Asterisk Documentation

Asterisk Development Team <asteriskteam@digium.com>

Home

This is the home of the official wiki for The Asterisk Project.

This is not the first wiki that has existed for Asterisk, but there are some significant things that are different about this wiki than others. The most significant difference is that this wiki was created to be the official source of documentation for the Asterisk project, maintained by the same development team that manages the code itself. That means that we are committed to the content being correct and up to date. To make that happen, editing the content is not open to the general public. However, all Asterisk users are encouraged to participate by leaving comments on pages.

If you are an Asterisk expert and would like to get involved with the development and maintenance of content for the Asterisk wiki, contact Malcolm Davenport - malcolmd@digium.com.

Thank you very much for your continued support of Asterisk!

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Navigate space

Asterisk 12 Documentation

Asterisk 12 is not yet released. All information in this space is subject to change.

This space provides documentation for Asterisk 12. As a Standard Release of Asterisk, this version has the following milestones:

Milestone	Date
Initial Beta Release	??/??/2013
Initial Release	??/??/2013
End of Maintenance Window	??/??/2014
End of Security Fix Window	??/??/2015

As a Standard Release, the focus of development for this release of Asterisk included architectural improvements as well as major new features. This includes:

- A new extensible and performant SIP channel driver built on the pjsip SIP stack.
- Application of the Asterisk bridging framework throughout the project, providing consistency to management of channels while they are in a bridge.
- Stasis, a new internal message bus that conveys state about channels, bridges, endpoints, devices, and other objects throughout Asterisk.
- A heavily revamped version of AMI, focusing on improved channel semantics and consistency of event information.
- A new RESTful interface, ARI, which allows an external application to manipulate channels, bridges, and other telephony primitives directly.

It is highly recommended that users of Asterisk upgrading to Asterisk 12 read the UPDATE notes. In particular, users of AMI, CDRs, and CEL should read the respective specifications to assist with the behavioral changes.

New in 12

Overview

Asterisk 12 is a standard release of the Asterisk project. As such, the focus of development for this release was on core architectural changes and major new features. This includes:

* A more flexible bridging core based on the Bridging API
* A new internal message bus, Stasis
* Major standardization and consistency improvements to AMI
* Addition of the Asterisk RESTful Interface (ARI)
* A new SIP channel driver, chan_pjsip

In addition, as the vast majority of bridging in Asterisk was migrated to the Bridging API used by ConfBridge, major changes were made to most of the interfaces in Asterisk. This includes not only AMI, but also CDRs and CEL.

Specifications have been written for the affected interfaces:

* AMI 1.4 Specification
* Asterisk 12 CEL Specification
* Asterisk 12 CDR Specification

It is **highly** recommended that anyone migrating to Asterisk 12 read the information regarding its release both in the CHANGES files and in the accompanying UPGRADE.txt file.

Build System

* Added build option `DISABLE_INLINE`. This option can be used to work around a bug in gcc. For more information, see http://gcc.gnu.org/bugzilla/show_bug.cgi?id=47816
* Removed the `CHANNEL_TRACE` development mode build option. Certain aspects of the `CHANNEL_TRACE` build option were incompatible with the new bridging architecture.
* Asterisk now optionally uses `libxslt` to improve XML documentation generation and maintainability. If libxslt is not available on the system, some XML documentation will be incomplete.
* Asterisk now depends on `libjansson`. If a package of `libjansson` is not available on your distro, please see http://www.digip.org/jansson/.
* Asterisk now depends on `libuuid` and, optionally, `uriparser`. It is recommended that you install `uriparser`, even if it is optional.
* The new SIP stack and channel driver currently use a particular version of PJSIP. Please see https://wiki.asterisk.org/wiki/x/J4GLAQ for more information on configuring and installing PJSIP for usage with Asterisk.

Applications

AgentLogin

* Along with AgentRequest, this application has been modified to be a replacement for `chan_agent`. The act of a channel calling the AgentLogin application places the channel into a pool of agents that can be requested by the AgentRequest application. Note that this application, as well as all other agent related functionality, is now provided by the `app_agent_pool` module. See chan_agent and AgentRequest for more information.
* This application no longer performs agent authentication. If authentication is desired, the dialplan needs to perform this function using the Authenticate or VMAuthenticate application or through an AGI script before running AgentLogin.
* If this application is called and the agent is already logged in, the dialplan will continue exection with the `AGENT_STATUS` channel variable set to `ALREADY_LOGGED_IN`.
* The *agents.conf* schema has changed. Rather than specifying agents on a single line in comma delineated fashion, each agent is defined in a separate context. This allows agents to use the power of context templates in their definition.
* A number of parameters from *agents.conf* have been removed. This includes:
    * `maxloginretries`
    * `autologoffunavail`
    * `updatecdr`
    * `goodbye`
    * `group`
    * `recordformat`
    * `urlprefix`
    * `savecallsin`

  These options were obsoleted by the move from a channel driver model to the bridging/application model provided by `app_agent_pool`

.

AgentRequest

- A new application, this will request a logged in agent from the pool and bridge the requested channel with the channel calling this application. Logged in agents are those channels that called the AgentLogin application. If an agent cannot be requested from the pool, the `AGENT_STATUS` dialplan application will be set with an appropriate error value.

AgentMonitorOutgoing

- This application has been removed. It was a holdover from when AgentCallbackLogin was removed in Asterisk 1.6.0.

AlarmReceiver

- **Added support for additional Ademco DTMF signalling formats, including Express 4+1, Express 4+2, High Speed and Super Fast.**
- Added channel variable `ALARMRECEIVER_CALL_LIMIT`. This sets the maximum call time, in milliseconds, to run the application.
- Added channel variable `ALARMRECEIVER_RETRIES_LIMIT`. This sets the maximum number of times to retry the call.
- Added a new configuration option `answait`. If set, the AlarmReceiver application will wait the number of milliseconds specified by `answait` after the channel has answered. Valid values range between 500 milliseconds and 10000 milliseconds.
- Added configuration option `no_group_meta`. If enabled, grouping of metadata information in the AlarmReceiver log file will be skipped.

BridgeWait

- A new application in Asterisk, this will place the calling channel into a holding bridge, optionally entertaining them with some form of media. Channels participating in a holding bridge do not interact with other channels in the same holding bridge. Optionally, however, a channel may join as an announcer. Any media passed from an announcer channel is played to all channels in the holding bridge. Channels leave a holding bridge either when an optional timer expires, or via the ChannelRedirect application or AMI Redirect action.

ConfBridge

- All participants in a bridge can now be kicked out of a conference room by specifying the channel parameter as 'all' in the ConfBridge kick CLI command, i.e., `confbridge kick <conference> all`
- CLI output for the `confbridge list` command has been improved. When displaying information about a particular bridge, flags will now be shown for the participating users indicating properties of that user.
- The ConfbridgeList event now contains the following fields: `WaitMarked`, `EndMarked`, and `Waiting`. This displays additional properties about the user's profile, as well as whether or not the user is waiting for a Marked user to enter the conference.
- Added a new option for conference recording, `record_file_append`. If enabled, when the recording is stopped and then re-started, the existing recording will be used and appended to.

ControlPlayback

- The channel variable `CPLAYBACKSTATUS` may now return the value `REMOTESTOPPED`. This occurs when playback is stopped by a remote interface, such as AMI. See the AMI action ControlPlayback for more information.

Directory

- Added the a option, which allows the caller to enter in an additional alias for the user in the directory. This option must be used in conjunction with the `f`, `l`, or b options. Note that the alias for a user can be specified in *voicemail.conf*.

DumpChan

- The output of DumpChan no longer includes the `DirectBridge` or `IndirectBridge` fields. Instead, if a channel is in a bridge, it includes a `BridgeID` field containing the unique ID of the bridge that the channel happens to be in.

ForkCDR

- ForkCDR no longer automatically resets the forked CDR. See the `r` option for more information.

- Variables are no longer purged from the original CDR. See the v option for more information.
- The A option has been removed. The Answer time on a CDR is never updated once set.
- The d option has been removed. The disposition on a CDR is a function of the state of the channel and cannot be altered.
- The D option has been removed. Who the Party B is on a CDR is a function of the state of the respective channels involved in the CDR and cannot be altered.
- The r option has been changed. Previously, ForkCDR always reset the CDR such that the start time and, if applicable, the answer time was updated. Now, by default, ForkCDR simply forks the CDR, maintaining any times. The r option now triggers the Reset, setting the start time (and answer time if applicable) to the current time. Note that the a option still sets the answer time to the current time if the channel was already answered.
- The s option has been removed. A variable can be set on the original CDR if desired using the CDR function, and removed from a forked CDR using the same function.
- The T option has been removed. The concept of `DONT_TOUCH` and `LOCKED` no longer applies in the CDR engine.
- The v option now prevents the copy of the variables from the original CDR to the forked CDR. Previously the variables were always copied but were removed from the original. This was changed as removing variables from a CDR can have unintended side effects - this option allows the user to prevent propagation of variables from the original to the forked without modifying the original.

MeetMe

- Added the n option to MeetMe to prevent application of the DENOISE function to a channel joining a conference. Some channel drivers that vary the number of audio samples in a voice frame will experience significant quality problems if a denoiser is attached to the channel; this option gives them the ability to remove the denoiser without having to unload `func_speex`.

MixMonitor

- The b option now includes conferences as well as sounds played to the participants.
- The AUDIOHOOK_INHERIT function is no longer needed to keep a MixMonitor running during a transfer. If a MixMonitor is started on a channel, the MixMonitor will continue to record the audio passing through the channel even in the presence of transfers.

NoCDR

- The NoCDR application is deprecated. Please use the CDR_PROP function to disable CDRs.
- While the NoCDR application will prevent CDRs for a channel from being propagated to registered CDR backends, it will not prevent that data from being collected. Hence, a subsequent call to ResetCDR or the CDR_PROP function that enables CDRs on a channel will restore those records that have not yet been finalized.

ParkAndAnnounce

- The `app_parkandannounce` module has been removed. The application ParkAndAnnounce is now provided by the `res_parking` module. See the res_parking changes for more information.

Queue

- Added queue available hint. The hint can be added to the dialplan using the following syntax:

```
exten => {exten},hint,Queue:{queue_name}_avail
```

For example, if the name of the queue is `markq` and the extension is `8501`:

```
exten => 8501,hint,Queue:markq_avail
```

This will report `In Use` if there are no logged in agents or no free agents. It will report `Idle` when an agent is free.

- Queues now support a hint for member paused state. The hint uses the following syntax:

```
exten => {exten},hint,Queue:{queue_name}_pause_{member_name}
```

  Where `queue_name` and `member_name` are the name of the queue and the name of the member to subscribe to, respectively. For example, for the sales queue, with queue member mark at extension 8501:

```
exten => 8501,hint,Queue:sales_pause_mark
```

  Members will show as `In Use` when paused.
- The configuration options `eventwhencalled` and `eventmemberstatus` have been removed. As a result, the AMI events QueueMemberStatus, AgentCalled, AgentConnect, AgentComplete, AgentDump, and AgentRingNoAnswer will always be sent. The *Variable* fields will also no longer exist on the `Agent*` events. These events can be filtered out from a connected AMI client using the `eventfilter` setting in *manager.conf*.
- The queue log now differentiates between blind and attended transfers. A blind transfer will result in a `BLINDTRANSFER` message with the destination context and extension. An attended transfer will result in an `ATTENDEDTRANSFER` message. 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. The queue log will also now detect externally initiated blind and attended transfers and record the transfer status accordingly.
- When performing queue pause/unpause on an interface without specifying an individual queue, the `PAUSEALL`/{{UNPAUSEALL}} event will only be logged if at least one member of any queue exists for that interface.
- Added the `queue_log_realtime_use_gmt` option to have timestamps in GMT for realtime queue log entries.

ResetCDR

- The e option has been deprecated. Use the CDR_PROP function to re-enable CDRs when they were previously disabled on a channel.
- The `w` and a options have been removed. Dispatching CDRs to registered backends occurs on an as-needed basis in order to preserve linkedid propagation and other needed behavior.

SayAlphaCase

- A new application, this is similar to SayAlpha except that it supports case sensitive playback of the specified characters. For example:

```
same => n,SayAlphaCase(u,aBc)
```

  Will result in 'a uppercase b c'.

SetAMAFlags

- This application is deprecated in favor of CHANNEL(amaflags).

SendDTMF

- The SendDTMF application will now accept `W` as valid input. This will cause the application to delay one second while streaming DTMF.

Stasis

- A new application in Asterisk 12, this hands control of the channel calling the application over to an external system. Currently, external systems manipulate channels in Stasis through the Asterisk RESTful Interface (ARI).

UserEvent

- UserEvent will now handle duplicate keys by overwriting the previous value assigned to the key.
- In addition to AMI, UserEvent invocations will now be distributed to any interested Stasis applications.

VoiceMail

- The *voicemail.conf* configuration file now has an `alias` configuration parameter for use with the Directory application. The voicemail realtime database table schema has also been updated with an `alias` column.

Codecs

- Pass through support has been added for both VP8 and Opus.
- Added format attribute negotiation for the Opus codec. Format attribute negotiation is provided by the `res_format_attr_opus` module.

Core

- Masquerades as an operation inside Asterisk have been effectively hidden by the migration to the Bridging API. As such, many 'quirks' of Asterisk no longer occur. This includes renaming of channels, "<ZOMBIE>" channels, dropping of frame/audio hooks, and other internal implementation details that users had to deal with. This fundamental change has large implications throughout the changes documented for this version.
- Multiple parties in a bridge may now be transferred. If a participant in a multi-party bridge initiates a blind transfer, a Local channel will be used to execute the dialplan location that the transferer sent the parties to. If a participant in a multi-party bridge initiates an attended transfer, several options are possible. If the attended transfer results in a transfer to an application, a Local channel is used. If the attended transfer results in a transfer to another channel, the resulting channels will be merged into a single bridge.
- The channel variable `ATTENDED_TRANSFER_COMPLETE_SOUND` is no longer channel driver specific. If the channel variable is set on the transferrer channel, the sound will be played to the target of an attended transfer.
- The channel variable `BRIDGEPEER` becomes a comma separated list of peers in a multi-party bridge. The `BRIDGEPEER` value can have a maximum of 10 peers listed. Any more peers in the bridge will not be included in the list. {{BRIDGEPEER}} is not valid in holding bridges like parking since those channels do not talk to each other even though they are in a bridge.
- The channel variable `BRIDGEPVTCALLID` is only valid for two party bridges and will contain a value if the `BRIDGEPEER`'s channel driver supports it.
- A channel variable `ATTENDEDTRANSFER` is now set which indicates which channel was responsible for an attended transfer in a similar fashion to `BLINDTRANSFER`.
- Modules using the Configuration Framework or Sorcery must have XML configuration documentation. This configuration documentation is included with the rest of Asterisk's XML documentation, and is accessible via CLI commands. See the CLI changes for more information.

