group must include at least one person from the Asterisk development team.

Please refer any questions about the guidelines to the asterisk-dev mailing list. Before petitioning to create a working group, try to ensure that you have met all the conditions listed above.

Community Services Signup

The asterisk.org community services use a centralized authentication system to manage user accounts. This means that the account that you use to log into issues.asterisk.org is the same account you can use to access wiki.asterisk.org.

To create an asterisk.org account, visit signup.asterisk.org and fill out the form.

The community services that currently support this centralized authentication are:

- issues.asterisk.org
- wiki.asterisk.org
- code.asterisk.org

Over time, accounts on our other community services will be moved to the same centralized authentication system.

Digium and Asterisk Community Export Compliance Notice

As a business based in the United States, Digium and Digium employees are subject to government regulations prohibiting U.S. organizations and businesses from engaging with people or entities that are located in embargoed countries. This pertains to all Asterisk community services which are provided by Digium, including the mailing lists, the Asterisk.org wiki, and any other hosted services that Digium provides.

If Digium becomes aware of someone from an embargoed country utilizing these services, we are under a regulatory obligation to remove any accounts in use by that person or persons. This is unfortunate for the project and all involved, but it is something that Digium as a company must do to comply with U.S. law.

For any U.S. citizens that find this situation to be unsatisfactory, please work with your political representatives to voice your opinion. For any non-U.S. citizens that this may apply to, we would urge you to work through your respective channels of government to work with U.S. ambassadors and other representatives to improve your situation.

For a list of embargoed countries see the two links below:

- https://www.bis.doc.gov/index.php/policy-guidance/country-guidance/sanctioned-destinations
- https://www.treasury.gov/resource-center/sanctions/Programs/Documents/ukraine_eo4.pdf

IRC

IRC

Use http://www.freenode.net IRC server to connect with Asterisk developers and users in realtime.

Asterisk Users Room

```
#asterisk
```

Asterisk Developers Room

```
#asterisk-dev
```

Mailing Lists

There are several mailing lists where community members can go do discuss Asterisk. Some of the key lists are:

List	Description
asterisk-addons-commits	SVN commits to the Asterisk addons project
asterisk-announce	Asterisk releases and community service announcements
asterisk-biz	Commercial and Business-Oriented Asterisk Discussion
Asterisk-BSD	Asterisk on BSD discussion
asterisk-bugs	Bugs
asterisk-commits	SVN commits to the Asterisk project
asterisk-dev	Discussions about the development of Asterisk. #Development List Note
asterisk-app-dev	Discussions about the development of applications that use Asterisk. #Development List Note
asterisk-doc	Discussions regarding The Asterisk Documentation Project
asterisk-embedded	Asterisk Embedded Development
asterisk-gui	Asterisk GUI project discussion
asterisk-gui-commits	SVN commits to the Asterisk-GUI project
asterisk-ha-clustering	Asterisk High Availability and Clustering List - Non-Commercial Discussion
Asterisk-i18n	Discussion of Asterisk internationalization
asterisk-security	Asterisk Security Discussion
asterisk-speech-rec	Use of speech recognition in Asterisk
asterisk-users	Discussions about the use and configuration of Asterisk.
asterisk-video	Development discussion of video media support in Asterisk
asterisk-wiki-changes	Changes to the Asterisk space on wiki.asterisk.org
svn-commits	SVN commits to the Digium repositories
Test-results	Results from automated testing

Development List Note

The asterisk-dev list is geared toward the development of Asterisk itself. For discussions about developing applications that *use* Asterisk, please see the asterisk-app-dev list. For discussions about deploying or using Asterisk, please see the asterisk-users mailing list.

Wiki Organization and Style Guide

Overview

This page should serve as a guide for creating organized, consistent, easy to consume content on the Asterisk wiki. It covers organization, content design and formatting.

You can find research all over the web on why these goals are important. Here's a link to Nielson Norman Group on How Users Read on the Web

Organization of Pages

Spaces, parent and child pages

All Asterisk documentation pages are contained within the Asterisk space. Most of those editing on wiki.asterisk.org will only have edit permissions within the Asterisk space.

Within the Asterisk space there are many top-level parent pages available. We've included some info about each below to help you decide where a page goes.

- About the Project - Any general information about the project. Licensing, History, What is Asterisk?
- Getting Started - Information relevant to completely new users, information on installation and how to get rolling into configuration and the rest of the documentation.
- Operation - Details concerning Asterisk's operation. That is, starting and stopping the Asterisk daemon, command line operation and other non-configuration tasks.
- Fundamentals - Basic, key and core concepts of Asterisk. Some of the most important foundational things to know about Asterisk.
- Configuration - How everything is configured. Where are the files? How do I use them? How do I program dialplan? How do I use the APIs?
- Deployment - Examples, tutorials, how-tos and recommendations for specific use-cases or scenarios of deployment. How do I deal with Asterisk in a NATed environment? How do I build a simple PBX?
- Development - All information regarding the development of Asterisk itself.
- Asterisk Test Suite Documentation - Documentation for the test suite! Primarily for developers.
- Asterisk Security Vulnerabilities - How the project handles security vulnerabilities.
- Asterisk Community - Anything falling under community. The places we meet, our code of conduct, community services.

Plus there are a few sections titled "Asterisk X Documentation" where X is the version. These sections are reserved for command reference content, mostly pushed to the wiki via scripts that pull from the source documentation.

> ⚠ Pages created or modified by "wikibot" are generated by scripts.

On this Page

Considerations before creating new content

Find out if a new page is really needed. Check the following:

- Have you searched the Asterisk space to see if it exists already?
- Is there already *some* content that you could use to build on?
- Is there related content which should be consolidated with your new content?