AMI (Asterisk Manager Interface)

- Major changes were made to both the syntax as well as the semantics of the AMI protocol. In particular, AMI events have been substantially improved in this version of Asterisk. For more information, please see the AMI specification.
- AMI events that reference a particular channel or bridge will now always contain a standard set of fields. When multiple channels or bridges are referenced in an event, fields for at least some subset of the channels and bridges in the event will be prefixed with a descriptive name to avoid name collisions. See the AMI event documentation for more information.
- The CLI command `manager show commands` no longer truncates command names longer than 15 characters and no longer shows authorization requirement for commands. `manager show command` now displays the privileges needed for using a given manager command instead.
- The SIPshowpeer action will now include a *SubscribeContext* field for a peer in its response if the peer has a subscribe context set.
- The SIPqualifypeer action now acknowledges the request once it has established that the request is against a known peer. It also issues a new event, SIPQualifyPeerDone, once the qualify action has been completed.
- The PlayDTMF action now supports an optional *Duration* parameter. This specifies the duration of the digit to be played, in milliseconds.
- Added VoicemailRefresh action to allow an external entity to trigger mailbox updates when changes occur instead of requiring the use of `pollmailboxes`.

- Added a new action ControlPlayback. The ControlPlayback action allows an AMI client to manipulate audio currently being played back on a channel. The supported operations depend on the application being used to send audio to the channel. When the audio playback was initiated using the ControlPlayback application or CONTROL STREAM FILE AGI command, the audio can be paused, stopped, restarted, reversed, or skipped forward. When initiated by other mechanisms (such as the Playback application), the audio can be stopped, reversed, or skipped forward.

- Channel related events now contain a snapshot of channel state, adding new fields to many of these events.
- The AMI event Newexten field *Extension* is deprecated, and may be removed in a future release. Please use the common *Exten* field instead.
- The AMI event UserEvent from `app_userevent` now contains the channel state fields. The channel state fields will come before the body fields.
- The AMI events ParkedCall, ParkedCallTimeOut, ParkedCallGiveUp, and UnParkedCall have changed significantly in the new `res_parking` module.
    - The *Channel* and *From* headers are gone.
        - For the channel that was parked or is coming out of parking, a *Parkee* channel snapshot is issued and it has a number of fields associated with it. The old *Channel* header relayed the same data as the new *ParkeeChannel* header.
        - The *From* field was ambiguous and changed meaning depending on the event. For most of these, it was the name of the channel that parked the call (the *Parker*).
    - There is no longer a header that provides this channel name, however the *ParkerDialString* will contain a dialstring to redial the device that parked the call.
    - On UnParkedCall events, the *From* header would instead represent the channel responsible for retrieving the parkee. It receives a channel snapshot labeled *Retriever*. The *From* field is is replaced with *RetrieverChannel*.
    - Lastly, the *Exten* field has been replaced with *ParkingSpace*.
- The AMI event Parkinglot (response to Parkinglots command) in a similar fashion has changed the field names *StartExten* and *StopExten* to *StartSpace* and *StopSpace* respectively.
- The deprecated use of | (pipe) as a separator in the `channelvars` setting in *manager.conf* has been removed.
- Channel Variables conveyed with a channel no longer contain the name of the channel as part of the key field, i.e., `ChanVariable(SIP/foo): bar=baz` is now `ChanVariable: bar=baz`. When multiple channels are present in a single AMI event, the various *ChanVariable* fields will contain a prefix that specifies which channel they correspond to.
- The NewPeerAccount AMI event is no longer raised. The NewAccountCode AMI event always conveys the AMI event for a particular channel.
- All Reload events have been consolidated into a single event type. This event will always contain a *Module* field specifying the name of the module and a *Status* field denoting the result of the reload. All modules now issue this event when being reloaded.
- The ModuleLoadReport event has been removed. Most AMI connections would fail to receive this event due to being connected after modules have loaded. AMI connections that want to know when Asterisk is ready should listen for the FullyBooted event.
- `app_fax` now sends the same send fax/receive fax events as `res_fax`. The FaxSent event is now the SendFAX event, and the FaxReceived event is now the ReceiveFAX event.
- The MusicOnHold event is now two events: MusicOnHoldStart and MusicOnHoldStop. The sub type field has been removed.
- The JabberEvent event has been removed. It is not AMI's purpose to be a carrier for another protocol.
- The Bridge Manager action's *Playtone* header now accepts more fine-grained options. `Channel1` and `Channel2` may be specified in order to play a tone to the specific channel. `Both` may be specified to play a tone to both
- channels. The old `yes` option is still accepted as a way of playing the tone to `Channel2` only.
- The AMI Status response event to the AMI Status action replaces the *BridgedChannel* and *BridgedUniqueid* headers with the *BridgeID* header to indicate what bridge the channel is currently in.
- The AMI Hold event has been moved out of individual channel drivers, into core, and is now two events: Hold and Unhold. The status field has been removed.
- The AMI events in `app_queue` have been made more consistent with each other. Events that reference channels (QueueCaller* and Agent*) will show information about each channel. The (infamous) Join and Leave AMI events have been changed to QueueCallerJoin and QueueCallerLeave.
- The MCID AMI event now publishes a channel snapshot when available and its non-channel-snapshot parameters now use either the *MCallerID* or *MConnectedID* prefixes with Subaddr*, Name*, and Num* suffixes instead of *CallerID* and *ConnectedID* to avoid confusion with similarly named parameters in the channel snapshot.
- The AMI events Agentlogin and Agentlogoff have been renamed AgentLogin and AgentLogoff respectively.
- The *Channel* key used in the AlarmClear, Alarm, and DNDState has been renamed *DAHDIChannel* since it does not convey an Asterisk channel name.
- ChannelUpdate events have been removed.
- All AMI events now contain a *SystemName* field, if available.
- Local channel optimization is now conveyed in two events: LocalOptimizationBegin and LocalOptimizationEnd. The Begin event is sent when the Local channel driver begins attempting to optimize itself out of the media path; the End event is sent after the channel halves have successfully optimized themselves out of the media path.
- Local channel information in events is now prefixed with *LocalOne* and *LocalTwo.* This replaces the suffix of '1' and '2' for the two halves of the Local channel. This affects the LocalBridge, LocalOptimizationBegin, and LocalOptimizationEnd events.

- The option `allowmultiplelogin` can now be set or overriden in a particular account. When set in the general context, it will act as the default setting for defined accounts.
- The BridgeAction event was removed. It technically added no value, as the Bridge Action already receives confirmation of the bridge through a successful completion Event.
- The BridgeExec events were removed. These events duplicated the events that occur in the Bridging API, and are conveyed now through BridgeCreate, BridgeEnter, and BridgeLeave events.
- The RTCPSent/RTCPReceived events have been significantly modified from previous versions. They now report all SR/RR packets sent/received, and have been restructured to better reflect the data sent in a SR/RR. In particular, the event structure now supports multiple report blocks.
- Added BlindTransfer and AttendedTransfer events. These events are raised when a blind transfer/attended transfer completes successfully. They contain information about the transfer that just completed, including the location of the transferred channel.

CDR (Call Detail Records)

- Significant changes have been made to the behavior of CDRs. The CDR engine was effectively rewritten and built on the Stasis message bus. For a full definition of CDR behavior in Asterisk 12, please read the CDR specification for Asterisk 12.
- CDRs will now be created between all participants in a bridge. For each pair of channels in a bridge, a CDR is created to represent the path of communication between those two endpoints. This lets an end user choose who to bill for what during bridge operations with multiple parties.
- The `duration`, `billsec`, `start`, `answer`, and `end` times now reflect the times associated with the current CDR for the channel, as opposed to a cumulative measurement of all CDRs for that channel.
- When a CDR is dispatched, user defined CDR variables from both parties are included in the resulting CDR. If both parties have the same variable, only the Party A value is provided.
- Added a new option to *cdr.conf*, `debug`. When enabled, significantly more information regarding the CDR engine is logged as verbose messages. This option should only be used if the behavior of the CDR engine needs to be debugged.
- Added CLI command `cdr set debug {on|off}`. This toggles the `debug` setting normally configured in *cdr.conf*.
- Added CLI command `cdr show active {channel}`. When `{channel}` is not specified, this command provides a summary of the channels with CDR information and their statistics. When `{channel}` is specified, it shows detailed information about all records associated with `{channel}`.

CEL (Channel Event Logging)

- CEL has undergone significant rework in Asterisk 12, and is now built on the Stasis message bus. Please see the specification for CEL fo r more detailed information.
- The *extra* field of all CEL events that use it now consists of a JSON blob with key/value pairs which are defined in the Asterisk 12 CEL documentation. This JSON blob will be interpreted by the various CEL backends into their specific format.
- `BLINDTRANSFER` events now report the transferee bridge unique identifier, extension, and context in a JSON blob as the extra string inst ead of the transferee channel name as the peer.
- `ATTENDEDTRANSFER` events now report the peer as NULL and additional information in the 'extra' string as a JSON blob. For transfers that occur between two bridged channels, the 'extra' JSON blob contains the primary bridge unique identifier, the secondary channel name, and the secondary bridge unique identifier. For transfers that occur between a bridged channel and a channel running an app, the 'extra' JSON blob contains the primary bridge unique identifier, the secondary channel name, and the app name.
- `LOCAL_OPTIMIZE` events have been added to convey local channel optimizations with the record occurring for the semi-one channel and the semi-two channel name in the peer field.
- `BRIDGE_START`, `BRIDGE_END`, `BRIDGE_UPDATE`, `3WAY_START`, `3WAY_END`, `CONF_ENTER`, `CONF_EXIT`, `CONF_START`, and `CONF_END` events have all been removed. These events have been replaced by `BRIDGE_ENTER`/ `BRIDGE_EXIT`. The `BRIDGE_ENTER` and `BRIDGE _EXIT` events are raised when a channel enters/exits any bridge, regardless of whether or not that bridge happens to contain multiple par ties.

CLI

- When compiled with `--enable-dev-mode`, the `astobj2` library will now add several CLI commands that allow for inspection of `ao2` co ntainers that register themselves with `astobj2`. The CLI commands are:
  - `astobj2 container dump`
  - `astobj2 container stats`
  - `astobj2 container check`
- Added specific CLI commands for bridge inspection. This includes:
  - `bridge show all` - list all bridges in the system
  - `bridge show {id}` - provide specific information about a bridge

* Added CLI command `bridge destroy`. This will destroy the specified bridge, ejecting the channels currently in the bridge. If the channels cannot continue in the dialplan or application that put them in the bridge, they will be hung up.
* Added command `bridge kick`. This will eject a single channel from a bridge.
* Added commands to inspect and manipulate the registered bridge technologies. This includes:
  * `bridge technology show` - list the registered bridge technologies
  * `bridge technology {suspend|unsuspend} {tech}` - control whether or not a registered bridge technology can be used during smart bridge operations. If a technology is suspended, it will not be used when a bridge technology is picked for channels; when unsuspended, it can be used again.
* The command `config show help` will show configuration documentation for modules with XML configuration documentation. This takes in three optional parameters - `module`, `type`, and `option -` which, when provided, provider greater detail.
  * When `module`, `type`, and `option` are omitted, a listing of all modules with registered documentation is displayed.
  * When `module` is specified, a listing of all configuration types for that module is displayed, along with their synopsis.
  * When `module` and `type` are specified, a listing of all configuration options for that type are displayed along with their synopsis.
  * When `module`, `type`, and `option` are specified, detailed information for that configuration option is displayed.
* Added `core show sounds` and `core show sound` CLI commands. These display a listing of all installed media sounds available on the system and detailed information about a sound, respectively.
* `xmldoc dump` has been added. This CLI command will dump the XML documentation DOM as a string to the specified file. The Asterisk core will populate certain XML elements pulled from the source files with additional run-time information; this command lets a user produce the XML documentation with all information.

Features

* Parking has been pulled from core and placed into a separate module called res_parking. Configuration for parking should now be performed in *res_parking.conf*. Configuration for parking in *features.conf* is now unsupported.
* Core attended transfers now have several new options. While performing an attended transfer, the transferer now has the following options:
  * `*1` - cancel the attended transfer (configurable via `atxferabort`)
  * `*2` - complete the attended transfer, dropping out of the call (configurable via `atxfercomplete`)
  * `*3` - complete the attended transfer, but stay in the call. This will turn the call into a multi-party bridge (configurable via `atxferthreeway`)
  * `*4` - swap to the other party. Once an attended transfer has begun, this options may be used multiple times (configurable via `atxferswap`)
* For DTMF blind and attended transfers, the channel variable `TRANSFER_CONTEXT` must be on the channel initiating the transfer to have any effect.
* The `BRIDGE_FEATURES` channel variable would previously only set features for the calling party and would set this feature regardless of whether the feature was in caps or in lowercase. Use of a caps feature for a letter will now apply the feature to the calling party while use of a lowercase letter will apply that feature to the called party.
* Add support for `automixmon` to the `BRIDGE_FEATURES` channel variable.
* The channel variable `DYNAMIC_PEERNAME` is redundant with `BRIDGEPEER` and is removed. The more useful `DYNAMIC_WHO_ACTIVATED` gives the channel name that activated the dynamic feature.
* The channel variables `DYNAMIC_FEATURENAME` and `DYNAMIC_WHO_ACTIVATED` are set only on the channel executing the dynamic feature. Executing a dynamic feature on the bridge peer in a multi-party bridge will execute it on all peers of the activating channel.
* You can now have the settings for a channel updated using the FEATURE() and FEATUREMAP() functions inherited to child channels by setting `FEATURE(inherit)=yes`.
* `automixmon` now supports additional channel variables from `automon` including: `TOUCH_MIXMONITOR_PREFIX`, `TOUCH_MIXMONITOR_MESSAGE_START`, and `TOUCH_MIXMONITOR_MESSAGE_STOP`.
* A new general *features.conf* option `recordingfailsound` has been added which allows setting a failure sound for a user tries to invoke a recording feature such as `automon` or `automixmon` and it fails.
* It is no longer necessary (or possible) to define the `ATXFER_NULL_TECH` in `features.c` for `atxferdropcall=no` to work properly. This option now just works.

Logging

* Added log rotation strategy `none`. If set, no log rotation strategy will be used. Given that this can cause the Asterisk log files to grow quickly, this option should only be used if an external mechanism for log management is preferred.

Realtime

* Dynamic realtime tables for SIP Users can now include a `path` field. This will store the path information for that peer when it registers.

Realtime tables can also use the `supportpath` field to enable Path header support.
- LDAP realtime configurations for SIP Users now have the `AstAccountPathSupport objectIdentifier`. This maps to the `supportpath` option in *sip.conf*.

Sorcery

- Sorcery is a new data abstraction and object persistence API in Asterisk. It provides modules a useful abstraction on top of the many storage mechanisms in Asterisk, including the Asterisk Database, static configuration files, static Realtime, and dynamic Realtime. It also provides a caching service. Users can configure a hierarchy of data storage layers for specific modules in *sorcery.conf*.
- 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.

Security Events Framework

- Security Event timestamps now use ISO 8601 formatted date/time instead of the `seconds-microseconds` format that it was using previously.

Stasis Message Bus

- The Stasis message bus is a publish/subscribe message bus internal to Asterisk. Many services in Asterisk are built on the Stasis message bus, including AMI, ARI, CDRs, and CEL. Parameters controlling the operation of Stasis can be configured in *stasis.conf*. Note that these parameters operate at a very low level in Asterisk, and generally will not require changes.

Channel Drivers

- When a channel driver is configured to enable jiterbuffers, they are now applied unconditionally when a channel joins a bridge. If a jitterbuffer is already set for that channel when it enters, such as by the JITTERBUFFER function, then the existing jitterbuffer will be used and the one set by the channel driver will not be applied.

chan_agent

- `chan_agent` has been removed and replaced with AgentLogin and AgentRequest dialplan applications provided by the `app_agent_pool` module. Agents are connected with callers using the new AgentRequest dialplan application. The `Agents:<agent-id>` device state is available to monitor the status of an agent. See *agents.conf.sample* for valid configuration options.
- The `updatecdr` option has been removed. Altering the names of channels on a CDR is not supported - the name of the channel is the name of the channel, and pretending otherwise helps no one. The `AGENTUPDATECDR` channel variable has also been removed, for the same reason.
- The `endcall` and `enddtmf` configuration options are removed. Use the dialplan function `CHANNEL(dtmf-features)` to set DTMF features on the agent channel before calling AgentLogin.

chan_bridge

- `chan_bridge` has been removed. Its functionality has been incorporated directly into the ConfBridge application itself.

chan_dahdi

- Added the CLI command `pri destroy span`. This will destroy the D-channel of the specified span and its B-channels. Note that this command should only be used if you understand the risks it entails.
- The CLI command `dahdi destroy channel` is now `dahdi destroy channels`. A range of channels can be specified to be destroyed. Note that this command should only be used if you understand the risks it entails.
- Added the CLI command `dahdi create channels`. A range of channels can be specified to be created, or the keyword `new` can be used to add channels not yet created.

chan_local

- The `/b` option has been removed.
- `chan_local` moved into the system core and is no longer a loadable module.

chan_mobile

- Added general support for busy detection.
- Added ECAM command support for Sony Ericsson phones.

chan_pjsip

- A new SIP channel driver for Asterisk, chan_pjsip is built on the PJSIP SIP stack. A collection of resource modules provides the bulk of the SIP functionality. For more information on configuring the new SIP channel driver, see the Configuring res_pjsip.

chan_sip

- Added support for RFC 3327 "Path" headers. This can be enabled in *sip.conf* using the `supportpath` setting, either on a global basis or on a peer basis. This setting enables Asterisk to route outgoing out-of-dialog requests via a set of proxies by using a pre-loaded route-set defined by the Path headers in the `REGISTER` request. See Realtime updates for more configuration information.
- The `SIP_CODEC` family of variables may now specify more than one codec. Each codec must be separated by a comma. The first codec specified is the preferred codec for the offer. This allows a dialplan writer to specify both audio and video codecs, e.g.,

```
same => n,Set(SIP_CODEC=ulaw,h264)
```

- The `callevents` parameter has been removed. Hold AMI events are now raised in the core, and can be filtered out using the `eventfilter` parameter in *manager.conf*.
- Added `ignore_requested_pref`. When enabled, this will use the preferred codecs configured for a peer instead of the requested codec.

chan_skinny

- Added the `immeddialkey` parameter. If set, when the user presses the configured key the already entered number will be immediately dialed. This is useful when the dialplan allows for variable length pattern matching. Valid options are `*` and `#`.
- Added the `callfwdtimeout` parameter. This configures the amount of time (in milliseconds) before a call forward is considered to not be answered.
- The `serviceurl` parameter allows Service URLs to be attached to line buttons.

Functions

AGENT

- The password option has been disabled, as the AgentLogin application no longer provides authentication.

AUDIOHOOK_INHERIT

- Due to changes in the Asterisk core, this function is no longer needed to preserve a MixMonitor on a channel during transfer operations and dialplan execution. It is effectively obsolete.

CDR (function)

- The `amaflags` and `accountcode` attributes for the CDR function are deprecated. Use the CHANNEL function instead to access these attributes.
- The `l` option has been removed. When reading a CDR attribute, the most recent record is always used. When writing a CDR attribute, all non-finalized CDRs are updated.
- The `r` option has been removed, for the same reason as the `l` option.
- The `s` option has been removed, as `LOCKED` semantics no longer exist in the CDR engine.

CDR_PROP

- A new function CDR_PROP has been added. This function lets you set properties on a channel's active CDRs. This function is write-only. Properties accept boolean values to set/clear them on the channel's CDRs. Valid properties include:
    - `party_a` - make this channel the preferred Party A in any CDR between two channels. If two channels have this property set, the creation time of the channel is used to determine who is Party A. Note that dialed channels are ever Party A in a CDR.
    - `disable` - disable CDRs on this channel. This is analogous to the NoCDR application when set to `True`, and analogous to the e option in ResetCDR when set to `False`.

CHANNEL

- Added the argument `dtmf_features`. This sets the DTMF features that will be enabled on a channel when it enters a bridge. Allowed values are `T`, `K`, `H`, `W`, and `X`, and are analogous to the parameters passed to the Dial application.
- Added the argument `after_bridge_goto`. This can be set to a parseable Goto string, i.e., `[[context],extension],priority`. When set, if a channel leaves a bridge but is not hung up it will resume dialplan execution at that location.

JITTERBUFFER

- JITTERBUFFER now accepts an argument of `disabled` which can be used to remove jitterbuffers previously set on a channel with JITTERBUFFER. The value of this setting is ignored when disabled is used for the argument.

PJSIP_DIAL_CONTACTS

- A new function provided by `chan_pjsip`, this function can be used in conjunction with the Dial application to construct a dial string that will dial all contacts on an Address of Record associated with a `chan_pjsip` endpoint.

PJSIP_MEDIA_OFFER

- Provided by `chan_pjsip`, this function sets the codecs to be offered on the outbound channel prior to dialing.

REDIRECTING

- Redirecting reasons can now be set to arbitrary strings. This means that the REDIRECTING dialplan function can be used to set the redirecting reason to any string. It also allows for custom strings to be read as the redirecting reason from SIP Diversion headers.

SPEECH_ENGINE

- The SPEECH_ENGINE function now supports read operations. When read from, it will return the current value of the requested attribute.

Resources

res_agi (Asterisk Gateway Interface)

- The manager event AGIExec has been split into AGIExecStart and AGIExecEnd.
- The manager event AsyncAGI has been split into AsyncAGIStart, AsyncAGIExec, and AsyncAGIEnd.
- The CONTROL STREAM FILE command now accepts an `offsetms` parameter. This will start the playback of the audio at the position specified. It will also return the final position of the file in `endpos`.
- The CONTROL STREAM FILE command will now populate the `CPLAYBACKSTATUS` channel variable if the user stopped the file playback or if a remote entity stopped the playback. If neither stopped the playback, it will indicate the overall success/failure of the playback. If stopped early, the final offset of the file will be set in the `CPLAYBACKOFFSET` channel variable.
- The SAY ALPHA command now accepts an additional parameter to control whether it specifies the case of uppercase, lowercase, or all letters to provide functionality similar to SayAlphaCase.

res_ari (Asterisk RESTful Interface) (and others)

- The Asterisk RESTful Interface (ARI) provides a mechanism to expose and control telephony primitives in Asterisk by remote client. This includes channels, bridges, endpoints, media, and other fundamental concepts. Users of ARI can develop their own communications applications, controlling multiple channels using an HTTP RESTful interface and receiving JSON events about the objects via a WebSocket connection. ARI can be configured in Asterisk via *ari.conf* .

res_parking

- Parking has been extracted from the Asterisk core as a loadable module, `res_parking`. Configuration for parking is now provided by *res_parking.conf*. Configuration through *features.conf* is no longer supported.

> ⚠ `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.

- Parked calls are now placed in bridges. While this is largely an architectural change, it does have implications on how channels in a parking lot are viewed. For example, commands that display channels in bridges will now also display the channels in a parking lot.
- The order of arguments for the new parking applications have been modified. Timeout and return `context/exten/priority` are now implemented as options, while the name of the parking lot is now the first parameter. See the application documentation for Park, ParkedCall, and ParkAndAnnounce for more in-depth information as well as syntax.
- Extensions are by default no longer automatically created in the dialplan to park calls or pickup parked calls. Generation of dialplan extensions can be enabled using the `parkext` configuration option.
- ADSI functionality for parking is no longer supported. The `adsipark` configuration option has been removed as a result.
- The `PARKINGSLOT` channel variable has been deprecated in favor of `PARKING_SPACE` to match the naming scheme of the new system.
- `PARKING_SPACE` and `PARKEDLOT` channel variables will now be set for a parked channel even when the configuration option `comebacktoorigin` is enabled.
- A new CLI command `parking show` has been added. This allows a user to inspect the parking lots that are currently in use. `parking show <parkinglot>` will also show the parked calls in a specific parking lot.
- The CLI command `parkedcalls` is now deprecated in favor of `parking show <parkinglot>`.
- The AMI command ParkedCalls will now accept a *ParkingLot* argument which can be used to get a list of parked calls for a specific parking lot.
- The AMI command Park field *Channel2* has been deprecated and replaced with *TimeoutChannel*. If both *Channel2* and *TimeoutChannel* are specified, *TimeoutChannel* will be used. The field *TimeoutChannel* is no longer a required argument.
- The ParkAndAnnounce application is now provided through `res_parking` instead of through the separate `app_parkandannounce` module.

- ParkAndAnnounce will no longer go to the next position in dialplan on timeout by default. Instead, it will follow the timeout rules of the parking lot. The old behavior can be reproduced by using the `c` option.
- Dynamic parking lots will now fail to be created under the following conditions:
  - If the parking lot specified by `PARKINGDYNAMIC` does not exist.
  - If they require exclusive park and parked call extensions which overlap with existing parking lots.
- Dynamic parking lots will be cleared on reload for dynamic parking lots that currently contain no calls. Dynamic parking lots containing parked calls will persist through the reloads without alteration.
- If `parkext_exclusive` is set for a parking lot and that extension is already in use when that parking lot tries to register it, this is now considered a parking system configuration error. Configurations which do this will be rejected.
- Added channel variable `PARKER_FLAT`. This contains the name of the extension that would be used if `comebacktoorigin` is enabled. This can be useful when `comebacktoorigin` is disabled, but the dialplan or an external control mechanism wants to use the extension in the park-dial context that was generated to re-dial the parker on timeout.

res_pjsip (and many others)

- A large number of resource modules make up the SIP stack based on PJSIP. The `chan_pjsip` channel driver users these resource modules to provide various SIP functionality in Asterisk. The majority of configuration for these modules is performed in *pjsip.conf* . Other modules may use their own configuration files. See Asterisk 12 Module Configuration for more information.

res_rtp_asterisk

- ICE/STUN/TURN support in `res_rtp_asterisk` has been made optional. To enable them, the Asterisk-specific version of PJSIP should now be installed. Tarballs are available from https://github.com/asterisk/pjproject/tags/.

res_statsd/res_chan_stats

- A new resource module, `res_statsd`, has been added, which acts as a statsd client. This module allows Asterisk to publish statistics to a statsd server. In conjunction with `res_chan_stats`, it will publish statistics about Asterisk channels to the statsd server. It can be configured via *res_statsd.conf*.

res_xmpp

- Device state for XMPP buddies is now available using the following format:

```
XMPP/<client name>/<buddy address>
```

  If any resource is available the device state is considered to be not in use. If no resources exist or all are unavailable the device state is considered to be unavailable.

Scripts

Realtime/Database Scripts

- Asterisk previously included example db schemas in the `contrib/realtime/` directory of the source tree. This has been replaced by a set of database migrations using the Alembic framework. This allows you to use Alembic to initialize the database for you. It will also serve as a database migration tool when upgrading Asterisk in the future. See contrib/ast-db-manage/README.md for more details.

sip_to_res_pjsip.py

- A new script has been added in the `contrib/scripts/sip_to_res_pjsip` folder. This python script will convert an existing *sip.conf* file to a *pjsip.conf* file, for use with the `chan_pjsip` channel driver. This script is meant to be an aid in converting an existing `chan_sip` configuration to a `chan_pjsip` configuration, but it is expected that configuration beyond what the script provides will be needed.

Upgrading to Asterisk 12

There are many significant architectural changes in Asterisk 12. It is recommended that you not only read through this document for important changes that affect an upgrade, but that you also read through the CHANGES document and the information about what is Ne w in 12 to better understand the new options available to you.

Upgrade Overview

Of particular note, the following systems in Asterisk underwent significant changes. Documentation for the changes and a specification for their behavior in Asterisk 12 are available in the Asterisk 12 Documentation section on this wiki.