If obsolete related content exists and you decide to delete, rename or move it, then check the following:

- check for inbound links. If there are many referrers from external sources, you should ask the Asterisk project manager and others before renaming or deleting that page.
- If there are inbound links from wiki.asterisk.org pages, then we can of course go update those. For links from external sources, we have to decide how much we value breaking or not breaking those links.

Will the content fill more than a single screen-worth of space? If so, then you probably want to split it into multiple pages, or one page with sub-pages. When linking on a forum or mailing list, this makes it easier to link to specific chunks of content with shorter URLs.

Use the General Asterisk Wiki Page template for creating a new page with a very basic outline. It should include the Table of Contents macro discussed later on. The template is listed in the templates dialog when creating a new page.

Content Design

Design the content for a particular audience

Consider the audience for the content you are writing. A good clue is the section of the wiki you are placing the article in.

For example, pages in the Getting Started section should be oriented towards a completely new Asterisk user. That means the content should have a very narrow focus, making sure to explain concepts discussed and if a tutorial, provide small manageable chunks of tasks.

While it is acceptable to add content for very experienced Asterisk users, on the other end of the scale we do want to limit ourselves. Though Asterisk runs on Linux, the purpose of this wiki is not to teach Linux fundamentals. Pages in the Getting Started section do set the expectation that you should be familiar with Linux to use Asterisk.

Set expectations. In a section like Deployment, which could contain content for new users or experienced administrators, you should state requirements or experience at the beginning of the content.

Style based on purpose of content

As the heading says, the purpose of your content should determine what style you craft it in.

- **Explanatory or Descriptive** - These styles are useful for reference material. Pages describing configuration sections, (as opposed to how-to configure) and feature or functionality descriptions all should be written in these styles.
- **Procedural** - Configuration how-to, tutorials, examples and guides often include this style which often features lots of bulleted lists and step by step instructions.
- **Frequently Asked Questions** - Often a FAQ appears as a list of questions with their answers. Only use this style when there is no other better way to do it. You could format a lot of content this way, but it doesn't make sense most of the time.

Grammar, Punctuation, Spelling

The Ubuntu documentation team has a good guide for grammar, punctuation and spelling. Other than those rules, follow typical English language and writing conventions. It is always a good idea to have at least one or two other editors review your content for obvious problems.

Content Formatting and Markup

Naming a page

Page names should be concise. i.e., The simplest summary of the content within it or under it. That means taking into consideration sub-pages if the page in question is a top-level parent page.

Page names should be written in Title Case.

The page name will be included in the URL the wiki generates, so try to avoid any special characters or symbols and keep it as short as possible.

Headings

Good use of headings allows a reader to quickly break-down the content on the page to find what they need. Each heading should be a concise summary of the content under that heading.

Most pages won't use more than two or three levels of headings. h1 for sections, h2 for sub-sections, and h3 if you really need to break those sub-sections down further. If you find yourself getting into a lot of h3 or h4 headings, consider creating sub-pages rather than including so much nested content on a single page.

Use Title Case for h1 headings. For lower headings use only sentence case; that is, only use capitalization how you would in a typical sentence.

Linking

As you review content that you have written, try to link words to other content relevant to what you are discussing. Especially if it would help the user understand that topic.

There are a variety of ways to link to other pages and content. All of them are described in the confluence documentation, in Working with Links.

When linking from an external site, outside Confluence, it is important to use a permalink. Here is an excerpt from the Confluence documentation:

> The best way to link to a Confluence page from outside Confluence, for example on a website or in an email message, is to use the tiny link which is a permanent URL. This ensures that the link to the page is not broken if the page name changes.
>
> To access the permanent URL for a page:
>
> 1. View the page you wish to link to.
> 2. Choose **Tools** > **Link to this page**.
> 3. Copy the **Tiny Link**.
> 4. Use the tiny link in your website or email message.
>
> You do not need to use the tiny link to link to pages within your Confluence site. Confluence automatically updates links when you rename or move a page to another space.

Common macros

Confluence provides a variety of macros serving all sorts of functions. The confluence documentation on Working with Macros is very helpful to understand their usage. Below we'll list some macros that are commonly used with notes on how to use them specific to the Asterisk wiki.

Macro - Table of Contents

The TOC macro should be put at the top right-hand side of the page.

To make sure it aligns correctly you need to place a section macro, then two column macros inside the section. After that put a panel macro inside the second column and finally the TOC goes inside the panel. You can then edit the panel title to say something like "On this Page". For an example you can edit this page itself. Look at the very top and see how the TOC is laid out inside the the other macros.

Always set the TOC maxlevel to 2, as it is generally not necessary for a page-level TOC to have more detail.

Macro - No Format

Often when pasting text from an Asterisk log or the CLI you can run into issues as characters may be interpreted by the WYSIWYG editor as markup. Use the No Format macro to include a block of text to which no formatting will be applied.

Macro - Expand

This macro gives you a click-able link where upon being clicked provides a slide-out panel of content. Use this when you need to fit many large chunks of example code or terminal output into a small section. It'll prevent the code from taking up all the screen real-estate and then requiring the user to scroll through it all.

Macro - Warning, Info, Note, Tip

Try to avoid using these call-out boxes too often as they can be visually distracting and a few too many can make a page noisy. They are most useful with sparing application.

- Info - Most tame visually. Use this when you want to call-out a specific, brief piece of info that isn't necessarily critical, but could be very helpful.
- Note - A bright yellow exclamation sign. Maybe you want to share a word of caution or advise about requirements. "This example only works with a feature added in 11.1.1!"
- Tip - A bright green check mark. Useful when you want to mention shortcuts to a process already described detail, or call out brief command-line trick related to your content.
- Warning - This is bright red and very eye-catching. Use when you need to call out a piece of information very critical that could cause major problems for the user if not read or understood.