- **AMI:** Many events were changed, and the semantics of channels and bridges were defined. In particular, how channels and bridges behave under transfer scenarios and situations involving multiple parties has changed significantly. See the AMI 1.4 Specification for more information.
- **CDRs:** CDR logic was extracted from the many locations it existed in across Asterisk and implemented as a consumer of Stasis message bus events. As a result, consistency of records has improved significantly and the behavior of CDRs in transfer scenarios has been defined in the CDR specification. However, significant behavioral changes in CDRs resulted from the transition. The most significant change is the addition of CDR entries when a channel who is the Party A in a CDR leaves a bridge. See the Asterisk 12 CDR Specification for more information.
- **CEL:** Much like CDRs, CEL was removed from the many locations it existed in across Asterisk and implemented as a consumer of Stasis message bus events. It now closely follows the Bridging API model of channels and bridges, and has a much closer consistency of

conveyed events as AMI. For the changes in events, see the Asterisk 12 CEL Specification.

Upgrade Changes

Build System:

- Removed the CHANNEL_TRACE development mode build option. Certain aspects of the CHANNEL_TRACE build option were incompatible with the new bridging architecture.
- Asterisk now depends on `libjansson`, `libuuid` and optionally (but recommended) `libxslt` and `uriparser`.
- The new SIP stack and channel driver uses a particular version of PJSIP. Please see Installing pjproject for more information on installing and configuring PJSIP for use with Asterisk 12.

AgentLogin, AgentRequest, and chan_agent:

- Along with AgentRequest, this application has been modified to be a replacement for `chan_agent`. The `chan_agent` module and the Agent channel driver have been removed from Asterisk, as the concept of a channel driver proxying in front of another channel driver was incompatible with the new architecture (and has had numerous problems through past versions of Asterisk). The act of a channel calling the AgentLogin application places the channel into a pool of agents that can be requested by the AgentRequest application. Note that this application, as well as all other agent related functionality, is now provided by the `app_agent_pool` module.
- This application no longer performs agent authentication. If authentication is desired, the dialplan needs to perform this function using the Authenticate or VMAuthenticate application or through an AGI script before running AgentLogin.
- The `agents.conf` schema has changed. Rather than specifying agents on a single line in comma delineated fashion, each agent is defined in a separate context. This allows agents to use the power of context templates in their definition.
- A number of parameters from `agents.conf` have been removed. This includes `maxloginretries`, `autologoffunavail`, `updatecdr`, `goodbye`, `group`, `recordformat`, `urlprefix`, and `savecallsin`. These options were obsoleted by the move from a channel driver model to the bridging/application model provided by `app_agent_pool`.
- The `AGENTUPDATECDR` channel variable has also been removed, for the same reason as the `updatecdr` option.
- The `endcall` and `enddtmf` configuration options are removed. Use the dialplan function `CHANNEL(dtmf-features)` to set DTMF features on the agent channel before calling AgentLogin.

AgentMonitorOutgoing

- This application has been removed. It was a holdover from when AgentCallbackLogin was removed.

ControlPlayback

- The channel variable `CPLAYBACKSTATUS` may now return the value `REMOTESTOPPED` when playback is stopped by an external entity.

DumpChan:

- The output of DumpChan no longer includes the *DirectBridge* or *IndirectBridge* fields. Instead, if a channel is in a bridge, it includes a *BridgeID* field containing the unique ID of the bridge that the channel happens to be in.

ForkCDR:

- Nearly every parameter in ForkCDR has been updated and changed to reflect the changes in CDRs. Please see the documentation for the ForkCDR application, as well as the CDR specification.

NoCDR:

- The NoCDR application has been deprecated. Please use the CDR_PROP function to disable CDRs on a channel.

ParkAndAnnounce:

- The `app_parkandannounce` module has been removed. The application ParkAndAnnounce is now provided by the res_parking module. See the Parking changes for more information.

ResetCDR:

- The w and a options have been removed. Dispatching CDRs to registered backends occurs on an as-needed basis in order to preserve linkedid propagation and other needed behavior.
- The e option is deprecated. Please use the CDR_PROP function to enable CDRs on a channel that they were previously disabled on.
- The ResetCDR application is no longer a part of core Asterisk, and instead is now delivered as part of `app_cdr`.

Queues:

- Queue strategy `rrmemory` now has a predictable order similar to strategy `rrordered`. Members will be called in the order that they are added to the queue.
- Removed the `queues.conf` `check_state_unknown` option. It is no longer necessary.
- It is now possible to play the Queue prompts to the first user waiting in a call queue. Note that this may impact the ability for agents to talk with users, as a prompt may still be playing when an agent connects to the user. This ability is disabled by default but can be enabled on an individual queue using the `announce-to-first-user` option.
- The configuration options `eventwhencalled` and `eventmemberstatus` have been removed. As a result, the AMI events QueueMemberStatus, AgentCalled, AgentConnect, AgentComplete, AgentDump, and AgentRingNoAnswer will always be sent. The *Variable* fields will also no longer exist on the *Agent** events. These events can be filtered out from a connected AMI client using the `eventfilter` setting in `manager.conf`.
- The queue log now differentiates between blind and attended transfers. A blind transfer will result in a `BLINDTRANSFER` message with the destination context and extension. An attended transfer will result in an `ATTENDEDTRANSFER` message. 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. The queue log will also now detect externally initiated blind and attended transfers and record the transfer status accordingly.
- When performing queue pause/unpause on an interface without specifying an individual queue, the `PAUSEALL`/ `UNPAUSEALL` event will only be logged if at least one member of any queue exists for that interface.

SetAMAFlags

- This application is deprecated in favor of CHANNEL(amaflags).

VoiceMail:

- The *voicemail.conf* configuration file now has an `alias` configuration parameter for use with the Directory application. The voicemail realtime database table schema has also been updated with an 'alias' column. Systems using voicemail with realtime should update their schemas accordingly.

Channel Drivers:

- When a channel driver is configured to enable jiterbuffers, they are now applied unconditionally when a channel joins a bridge. If a jitterbuffer is already set for that channel when it enters, such as by the JITTERBUFFER function, then the existing jitterbuffer will be used and the one set by the channel driver will not be applied.

chan_bridge

- chan_bridge is removed and its functionality is incorporated into ConfBridge itself.

chan_dahdi:

- Analog port dialing and deferred DTMF dialing for PRI now distinguishes between `w` and `W`. The `w` pauses dialing for half a second. The `W` pauses dialing for one second.
- The default for `inband_on_proceeding` has changed to no.
- The CLI command `dahdi destroy channel` is now `dahdi destroy channels`. A range of channels can be specified to be destroyed. Note that this command should only be used if you understand the risks it entails.

chan_local:

- The `/b` option has been removed.
- `chan_local` moved into the system core and is no longer a loadable module.

chan_sip:

- The `callevents` parameter has been removed. Hold AMI events are now raised in the core, and can be filtered out using the `eventfilter` parameter in *manager.conf*.
- Dynamic realtime tables for SIP Users can now include a `path` field. This will store the path information for that peer when it registers. Realtime tables can also use the `supportpath` field to enable Path header support.
- LDAP realtime configurations for SIP Users now have the `AstAccountPathSupport` `objectIdentifier`. This maps to the `supportpath` option in *sip.conf*.

Core:

- Masquerades as an operation inside Asterisk have been effectively hidden by the migration to the Bridging API. As such, many 'quirks' of Asterisk no longer occur. This includes renaming of channels, "<ZOMBIE>" channels, dropping of frame/audio hooks, and other internal implementation details that users had to deal with. This fundamental change has large implications throughout the changes documented for this version. For more information about the new core architecture of Asterisk, please see the Asterisk wiki.
- The following channel variables have changed behavior which is described in the CHANGES file: `TRANSFER_CONTEXT`, `BRIDGEPEER`, `BRIDGEPVTCALLID`, `ATTENDED_TRANSFER_COMPLETE_SOUND`, `DYNAMIC_FEATURENAME`, and `DYNAMIC_PEERNAME`.
- All bridging in Asterisk is now performed using the Bridging API, which is the same bridging core that powers the ConfBridge application. As a result, significant changes have been made in all areas of Asterisk that are affected by bridging. Those changes are noted in the appropriate areas.

AMI (Asterisk Manager Interface):

- The Version has been increased to 1.4. For a full listing of the semantics changes in AMI, see the AMI 1.4 Specification.
- The details of what happens to a channel when a masquerade happens (transfers, parking, etc) have changed. Channels no longer swap Uniqueid's as a result of the masquerade. In general, AMI clients will never actually "see" a masquerade, as the operation has been effectively hidden from external systems.
- **Major** changes were made to both the syntax as well as the semantics of the AMI protocol. In particular, AMI events have been substantially modified and improved in this version of Asterisk. The major event changes are listed below:
  - NewPeerAccount has been removed. NewAccountCode is raised instead.
  - Reload events have been consolidated and standardized.
  - ModuleLoadReport has been removed.
  - FaxSent is now SendFAX; FaxReceived is now ReceiveFAX. This standardizes `app_fax` and `res_fax` events.
  - MusicOnHold has been replaced with MusicOnHoldStart and MusicOnHoldStop.
  - JabberEvent has been removed.
  - Hold is now in the core and will now raise Hold and Unhold events.
  - Join is now QueueCallerJoin.
  - Leave is now QueueCallerLeave.
  - Agentlogin/Agentlogoff is now AgentLogin/AgentLogoff, respectively.
  - ChannelUpdate has been removed.
  - Local channel optimization is now conveyed via LocalOptimizationBegin and LocalOptimizationEnd.
  - BridgeAction and BridgeExec have been removed.
  - BlindTransfer and AttendedTransfer events were added.
  - Dial is now DialBegin and DialEnd.
  - DTMF is now DTMFBegin and DTMFEnd.
  - Bridge has been replaced with BridgeCreate, BridgeEnter, BridgeLeave, and BridgeDestroy
  - The Masquerade event is no longer raised. The Rename event still exists, but only silly channel drivers (`chan_misdn`) will ever think of raising it, and you shouldn't be using `chan_misdn` anyway.
  - MusicOnHold has been replaced with MusicOnHoldStart and MusicOnHoldStop
  - AGIExec is now AGIExecStart and AGIExecEnd
  - AsyncAGI is now AsyncAGIStart, AsyncAGIExec, and AsyncAGIEnd
- The MCID AMI event now publishes a channel snapshot when available and its non-channel-snapshot parameters now use either the *MCallerID* or *MConnectedID* prefixes with *Subaddr**, *Name**, and *Num** suffixes instead of *CallerID* and *ConnectedID* to avoid confusion with similarly named parameters in the channel snapshot.
- The *Channel* key used in the AlarmClear, Alarm, and DNDState has been renamed *DAHDIChannel* since it does not convey an Asterisk channel name.
- All AMI events now contain a *SystemName* field, if available.
- Local channel information in events is now prefixed with *LocalOne* and *LocalTwo*. This replaces the suffix of '1' and '2' for the two halves of the Local channel. This affects the LocalBridge, LocalOptimizationBegin, and LocalOptimizationEnd events.
- The RTCPSent/RTCPReceived events have been significantly modified from previous versions. They now report all SR/RR packets sent/received, and have been restructured to better reflect the data sent in a SR/RR. In particular, the event structure now supports

multiple report blocks.

- The deprecated use of | (pipe) as a separator in the `channelvars` setting in *manager.conf* has been removed.
- The SIP SIPqualifypeer action now sends a response indicating it will qualify a peer once a peer has been found to qualify. Once the qualify has been completed it will now issue a SIPqualifypeerdone event.
- The AMI event Newexten field *Extension* is deprecated, and may be removed in a future release. Please use the common *Exten* field instead.
- The AMI events ParkedCall, ParkedCallTimeOut, ParkedCallGiveUp, and UnParkedCall have changed significantly in the new `res_parking` module.
  - The *Channel* and *From* headers are gone. For the channel that was parked or is coming out of parking, a *Parkee* channel snapshot is issued and it has a number of fields associated with it. The old *Channel* header relayed the same data as the new *ParkeeChannel* header.
  - The *From* field was ambiguous and changed meaning depending on the event. For most of these, it was the name of the channel that parked the call (the *Parker*). There is no longer a header that provides this channel name, however the *ParkerDialString* will contain a dialstring to redial the device that parked the call.
  - On UnParkedCall events, the *From* header would instead represent the channel responsible for retrieving the *parkee*. It receives a channel snapshot labeled *Retriever*. The *from* field is is replaced with *RetrieverChannel*.
  - Lastly, the *Exten* field has been replaced with *ParkingSpace*.
- The AMI event Parkinglot (response to Parkinglots command) in a similar fashion has changed the field names *StartExten* and *StopExten* to *StartSpace* and *StopSpace* respectively.
- The AMI Status response event to the AMI Status action replaces the *BridgedChannel* and *BridgedUniqueid* headers with the *BridgeID* header to indicate what bridge the channel is currently in.

CDR (Call Detail Records)

- Significant changes have been made to the behavior of CDRs. The CDR engine was effectively rewritten and built on the Stasis message bus. For a full definition of CDR behavior in Asterisk 12, please read the Asterisk 12 CDR Specification.
- CDRs will now be created between all participants in a bridge. For each pair of channels in a bridge, a CDR is created to represent the path of communication between those two endpoints. This lets an end user choose who to bill for what during bridge operations with multiple parties.
- The duration, billsec, start, answer, and end times now reflect the times associated with the current CDR for the channel, as opposed to a cumulative measurement of all CDRs for that channel.

CEL:

- The *Uniqueid* field for a channel is now a stable identifier, and will not change due to transfers, parking, etc.
- CEL has undergone significant rework in Asterisk 12, and is now built on the Stasis message bus. Please see the Asterisk 12 CEL Specification for more detailed information. A summary of the affected events is below:
  - `BRIDGE_START`, `BRIDGE_END`, `BRIDGE_UPDATE`, `3WAY_START`, `3WAY_END`, `CONF_ENTER`, `CONF_EXIT`, `CONF_START`, and `CONF_END` events have all been removed. These events have been replaced by `BRIDGE_ENTER`/ `BRIDGE_EXIT`.
  - `BLINDTRANSFER`/ `ATTENDEDTRANSFER` events now report the peer as NULL and additional information in the extra string field.

Dialplan:

- All channel and global variable names are evaluated in a case-sensitive manner. In previous versions of Asterisk, variables created and evaluated in the dialplan were evaluated case-insensitively, but built-in variables and variable evaluation done internally within Asterisk was done case-sensitively.
- Asterisk has always had code to ignore dash '-' characters that are not part of a character set in the dialplan extensions. The code now consistently ignores these characters when matching dialplan extensions.
- `BRIDGE_FEATURES` channel variable is now case sensitive for feature letter codes. Uppercase variants apply them to the calling party while lowercase variants apply them to the called party.

Features:

- The *features.conf* `[applicationmap] <FeatureName> ActivatedBy` option is no longer honored. The feature is always activated by the channel that has `DYNAMIC_FEATURES` defined on it when it enters the bridge. Use predial to set different values of `DYNAMIC_FEATURES` on the channels
- Executing a dynamic feature on the bridge peer in a multi-party bridge will execute it on all peers of the activating channel.
- There is no longer an explicit `features reload` CLI command. Features can still be reloaded using `module reload features`.
- It is no longer necessary (or possible) to define the ATXFER_NULL_TECH in *features.c* for `atxferdropcall=no` to work properly. This

option now just works.

Parking:

- Parking has been extracted from the Asterisk core as a loadable module, `res_parking`.
- Configuration is found in *res_parking.conf*. It is no longer supported in *features.conf*
- The arguments for the Park, ParkedCall, and ParkAndAnnounce applications have been modified significantly. See the application documents for specific details.
- Numerous changes to Parking related applications, AMI and CLI commands and internal inter-workings have been made. Please read the CHANGES file or the New in 12 page for the detailed list.

Security Events Framework:

- Security Event timestamps now use ISO 8601 formatted date/time instead of the "seconds-microseconds" format that it was using previously.

AGENT:

- The password option has been disabled, as the AgentLogin application no longer provides authentication.

AUDIOHOOK_INHERIT:

- Due to changes in the Asterisk core, this function is no longer needed to preserve a MixMonitor on a channel during transfer operations and dialplan execution. It is effectively obsolete.

CDR: (function)

- The `amaflags` and `accountcode` attributes for the CDR function are deprecated. Use the CHANNEL function instead to access these attributes.
- The `l` option has been removed. When reading a CDR attribute, the most recent record is always used. When writing a CDR attribute, all non-finalized CDRs are updated.
- The `r` option has been removed, for the same reason as the `l` option.
- The `s` option has been removed, as `LOCKED` semantics no longer exist in the CDR engine.

res_rtp_asterisk:

- ICE/STUN/TURN support in `res_rtp_asterisk` has been made optional. To enable them, an Asterisk-specific version of PJSIP needs to be installed. Tarballs are available from https://github.com/asterisk/pjproject/tags/.

Asterisk 12 Command Reference

This page is the top level page for the XML/JSON derived documentation in Asterisk 12:

- Dialplan applications and functions
- Manager actions and events
- AGI commands
- ARI HTTP requests and events
- Asterisk module configurations

Note that all documentation contained in this section is auto-generated. Requested changes to the documentation in this section should be made as patches to the Asterisk source through the Asterisk issue tracker.

Asterisk 12 AGI Commands

Asterisk 12 AGICommand_answer

ANSWER

Synopsis

Answer channel

Description

Answers channel if not already in answer state. Returns `-1` on channel failure, or `0` if successful.

Syntax

```
ANSWER
```

Arguments

See Also

- Asterisk 12 AGICommand_hangup

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 AGICommand_asyncagi break

ASYNCAGI BREAK

Synopsis

Interrupts Async AGI

Description

Interrupts expected flow of Async AGI commands and returns control to previous source (typically, the PBX dialplan).

Syntax

```
ASYNCAGI BREAK
```

Arguments

See Also

- Asterisk 12 AGICommand_hangup

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 AGICommand_channel status

CHANNEL STATUS

Synopsis

Returns status of the connected channel.

Description

Returns the status of the specified *channelname*. If no channel name is given then returns the status of the current channel.

Return values:

- `0` - Channel is down and available.
- `1` - Channel is down, but reserved.
- `2` - Channel is off hook.
- `3` - Digits (or equivalent) have been dialed.
- `4` - Line is ringing.
- `5` - Remote end is ringing.
- `6` - Line is up.
- `7` - Line is busy.

Syntax

```
CHANNEL STATUS CHANNELNAME
```

Arguments

- `channelname`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 AGICommand_control stream file

CONTROL STREAM FILE

Synopsis

Sends audio file on channel and allows the listener to control the stream.

Description

Send the given file, allowing playback to be controlled by the given digits, if any. Use double quotes for the digits if you wish none to be permitted. If offsetms is provided then the audio will seek to offsetms before play starts. Returns `0` if playback completes without a digit being pressed, or the ASCII numerical value of the digit if one was pressed, or `-1` on error or if the channel was disconnected. Returns the position where playback was terminated as endpos.

It sets the following channel variables upon completion:

- `CPLAYBACKSTATUS` - Contains the status of the attempt as a text string
  - SUCCESS
  - USERSTOPPED
  - REMOTESTOPPED
  - ERROR
- `CPLAYBACKOFFSET` - Contains the offset in ms into the file where playback was at when it stopped. `-1` is end of file.
- `CPLAYBACKSTOPKEY` - If the playback is stopped by the user this variable contains the key that was pressed.

Syntax

```
CONTROL STREAM FILE FILENAME ESCAPE_DIGITS SKIPMS FFCHAR REWCHR PAUSECHR
OFFSETMS
```

Arguments

- `filename` - The file extension must not be included in the filename.
- `escape_digits`
- `skipms`
- `ffchar` - Defaults to `*`
- `rewchr` - Defaults to `#`
- `pausechr`
- `offsetms` - Offset, in milliseconds, to start the audio playback

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 AGICommand_database del

DATABASE DEL

Synopsis

Removes database key/value

Description

Deletes an entry in the Asterisk database for a given *family* and *key*.

Returns `1` if successful, `0` otherwise.

Syntax

```
DATABASE DEL FAMILY KEY
```

Arguments

- `family`
- `key`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 AGICommand_database deltree

DATABASE DELTREE

Synopsis

Removes database keytree/value

Description

Deletes a *family* or specific *keytree* within a *family* in the Asterisk database.

Returns `1` if successful, `0` otherwise.

Syntax

```
DATABASE DELTREE FAMILY KEYTREE
```

Arguments

- `family`
- `keytree`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 AGICommand_database get

DATABASE GET

Synopsis

Gets database value

Description

Retrieves an entry in the Asterisk database for a given *family* and *key*.

Returns `0` if *key* is not set. Returns `1` if *key* is set and returns the variable in parenthesis.

Example return code: 200 result=1 (testvariable)

Syntax

```
DATABASE GET FAMILY KEY
```

Arguments

- `family`
- `key`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 AGICommand_database put

DATABASE PUT

Synopsis

Adds/updates database value

Description

Adds or updates an entry in the Asterisk database for a given *family*, *key*, and *value*.

Returns `1` if successful, `0` otherwise.

Syntax

```
DATABASE PUT FAMILY KEY VALUE
```

Arguments

- `family`
- `key`
- `value`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 AGICommand_exec

EXEC

Synopsis

Executes a given Application

Description

Executes *application* with given *options*.

Returns whatever the *application* returns, or `-2` on failure to find *application*.

Syntax

```
EXEC APPLICATION OPTIONS
```

Arguments

- `application`
- `options`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 AGICommand_get data

GET DATA

Synopsis

Prompts for DTMF on a channel

Description

Stream the given *file*, and receive DTMF data.

Returns the digits received from the channel at the other end.

Syntax

```
GET DATA FILE TIMEOUT MAXDIGITS
```

Arguments

- `file`
- `timeout`
- `maxdigits`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 AGICommand_get full variable

GET FULL VARIABLE

Synopsis

Evaluates a channel expression

Description

Returns `0` if *variablename* is not set or channel does not exist. Returns `1` if *variablename* is set and returns the variable in parenthesis. Understands complex variable names and builtin variables, unlike GET VARIABLE.

Example return code: 200 result=1 (testvariable)

Syntax

```
GET FULL VARIABLE VARIABLENAME CHANNEL NAME
```

Arguments

- `variablename`
- `channel name`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 AGICommand_get option

GET OPTION

Synopsis

Stream file, prompt for DTMF, with timeout.

Description

Behaves similar to STREAM FILE but used with a timeout option.

Syntax

```
GET OPTION FILENAME ESCAPE_DIGITS TIMEOUT
```

Arguments

- `filename`
- `escape_digits`
- `timeout`

See Also

- Asterisk 12 AGICommand_stream file

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 AGICommand_get variable

GET VARIABLE

Synopsis

Gets a channel variable.

Description

Returns `0` if *variablename* is not set. Returns `1` if *variablename* is set and returns the variable in parentheses.

Example return code: 200 result=1 (testvariable)

Syntax

```
GET VARIABLE VARIABLENAME
```

Arguments

- `variablename`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 AGICommand_gosub

GOSUB

Synopsis

Cause the channel to execute the specified dialplan subroutine.

Description

Cause the channel to execute the specified dialplan subroutine, returning to the dialplan with execution of a Return().

Syntax

```
GOSUB CONTEXT EXTENSION PRIORITY OPTIONAL-ARGUMENT
```

Arguments

- `context`
- `extension`
- `priority`
- `optional-argument`

See Also

- Asterisk 12 Application_GoSub

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 AGICommand_hangup

HANGUP

Synopsis

Hangup a channel.

Description

Hangs up the specified channel. If no channel name is given, hangs up the current channel

Syntax

```
HANGUP CHANNELNAME
```

Arguments

- `channelname`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 AGICommand_noop

NOOP

Synopsis

Does nothing.

Description

Does nothing.

Syntax

```
NOOP
```

Arguments

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 AGICommand_receive char

RECEIVE CHAR

Synopsis

Receives one character from channels supporting it.

Description

Receives a character of text on a channel. Most channels do not support the reception of text. Returns the decimal value of the character if one is received, or `0` if the channel does not support text reception. Returns `-1` only on error/hangup.

Syntax

```
RECEIVE CHAR TIMEOUT
```

Arguments

- `timeout` - The maximum time to wait for input in milliseconds, or `0` for infinite. Most channels

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 AGICommand_receive text

RECEIVE TEXT

Synopsis

Receives text from channels supporting it.

Description

Receives a string of text on a channel. Most channels do not support the reception of text. Returns `-1` for failure or `1` for success, and the string in parenthesis.

Syntax

```
RECEIVE TEXT TIMEOUT
```

Arguments

- `timeout` - The timeout to be the maximum time to wait for input in milliseconds, or `0` for infinite.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 AGICommand_record file

RECORD FILE

Synopsis

Records to a given file.

Description

Record to a file until a given dtmf digit in the sequence is received. Returns `-1` on hangup or error. The format will specify what kind of file will be recorded. The *timeout* is the maximum record time in milliseconds, or `-1` for no *timeout*. *offset samples* is optional, and, if provided, will seek to the offset without exceeding the end of the file. *silence* is the number of seconds of silence allowed before the function returns despite the lack of dtmf digits or reaching *timeout*. *sile nce* value must be preceded by `s=` and is also optional.

Syntax

```
RECORD FILE FILENAME FORMAT ESCAPE_DIGITS TIMEOUT OFFSET SAMPLES BEEP
S=SILENCE
```

Arguments

- `filename`
- `format`

- `escape_digits`
- `timeout`
- `offset samples`
- `BEEP`
- `s=silence`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 AGICommand_say alpha

SAY ALPHA

Synopsis

Says a given character string.

Description

Say a given character string, returning early if any of the given DTMF digits are received on the channel. Returns `0` if playback completes without a digit being pressed, or the ASCII numerical value of the digit if one was pressed or `-1` on error/hangup.

Syntax

```
SAY ALPHA NUMBER ESCAPE_DIGITS
```

Arguments

- `number`
- `escape_digits`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 AGICommand_say date

SAY DATE

Synopsis

Says a given date.

Description

Say a given date, returning early if any of the given DTMF digits are received on the channel. Returns `0` if playback completes without a digit being pressed, or the ASCII numerical value of the digit if one was pressed or `-1` on error/hangup.

Syntax

```
SAY DATE DATE ESCAPE_DIGITS
```

Arguments

- `date` - Is number of seconds elapsed since 00:00:00 on January 1, 1970. Coordinated Universal Time (UTC).
- `escape_digits`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 AGICommand_say datetime

SAY DATETIME

Synopsis

Says a given time as specified by the format given.

Description

Say a given time, returning early if any of the given DTMF digits are received on the channel. Returns `0` if playback completes without a digit being pressed, or the ASCII numerical value of the digit if one was pressed or `-1` on error/hangup.

Syntax

```
SAY DATETIME TIME ESCAPE_DIGITS FORMAT TIMEZONE
```

Arguments

- `time` - Is number of seconds elapsed since 00:00:00 on January 1, 1970, Coordinated Universal Time (UTC)
- `escape_digits`
- `format` - Is the format the time should be said in. See `voicemail.conf` (defaults to `ABdY 'digits/at' IMp`).
- `timezone` - Acceptable values can be found in `/usr/share/zoneinfo` Defaults to machine default.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 AGICommand_say digits

SAY DIGITS

Synopsis

Says a given digit string.

Description

Say a given digit string, returning early if any of the given DTMF digits are received on the channel. Returns `0` if playback completes without a digit being pressed, or the ASCII numerical value of the digit if one was pressed or `-1` on error/hangup.

Syntax

```
SAY DIGITS NUMBER ESCAPE_DIGITS
```

Arguments

- `number`
- `escape_digits`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 AGICommand_say number

SAY NUMBER

Synopsis

Says a given number.

Description

Say a given number, returning early if any of the given DTMF digits are received on the channel. Returns `0` if playback completes without a digit being pressed, or the ASCII numerical value of the digit if one was pressed or `-1` on error/hangup.

Syntax

```
SAY NUMBER NUMBER ESCAPE_DIGITS GENDER
```

Arguments

- `number`
- `escape_digits`
- `gender`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 AGICommand_say phonetic

SAY PHONETIC

Synopsis

Says a given character string with phonetics.

Description

Say a given character string with phonetics, returning early if any of the given DTMF digits are received on the channel. Returns `0` if playback completes without a digit pressed, the ASCII numerical value of the digit if one was pressed, or `-1` on error/hangup.

Syntax

```
SAY PHONETIC STRING ESCAPE_DIGITS
```

Arguments

- `string`
- `escape_digits`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 AGICommand_say time

SAY TIME

Synopsis

Says a given time.

Description

Say a given time, returning early if any of the given DTMF digits are received on the channel. Returns `0` if playback completes without a digit being pressed, or the ASCII numerical value of the digit if one was pressed or `-1` on error/hangup.

Syntax

```
SAY TIME TIME ESCAPE_DIGITS
```

Arguments

- `time` - Is number of seconds elapsed since 00:00:00 on January 1, 1970. Coordinated Universal Time (UTC).
- `escape_digits`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 AGICommand_send image

SEND IMAGE

Synopsis

Sends images to channels supporting it.

Description

Sends the given image on a channel. Most channels do not support the transmission of images. Returns `0` if image is sent, or if the channel does not support image transmission. Returns `-1` only on error/hangup. Image names should not include extensions.

Syntax

```
SEND IMAGE IMAGE
```

Arguments

- `image`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 AGICommand_send text

SEND TEXT

Synopsis

Sends text to channels supporting it.

Description

Sends the given text on a channel. Most channels do not support the transmission of text. Returns `0` if text is sent, or if the channel does not support text transmission. Returns `-1` only on error/hangup.

Syntax

```
SEND TEXT TEXT TO SEND
```

Arguments

- `text to send` - Text consisting of greater than one word should be placed in quotes since the command only accepts a single argument.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 AGICommand_set autohangup

SET AUTOHANGUP

Synopsis

Autohangup channel in some time.

Description

Cause the channel to automatically hangup at *time* seconds in the future. Of course it can be hungup before then as well. Setting to `0` will cause the autohangup feature to be disabled on this channel.

Syntax

```
SET AUTOHANGUP TIME
```

Arguments

- `time`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 AGICommand_set callerid

SET CALLERID

Synopsis

Sets callerid for the current channel.

Description

Changes the callerid of the current channel.

Syntax

```
SET CALLERID NUMBER
```

Arguments

- `number`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 AGICommand_set context

SET CONTEXT

Synopsis

Sets channel context.

Description

Sets the context for continuation upon exiting the application.

Syntax

```
SET CONTEXT DESIRED CONTEXT
```

Arguments

- `desired context`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 AGICommand_set extension

SET EXTENSION

Synopsis

Changes channel extension.

Description

Changes the extension for continuation upon exiting the application.

Syntax

```
SET EXTENSION NEW EXTENSION
```

Arguments

- `new extension`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 AGICommand_set music

SET MUSIC

Synopsis

Enable/Disable Music on hold generator

Description

Enables/Disables the music on hold generator. If *class* is not specified, then the `default` music on hold class will be used. This generator will be stopped automatically when playing a file.

Always returns `0`.

Syntax

```
SET MUSIC  CLASS
```

Arguments

- {{}}
    - {{}}
        -
            - `on`
    - {{}}
        -
            - `off`
- `class`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 AGICommand_set priority

SET PRIORITY

Synopsis

Set channel dialplan priority.

Description

Changes the priority for continuation upon exiting the application. The priority must be a valid priority or label.

Syntax

```
SET PRIORITY PRIORITY
```

Arguments

- `priority`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 AGICommand_set variable

SET VARIABLE

Synopsis

Sets a channel variable.

Description

Sets a variable to the current channel.

Syntax

```
SET VARIABLE VARIABLENAME VALUE
```

Arguments

- `variablename`
- `value`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 AGICommand_speech activate grammar

SPEECH ACTIVATE GRAMMAR

Synopsis

Activates a grammar.

Description

Activates the specified grammar on the speech object.

Syntax

```
SPEECH ACTIVATE GRAMMAR GRAMMAR NAME
```

Arguments

- `grammar name`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 AGICommand_speech create

SPEECH CREATE

Synopsis

Creates a speech object.

Description

Create a speech object to be used by the other Speech AGI commands.

Syntax

```
SPEECH CREATE ENGINE
```

Arguments

- `engine`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 AGICommand_speech deactivate grammar

SPEECH DEACTIVATE GRAMMAR

Synopsis

Deactivates a grammar.

Description

Deactivates the specified grammar on the speech object.

Syntax

```
SPEECH DEACTIVATE GRAMMAR GRAMMAR NAME
```

Arguments

- `grammar name`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 AGICommand_speech destroy

SPEECH DESTROY

Synopsis

Destroys a speech object.

Description

Destroy the speech object created by `SPEECH CREATE`.

Syntax

```
SPEECH DESTROY
```

Arguments

See Also

- Asterisk 12 AGICommand_speech create

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 AGICommand_speech load grammar

SPEECH LOAD GRAMMAR

Synopsis

Loads a grammar.

Description

Loads the specified grammar as the specified name.

Syntax

```
SPEECH LOAD GRAMMAR GRAMMAR NAME PATH TO GRAMMAR
```

Arguments

- `grammar name`
- `path to grammar`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 AGICommand_speech recognize

SPEECH RECOGNIZE

Synopsis

Recognizes speech.

Description

Plays back given *prompt* while listening for speech and dtmf.

Syntax

```
SPEECH RECOGNIZE PROMPT TIMEOUT OFFSET
```

Arguments

- `prompt`
- `timeout`
- `offset`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 AGICommand_speech set

SPEECH SET

Synopsis

Sets a speech engine setting.

Description

Set an engine-specific setting.

Syntax

```
SPEECH SET NAME VALUE
```

Arguments

- `name`
- `value`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 AGICommand_speech unload grammar

SPEECH UNLOAD GRAMMAR

Synopsis

Unloads a grammar.

Description

Unloads the specified grammar.

Syntax

```
SPEECH UNLOAD GRAMMAR GRAMMAR NAME
```

Arguments

- `grammar name`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 AGICommand_stream file

STREAM FILE

Synopsis

Sends audio file on channel.

Description

Send the given file, allowing playback to be interrupted by the given digits, if any. Returns `0` if playback completes without a digit being pressed, or the ASCII numerical value of the digit if one was pressed, or `-1` on error or if the channel was disconnected. If musiconhold is playing before calling stream file it will be automatically stopped and will not be restarted after completion.

It sets the following channel variables upon completion:

- PLAYBACKSTATUS - The status of the playback attempt as a text string.
  - SUCCESS
  - FAILED

Syntax

```
STREAM FILE FILENAME ESCAPE_DIGITS SAMPLE OFFSET
```

Arguments

- `filename` - File name to play. The file extension must not be included in the *filename*.
- `escape_digits` - Use double quotes for the digits if you wish none to be permitted.
- `sample offset` - If sample offset is provided then the audio will seek to sample offset before play starts.

See Also

- Asterisk 12 AGICommand_control stream file

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 AGICommand_tdd mode

TDD MODE

Synopsis

Toggles TDD mode (for the deaf).

Description

Enable/Disable TDD transmission/reception on a channel. Returns `1` if successful, or `0` if channel is not TDD-capable.

Syntax

```
TDD MODE BOOLEAN
```

Arguments

- `boolean`
    - `on`
    - `off`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 AGICommand_verbose

VERBOSE

Synopsis

Logs a message to the asterisk verbose log.

Description

Sends *message* to the console via verbose message system. *level* is the verbose level (1-4). Always returns `1`

Syntax

```
VERBOSE MESSAGE LEVEL
```

Arguments

- `message`
- `level`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 AGICommand_wait for digit

WAIT FOR DIGIT

Synopsis

Waits for a digit to be pressed.

Description

Waits up to *timeout* milliseconds for channel to receive a DTMF digit. Returns `-1` on channel failure, `0` if no digit is received in the timeout, or the numerical value of the ascii of the digit if one is received. Use `-1` for the *timeout* value if you desire the call to block indefinitely.

Syntax

```
WAIT FOR DIGIT TIMEOUT
```

Arguments

- `timeout`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 AMI Actions

Asterisk 12 ManagerAction_AbsoluteTimeout

AbsoluteTimeout

Synopsis

Set absolute timeout.

Description

Hangup a channel after a certain time. Acknowledges set time with `Timeout Set` message.

Syntax

```
Action: AbsoluteTimeout
ActionID: <value>
Channel: <value>
Timeout: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.
- `Channel` - Channel name to hangup.
- `Timeout` - Maximum duration of the call (sec).

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_AgentLogoff

AgentLogoff

Synopsis

Sets an agent as no longer logged in.

Description

Sets an agent as no longer logged in.

Syntax

```
Action: AgentLogoff
ActionID: <value>
Agent: <value>
Soft: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.
- `Agent` - Agent ID of the agent to log off.
- `Soft` - Set to `true` to not hangup existing calls.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_Agents

Agents

Synopsis

Lists agents and their status.

Description

Will list info about all defined agents.

Syntax

```
Action: Agents
ActionID: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.

See Also

- Asterisk 12 ManagerEvent_Agents
- Asterisk 12 ManagerEvent_AgentsComplete

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_AGI

AGI

Synopsis

Add an AGI command to execute by Async AGI.

Description

Add an AGI command to the execute queue of the channel in Async AGI.

Syntax

```
Action: AGI
ActionID: <value>
Channel: <value>
Command: <value>
CommandID: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.
- `Channel` - Channel that is currently in Async AGI.
- `Command` - Application to execute.
- `CommandID` - This will be sent back in CommandID header of AsyncAGI exec event notification.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_AOCMessage

AOCMessage

Synopsis

Generate an Advice of Charge message on a channel.

Description

Generates an AOC-D or AOC-E message on a channel.

Syntax

```
Action: AOCMessage
ActionID: <value>
Channel: <value>
ChannelPrefix: <value>
MsgType: <value>
ChargeType: <value>
UnitAmount(0): <value>
UnitType(0): <value>
CurrencyName: <value>
CurrencyAmount: <value>
CurrencyMultiplier: <value>
TotalType: <value>
AOCBillingId: <value>
ChargingAssociationId: <value>
ChargingAssociationNumber: <value>
ChargingAssociationPlan: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.
- `Channel` - Channel name to generate the AOC message on.
- `ChannelPrefix` - Partial channel prefix. By using this option one can match the beginning part of a channel name without having to put the entire name in. For example if a channel name is SIP/snom-00000001 and this value is set to SIP/snom, then that channel matches and the message will be sent. Note however that only the first matched channel has the message sent on it.
- `MsgType` - Defines what type of AOC message to create, AOC-D or AOC-E
    - `D`
    - `E`
- `ChargeType` - Defines what kind of charge this message represents.
    - `NA`
    - `FREE`
    - `Currency`
    - `Unit`
- `UnitAmount(0)` - This represents the amount of units charged. The ETSI AOC standard specifies that this value along with the optional UnitType value are entries in a list. To accommodate this these values take an index value starting at 0 which can be used to generate this list of unit entries. For Example, If two unit entires were required this could be achieved by setting the paramter UnitAmount(0)=1234 and UnitAmount(1)=5678. Note that UnitAmount at index 0 is required when ChargeType=Unit, all other entries in the list are optional.
- `UnitType(0)` - Defines the type of unit. ETSI AOC standard specifies this as an integer value between 1 and 16, but this value is left open to accept any positive integer. Like the UnitAmount parameter, this value represents a list entry and has an index parameter that starts at 0.
- `CurrencyName` - Specifies the currency's name. Note that this value is truncated after 10 characters.
- `CurrencyAmount` - Specifies the charge unit amount as a positive integer. This value is required when ChargeType==Currency.
- `CurrencyMultiplier` - Specifies the currency multiplier. This value is required when ChargeType==Currency.
    - `OneThousandth`
    - `OneHundredth`
    - `OneTenth`
    - `One`
    - `Ten`
    - `Hundred`
    - `Thousand`
- `TotalType` - Defines what kind of AOC-D total is represented.
    - `Total`
    - `SubTotal`
- `AOCBillingId` - Represents a billing ID associated with an AOC-D or AOC-E message. Note that only the first 3 items of the enum are valid AOC-D billing IDs
    - `Normal`

        * `ReverseCharge`
        * `CreditCard`
        * `CallFwdUnconditional`
        * `CallFwdBusy`
        * `CallFwdNoReply`
        * `CallDeflection`
        * `CallTransfer`
* `ChargingAssociationId` - Charging association identifier. This is optional for AOC-E and can be set to any value between -32768 and 32767
* `ChargingAssociationNumber` - Represents the charging association party number. This value is optional for AOC-E.
* `ChargingAssociationPlan` - Integer representing the charging plan associated with the ChargingAssociationNumber. The value is bits 7 through 1 of the Q.931 octet containing the type-of-number and numbering-plan-identification fields.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_Atxfer

Atxfer

Synopsis

Attended transfer.

Description

Attended transfer.

Syntax

```
Action: Atxfer
ActionID: <value>
Channel: <value>
Exten: <value>
Context: <value>
```

Arguments

* `ActionID` - ActionID for this transaction. Will be returned.
* `Channel` - Transferer's channel.
* `Exten` - Extension to transfer to.
* `Context` - Context to transfer to.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_BlindTransfer

BlindTransfer

Synopsis

Blind transfer channel(s) to the given destination

Description

Redirect all channels currently bridged to the specified channel to the specified destination.

Syntax

```
Action: BlindTransfer
Channel: <value>
Context: <value>
Exten: <value>
```

Arguments

- `Channel`
- `Context`
- `Exten`

See Also

- Asterisk 12 ManagerAction_Redirect

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_Bridge

Bridge

Synopsis

Bridge two channels already in the PBX.

Description

Bridge together two channels already in the PBX.

Syntax

```
Action: Bridge
ActionID: <value>
Channel1: <value>
Channel2: <value>
Tone: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.
- `Channel1` - Channel to Bridge to Channel2.
- `Channel2` - Channel to Bridge to Channel1.
- `Tone` - Play courtesy tone to Channel 2.
    - `no`
    - `Channel1`
    - `Channel2`
    - `Both`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_BridgeInfo

BridgeInfo

Synopsis

Get information about a bridge.

Description

Returns detailed information about a bridge and the channels in it.

Syntax

```
Action: BridgeInfo
ActionID: <value>
BridgeUniqueid: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.
- `BridgeUniqueid` - The unique ID of the bridge about which to retreive information.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_BridgeList

BridgeList

Synopsis

Get a list of bridges in the system.

Description

Returns a list of bridges, optionally filtering on a bridge type.

Syntax

```
Action: BridgeList
ActionID: <value>
BridgeType: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.

* `BridgeType` - Optional type for filtering the resulting list of bridges.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_Challenge

Challenge

Synopsis

Generate Challenge for MD5 Auth.

Description

Generate a challenge for MD5 authentication.

Syntax

```
Action: Challenge
ActionID: <value>
AuthType: <value>
```

Arguments

* `ActionID` - ActionID for this transaction. Will be returned.
* `AuthType` - Digest algorithm to use in the challenge. Valid values are:
    * `MD5`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_ChangeMonitor

ChangeMonitor

Synopsis

Change monitoring filename of a channel.

Description

This action may be used to change the file started by a previous 'Monitor' action.

Syntax

```
Action: ChangeMonitor
ActionID: <value>
Channel: <value>
File: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.
- `Channel` - Used to specify the channel to record.
- `File` - Is the new name of the file created in the monitor spool directory.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_Command

Command

Synopsis

Execute Asterisk CLI Command.

Description

Run a CLI command.

Syntax

```
Action: Command
ActionID: <value>
Command: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.
- `Command` - Asterisk CLI command to run.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_ConfbridgeKick

ConfbridgeKick

Synopsis

Kick a Confbridge user.

Description

Syntax

```
Action: ConfbridgeKick
ActionID: <value>
Conference: <value>
Channel: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.
- `Conference`
- `Channel`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_ConfbridgeList

ConfbridgeList

Synopsis

List participants in a conference.

Description

Lists all users in a particular ConfBridge conference. ConfbridgeList will follow as separate events, followed by a final event called ConfbridgeListComplete.

Syntax

```
Action: ConfbridgeList
ActionID: <value>
Conference: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.
- `Conference` - Conference number.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_ConfbridgeListRooms

ConfbridgeListRooms

Synopsis

List active conferences.

Description

Lists data about all active conferences. ConfbridgeListRooms will follow as separate events, followed by a final event called ConfbridgeListRoomsComplete.

Syntax

```
Action: ConfbridgeListRooms
ActionID: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_ConfbridgeLock

ConfbridgeLock

Synopsis

Lock a Confbridge conference.

Description

Syntax

```
Action: ConfbridgeLock
ActionID: <value>
Conference: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.
- `Conference`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_ConfbridgeMute

ConfbridgeMute

Synopsis

Mute a Confbridge user.

Description

Syntax

```
Action: ConfbridgeMute
ActionID: <value>
Conference: <value>
Channel: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.
- `Conference`
- `Channel`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_ConfbridgeSetSingleVideoSrc

ConfbridgeSetSingleVideoSrc

Synopsis

Set a conference user as the single video source distributed to all other participants.

Description

Syntax

```
Action: ConfbridgeSetSingleVideoSrc
ActionID: <value>
Conference: <value>
Channel: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.
- `Conference`
- `Channel`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_ConfbridgeStartRecord

ConfbridgeStartRecord

Synopsis

Start recording a Confbridge conference.

Description

Start recording a conference. If recording is already present an error will be returned. If RecordFile is not provided, the default record file specified in the conference's bridge profile will be used, if that is not present either a file will automatically be generated in the monitor directory.

Syntax

```
Action: ConfbridgeStartRecord
ActionID: <value>
Conference: <value>
[RecordFile:] <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.
- `Conference`
- `RecordFile`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_ConfbridgeStopRecord

ConfbridgeStopRecord

Synopsis

Stop recording a Confbridge conference.

Description

Syntax

```
Action: ConfbridgeStopRecord
ActionID: <value>
Conference: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.
- `Conference`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_ConfbridgeUnlock

ConfbridgeUnlock

Synopsis

Unlock a Confbridge conference.

Description

Syntax

```
Action: ConfbridgeUnlock
ActionID: <value>
Conference: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.
- `Conference`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_ConfbridgeUnmute

ConfbridgeUnmute

Synopsis

Unmute a Confbridge user.

Description

Syntax

```
Action: ConfbridgeUnmute
ActionID: <value>
Conference: <value>
Channel: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.
- `Conference`
- `Channel`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_ControlPlayback

ControlPlayback

Synopsis

Control the playback of a file being played to a channel.

Description

Control the operation of a media file being played back to a channel. Note that this AMI action does not initiate playback of media to channel, but rather controls the operation of a media operation that was already initiated on the channel.

> **Note**
> The `pause` and `restart` *Control* options will stop a playback operation if that operation was not initiated from the *ControlPlayback* application or the *control stream file* AGI command.

Syntax

```
Action: ControlPlayback
ActionID: <value>
Channel: <value>
Control: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.
- `Channel` - The name of the channel that currently has a file being played back to it.
- `Control`
  - `stop` - Stop the playback operation.
  - `forward` - Move the current position in the media forward. The amount of time that the stream moves forward is determined by the *skipms* value passed to the application that initiated the playback.

    > **Note**
    > The default skipms value is `3000` ms.

  - `reverse` - Move the current position in the media backward. The amount of time that the stream moves backward is determined by the *skipms* value passed to the application that initiated the playback.

    > **Note**
    > The default skipms value is `3000` ms.

  - `pause` - Pause/unpause the playback operation, if supported. If not supported, stop the playback.
  - `restart` - Restart the playback operation, if supported. If not supported, stop the playback.

See Also

- Asterisk 12 Application_Playback
- Asterisk 12 Application_ControlPlayback
- Asterisk 12 AGICommand_stream file
- Asterisk 12 AGICommand_control stream file

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_CoreSettings

CoreSettings

Synopsis

Show PBX core settings (version etc).

Description

Query for Core PBX settings.

Syntax

```
Action: CoreSettings
ActionID: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_CoreShowChannels

CoreShowChannels

Synopsis

List currently active channels.

Description

List currently defined channels and some information about them.

Syntax

```
Action: CoreShowChannels
ActionID: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_CoreStatus

CoreStatus

Synopsis

Show PBX core status variables.

Description

Query for Core PBX status.

Syntax

```
Action: CoreStatus
ActionID: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_CreateConfig

CreateConfig

Synopsis

Creates an empty file in the configuration directory.

Description

This action will create an empty file in the configuration directory. This action is intended to be used before an UpdateConfig action.

Syntax

```
Action: CreateConfig
ActionID: <value>
Filename: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.
- `Filename` - The configuration filename to create (e.g. `foo.conf`).

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_DAHDIDialOffhook

DAHDIDialOffhook

Synopsis

Dial over DAHDI channel while offhook.

Description

Generate DTMF control frames to the bridged peer.

Syntax

```
Action: DAHDIDialOffhook
ActionID: <value>
DAHDIChannel: <value>
Number: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.
- `DAHDIChannel` - DAHDI channel number to dial digits.
- `Number` - Digits to dial.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_DAHDIDNDoff

DAHDIDNDoff

Synopsis

Toggle DAHDI channel Do Not Disturb status OFF.

Description

Equivalent to the CLI command "dahdi set dnd `channel` off".

> **Note**
> Feature only supported by analog channels.

Syntax

```
Action: DAHDIDNDoff
ActionID: <value>
DAHDIChannel: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.
- `DAHDIChannel` - DAHDI channel number to set DND off.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_DAHDIDNDon

DAHDIDNDon

Synopsis

Toggle DAHDI channel Do Not Disturb status ON.

Description

Equivalent to the CLI command "dahdi set dnd `channel` on".

> **Note**
> Feature only supported by analog channels.

Syntax

```
Action: DAHDIDNDon
ActionID: <value>
DAHDIChannel: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.
- `DAHDIChannel` - DAHDI channel number to set DND on.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_DAHDIHangup

DAHDIHangup

Synopsis

Hangup DAHDI Channel.

Description

Simulate an on-hook event by the user connected to the channel.

> **Note**
> Valid only for analog channels.

Syntax

```
Action: DAHDIHangup
ActionID: <value>
DAHDIChannel: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.
- `DAHDIChannel` - DAHDI channel number to hangup.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_DAHDIRestart

DAHDIRestart

Synopsis

Fully Restart DAHDI channels (terminates calls).

Description

Equivalent to the CLI command "dahdi restart".

Syntax

```
Action: DAHDIRestart
ActionID: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_DAHDIShowChannels

DAHDIShowChannels

Synopsis

Show status of DAHDI channels.

Description

Similar to the CLI command "dahdi show channels".

Syntax

```
Action: DAHDIShowChannels
ActionID: <value>
DAHDIChannel: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.
- `DAHDIChannel` - Specify the specific channel number to show. Show all channels if zero or not present.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_DAHDITransfer

DAHDITransfer

Synopsis

Transfer DAHDI Channel.

Description

Simulate a flash hook event by the user connected to the channel.

> **Note**
> Valid only for analog channels.

Syntax

```
Action: DAHDITransfer
ActionID: <value>
DAHDIChannel: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.
- `DAHDIChannel` - DAHDI channel number to transfer.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_DataGet

DataGet

Synopsis

Retrieve the data api tree.

Description

Retrieve the data api tree.

Syntax

```
Action: DataGet
ActionID: <value>
Path: <value>
Search: <value>
Filter: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.
- `Path`
- `Search`
- `Filter`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_DBDel

DBDel

Synopsis

Delete DB entry.

Description

Syntax

```
Action: DBDel
ActionID: <value>
Family: <value>
Key: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.
- `Family`
- `Key`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_DBDelTree

DBDelTree

Synopsis

Delete DB Tree.

Description

Syntax

```
Action: DBDelTree
ActionID: <value>
Family: <value>
Key: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.
- `Family`
- `Key`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_DBGet

DBGet

Synopsis

Get DB Entry.

Description

Syntax

```
Action: DBGet
ActionID: <value>
Family: <value>
Key: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.
- `Family`
- `Key`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_DBPut

DBPut

Synopsis

Put DB entry.

Description

Syntax

```
Action: DBPut
ActionID: <value>
Family: <value>
Key: <value>
Val: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.
- `Family`
- `Key`
- `Val`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_Events

Events

Synopsis

Control Event Flow.

Description

Enable/Disable sending of events to this manager client.

Syntax

```
Action: Events
ActionID: <value>
EventMask: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.
- `EventMask`
    - on - If all events should be sent.
    - `off` - If no events should be sent.
    - `system,call,log,...` - To select which flags events should have to be sent.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_ExtensionState

ExtensionState

Synopsis

Check Extension Status.

Description

Report the extension state for given extension. If the extension has a hint, will use devicestate to check the status of the device connected to the extension.

Will return an `Extension Status` message. The response will include the hint for the extension and the status.

Syntax

```
Action: ExtensionState
ActionID: <value>
Exten: <value>
Context: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.
- `Exten` - Extension to check state on.
- `Context` - Context for extension.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_Filter

Filter

Synopsis

Dynamically add filters for the current manager session.

Description

The filters added are only used for the current session. Once the connection is closed the filters are removed.

This comand requires the system permission because this command can be used to create filters that may bypass filters defined in manager.conf

Syntax

```
Action: Filter
ActionID: <value>
Operation: <value>
Filter: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.
- `Operation`
    - Add - Add a filter.
- `Filter` - Filters can be whitelist or blacklist
  Example whitelist filter: "Event: Newchannel"
  Example blacklist filter: "!Channel: DAHDI.*"
  This filter option is used to whitelist or blacklist events per user to be reported with regular expressions and are allowed if both the regex matches and the user has read access as defined in manager.conf. Filters are assumed to be for whitelisting unless preceeded by an exclamation point, which marks it as being black. Evaluation of the filters is as follows:

- If no filters are configured all events are reported as normal.
- If there are white filters only: implied black all filter processed first, then white filters.
- If there are black filters only: implied white all filter processed first, then black filters.
- If there are both white and black filters: implied black all filter processed first, then white filters, and lastly black filters.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_FilterList

FilterList

Synopsis

Show current event filters for this session

Description

The filters displayed are for the current session. Only those filters defined in manager.conf will be present upon starting a new session.

Syntax

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_GetConfig

GetConfig

Synopsis

Retrieve configuration.

Description

This action will dump the contents of a configuration file by category and contents or optionally by specified category only.

Syntax

```
Action: GetConfig
ActionID: <value>
Filename: <value>
Category: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.
- `Filename` - Configuration filename (e.g. `foo.conf`).
- `Category` - Category in configuration file.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_GetConfigJSON

GetConfigJSON

Synopsis

Retrieve configuration (JSON format).

Description

This action will dump the contents of a configuration file by category and contents in JSON format. This only makes sense to be used using rawman over the HTTP interface.

Syntax

```
Action: GetConfigJSON
ActionID: <value>
Filename: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.
- `Filename` - Configuration filename (e.g. `foo.conf`).

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_Getvar

Getvar

Synopsis

Gets a channel variable.

Description

Get the value of a global or local channel variable.

> **Note**
> If a channel name is not provided then the variable is global.

Syntax

```
Action: Getvar
ActionID: <value>
Channel: <value>
Variable: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.
- `Channel` - Channel to read variable from.
- `Variable` - Variable name.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_Hangup

Hangup

Synopsis

Hangup channel.

Description

Hangup a channel.

Syntax

```
Action: Hangup
ActionID: <value>
Channel: <value>
Cause: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.

- `Channel` - The exact channel name to be hungup, or to use a regular expression, set this parameter to: /regex/
  Example exact channel: SIP/provider-0000012a
  Example regular expression: /^SIP/provider-.*$/
- `Cause` - Numeric hangup cause.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_IAXnetstats

IAXnetstats

Synopsis

Show IAX Netstats.

Description

Show IAX channels network statistics.

Syntax

```
Action: IAXnetstats
```

Arguments

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_IAXpeerlist

IAXpeerlist

Synopsis

List IAX Peers.

Description

List all the IAX peers.

Syntax

```
Action: IAXpeerlist
ActionID: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_IAXpeers

IAXpeers

Synopsis

List IAX peers.

Description

Syntax

```
Action: IAXpeers
ActionID: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_IAXregistry

IAXregistry

Synopsis

Show IAX registrations.

Description

Show IAX registrations.

Syntax

```
Action: IAXregistry
ActionID: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_JabberSend_res_jabber

JabberSend - [res_jabber]

Synopsis

Sends a message to a Jabber Client.

Description

Sends a message to a Jabber Client.

Syntax

```
Action: JabberSend
ActionID: <value>
Jabber: <value>
JID: <value>
Message: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.
- `Jabber` - Client or transport Asterisk uses to connect to JABBER.
- `JID` - XMPP/Jabber JID (Name) of recipient.
- `Message` - Message to be sent to the buddy.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_JabberSend_res_xmpp

JabberSend - [res_xmpp]

Synopsis

Sends a message to a Jabber Client.

Description

Sends a message to a Jabber Client.

Syntax

```
Action: JabberSend
ActionID: <value>
Jabber: <value>
JID: <value>
Message: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.
- `Jabber` - Client or transport Asterisk uses to connect to JABBER.

- `JID` - XMPP/Jabber JID (Name) of recipient.
- `Message` - Message to be sent to the buddy.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_ListCategories

ListCategories

Synopsis

List categories in configuration file.

Description

This action will dump the categories in a given file.

Syntax

```
Action: ListCategories
ActionID: <value>
Filename: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.
- `Filename` - Configuration filename (e.g. `foo.conf`).

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_ListCommands

ListCommands

Synopsis

List available manager commands.

Description

Returns the action name and synopsis for every action that is available to the user.

Syntax

```
Action: ListCommands
ActionID: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_LocalOptimizeAway

LocalOptimizeAway

Synopsis

Optimize away a local channel when possible.

Description

A local channel created with "/n" will not automatically optimize away. Calling this command on the local channel will clear that flag and allow it to optimize away if it's bridged or when it becomes bridged.

Syntax

```
Action: LocalOptimizeAway
ActionID: <value>
Channel: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.
- `Channel` - The channel name to optimize away.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_Login

Login

Synopsis

Login Manager.

Description

Login Manager.

Syntax

```
Action: Login
ActionID: <value>
Username: <value>
Secret: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.
- `Username` - Username to login with as specified in manager.conf.
- `Secret` - Secret to login with as specified in manager.conf.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_Logoff

Logoff

Synopsis

Logoff Manager.

Description

Logoff the current manager session.

Syntax

```
Action: Logoff
ActionID: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_MailboxCount

MailboxCount

Synopsis

Check Mailbox Message Count.

Description

Checks a voicemail account for new messages.

Returns number of urgent, new and old messages.

Message: Mailbox Message Count

Mailbox: *mailboxid*

UrgentMessages: *count*

NewMessages: *count*

OldMessages: *count*

Syntax

```
Action: MailboxCount
ActionID: <value>
Mailbox: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.
- `Mailbox` - Full mailbox ID *mailbox@vm-context*.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_MailboxStatus

MailboxStatus

Synopsis

Check mailbox.

Description

Checks a voicemail account for status.

Returns whether there are messages waiting.

Message: Mailbox Status.

Mailbox: *mailboxid*.

Waiting: `0` if messages waiting, `1` if no messages waiting.

Syntax

```
Action: MailboxStatus
ActionID: <value>
Mailbox: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.
- `Mailbox` - Full mailbox ID *mailbox@vm-context*.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_MeetmeList

MeetmeList

Synopsis

List participants in a conference.

Description

Lists all users in a particular MeetMe conference. MeetmeList will follow as separate events, followed by a final event called MeetmeListComplete.

Syntax

```
Action: MeetmeList
ActionID: <value>
[Conference:] <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.
- `Conference` - Conference number.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_MeetmeListRooms

MeetmeListRooms

Synopsis

List active conferences.

Description

Lists data about all active conferences. MeetmeListRooms will follow as separate events, followed by a final event called MeetmeListRoomsComplete.

Syntax

```
Action: MeetmeListRooms
ActionID: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_MeetmeMute

MeetmeMute

Synopsis

Mute a Meetme user.

Description

Syntax

```
Action: MeetmeMute
ActionID: <value>
Meetme: <value>
Usernum: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.
- `Meetme`
- `Usernum`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_MeetmeUnmute

MeetmeUnmute

Synopsis

Unmute a Meetme user.

Description

Syntax

```
Action: MeetmeUnmute
ActionID: <value>
Meetme: <value>
Usernum: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.
- `Meetme`
- `Usernum`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_MessageSend

MessageSend

Synopsis

Send an out of call message to an endpoint.

Description

Syntax

```
Action: MessageSend
ActionID: <value>
To: <value>
From: <value>
Body: <value>
Base64Body: <value>
Variable: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.
- `To` - The URI the message is to be sent to.

Technology: SIP

Specifying a prefix of `sip:` will send the message as a SIP MESSAGE request.

Technology: XMPP

Specifying a prefix of `xmpp:` will send the message as an XMPP chat message.

- `From` - A From URI for the message if needed for the message technology being used to send this message.

Technology: SIP

The `from` parameter can be a configured peer name or in the form of "display-name" <URI>.

Technology: XMPP

Specifying a prefix of `xmpp:` will specify the account defined in `xmpp.conf` to send the message from. Note that this field is required for XMPP messages.

- `Body` - The message body text. This must not contain any newlines as that conflicts with the AMI protocol.
- `Base64Body` - Text bodies requiring the use of newlines have to be base64 encoded in this field. Base64Body will be decoded before being sent out. Base64Body takes precedence over Body.
- `Variable` - Message variable to set, multiple Variable: headers are allowed. The header value is a comma separated list of name=value pairs.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_MixMonitor

MixMonitor

Synopsis

Record a call and mix the audio during the recording. Use of StopMixMonitor is required to guarantee the audio file is available for processing during dialplan execution.

Description

This action records the audio on the current channel to the specified file.

- `MIXMONITOR_FILENAME` - Will contain the filename used to record the mixed stream.

Syntax

```
Action: MixMonitor
ActionID: <value>
Channel: <value>
File: <value>
options: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.
- `Channel` - Used to specify the channel to record.
- `File` - Is the name of the file created in the monitor spool directory. Defaults to the same name as the channel (with slashes replaced with dashes). This argument is optional if you specify to record unidirectional audio with either the r(filename) or t(filename) options in the options field. If neither MIXMONITOR_FILENAME or this parameter is set, the mixed stream won't be recorded.
- `options` - Options that apply to the MixMonitor in the same way as they would apply if invoked from the MixMonitor application. For a list of available options, see the documentation for the mixmonitor application.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_MixMonitorMute

MixMonitorMute

Synopsis

Mute / unMute a Mixmonitor recording.

Description

This action may be used to mute a MixMonitor recording.

Syntax

```
Action: MixMonitorMute
ActionID: <value>
Channel: <value>
Direction: <value>
State: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.
- `Channel` - Used to specify the channel to mute.
- `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.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_ModuleCheck

ModuleCheck

Synopsis

Check if module is loaded.

Description

Checks if Asterisk module is loaded. Will return Success/Failure. For success returns, the module revision number is included.

Syntax

```
Action: ModuleCheck
Module: <value>
```

Arguments

- `Module` - Asterisk module name (not including extension).

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_ModuleLoad

ModuleLoad

Synopsis

Module management.

Description

Loads, unloads or reloads an Asterisk module in a running system.

Syntax

```
Action: ModuleLoad
ActionID: <value>
Module: <value>
LoadType: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.
- `Module` - Asterisk module name (including .so extension) or subsystem identifier:
  - `cdr`
  - `dnsmgr`
  - `extconfig`
  - `enum`
  - `acl`
  - `manager`
  - `http`
  - `logger`
  - `features`
  - `dsp`
  - `udptl`
  - `indications`
  - `cel`
  - `plc`
- `LoadType` - The operation to be done on module. Subsystem identifiers may only be reloaded.
  - `load`
  - `unload`
  - `reload`
    If no module is specified for a `reload` loadtype, all modules are reloaded.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_Monitor

Monitor

Synopsis

Monitor a channel.

Description

This action may be used to record the audio on a specified channel.

Syntax

```
Action: Monitor
ActionID: <value>
Channel: <value>
File: <value>
Format: <value>
Mix: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.
- `Channel` - Used to specify the channel to record.
- `File` - Is the name of the file created in the monitor spool directory. Defaults to the same name as the channel (with slashes replaced with dashes).
- `Format` - Is the audio recording format. Defaults to `wav`.
- `Mix` - Boolean parameter as to whether to mix the input and output channels together after the recording is finished.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_MuteAudio

MuteAudio

Synopsis

Mute an audio stream.

Description

Mute an incoming or outgoing audio stream on a channel.

Syntax

```
Action: MuteAudio
ActionID: <value>
Channel: <value>
Direction: <value>
State: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.
- `Channel` - The channel you want to mute.
- `Direction`

    - `in` - Set muting on inbound audio stream. (to the PBX)
    - `out` - Set muting on outbound audio stream. (from the PBX)
    - `all` - Set muting on inbound and outbound audio streams.
- `State`
    - `on` - Turn muting on.
    - `off` - Turn muting off.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_Originate

Originate

Synopsis

Originate a call.

Description

Generates an outgoing call to a *Extension*/*Context*/*Priority* or *Application*/*Data*

Syntax

```
Action: Originate
ActionID: <value>
Channel: <value>
Exten: <value>
Context: <value>
Priority: <value>
Application: <value>
Data: <value>
Timeout: <value>
CallerID: <value>
Variable: <value>
Account: <value>
EarlyMedia: <value>
Async: <value>
Codecs: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.
- `Channel` - Channel name to call.
- `Exten` - Extension to use (requires `Context` and `Priority`)
- `Context` - Context to use (requires `Exten` and `Priority`)
- `Priority` - Priority to use (requires `Exten` and `Context`)
- `Application` - Application to execute.
- `Data` - Data to use (requires `Application`).
- `Timeout` - How long to wait for call to be answered (in ms.).
- `CallerID` - Caller ID to be set on the outgoing channel.
- `Variable` - Channel variable to set, multiple Variable: headers are allowed.
- `Account` - Account code.

* `EarlyMedia` - Set to `true` to force call bridge on early media..
* `Async` - Set to `true` for fast origination.
* `Codecs` - Comma-separated list of codecs to use for this call.

See Also

* Asterisk 12 ManagerEvent_OriginateResponse

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_Park

Park

Synopsis

Park a channel.

Description

Park an arbitrary channel with optional arguments for specifying the parking lot used, how long the channel should remain parked, and what dial string to use as the parker if the call times out.

Syntax

```
Action: Park
ActionID: <value>
Channel: <value>
[TimeoutChannel:] <value>
[Timeout:] <value>
[Parkinglot:] <value>
```

Arguments

* `ActionID` - ActionID for this transaction. Will be returned.
* `Channel` - Channel name to park.
* `TimeoutChannel` - Channel name to use when constructing the dial string that will be dialed if the parked channel times out.
* `Timeout` - Overrides the timeout of the parking lot for this park action. Specified in milliseconds, but will be converted to seconds. Use a value of 0 to nullify the timeout.
* `Parkinglot` - The parking lot to use when parking the channel

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_ParkedCalls

ParkedCalls

Synopsis

List parked calls.

Description

List parked calls.

Syntax

```
Action: ParkedCalls
ActionID: <value>
ParkingLot: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.
- `ParkingLot` - If specified, only show parked calls from the parking lot with this name.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_Parkinglots

Parkinglots

Synopsis

Get a list of parking lots

Description

List all parking lots as a series of AMI events

Syntax

```
Action: Parkinglots
ActionID: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_PauseMonitor

PauseMonitor

Synopsis

Pause monitoring of a channel.

Description

This action may be used to temporarily stop the recording of a channel.

Syntax

```
Action: PauseMonitor
ActionID: <value>
Channel: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.
- `Channel` - Used to specify the channel to record.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_Ping

Ping

Synopsis

Keepalive command.

Description

A 'Ping' action will ellicit a 'Pong' response. Used to keep the manager connection open.

Syntax

```
Action: Ping
ActionID: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_PJSIPNotify

PJSIPNotify

Synopsis

Send a NOTIFY to an endpoint.

Description

Send a NOTIFY to an endpoint.

Parameters will be placed into the notify as SIP headers.

Syntax

```
Action: PJSIPNotify
ActionID: <value>
Endpoint: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.
- `Endpoint` - The endpoint to which to send the NOTIFY.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_PJSIPQualify

PJSIPQualify

Synopsis

Qualify a chan_pjsip endpoint.

Description

Qualify a chan_pjsip endpoint.

Syntax

```
Action: PJSIPQualify
ActionID: <value>
Endpoint: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.
- `Endpoint` - The endpoint you want to qualify.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_PJSIPUnregister

PJSIPUnregister

Synopsis

Unregister an outbound registration.

Description

Syntax

```
Action: PJSIPUnregister
ActionID: <value>
Registration: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.
- `Registration` - The outbound registration to unregister.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_PlayDTMF

PlayDTMF

Synopsis

Play DTMF signal on a specific channel.

Description

Plays a dtmf digit on the specified channel.

Syntax

```
Action: PlayDTMF
ActionID: <value>
Channel: <value>
Digit: <value>
[Duration:] <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.
- `Channel` - Channel name to send digit to.
- `Digit` - The DTMF digit to play.
- `Duration` - The duration, in milliseconds, of the digit to be played.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_PresenceState

PresenceState

Synopsis

Check Presence State

Description

Report the presence state for the given presence provider.

Will return a `Presence State` message. The response will include the presence state and, if set, a presence subtype and custom message.

Syntax

```
Action: PresenceState
ActionID: <value>
Provider: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.
- `Provider` - Presence Provider to check the state of

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_PRIShowSpans

PRIShowSpans

Synopsis

Show status of PRI spans.

Description

Similar to the CLI command "pri show spans".

Syntax

```
Action: PRIShowSpans
ActionID: <value>
Span: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.
- `Span` - Specify the specific span to show. Show all spans if zero or not present.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_QueueAdd

QueueAdd

Synopsis

Add interface to queue.

Description

Syntax

```
Action: QueueAdd
ActionID: <value>
Queue: <value>
Interface: <value>
Penalty: <value>
Paused: <value>
MemberName: <value>
StateInterface: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.
- `Queue`
- `Interface`
- `Penalty`
- `Paused`
- `MemberName`
- `StateInterface`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_QueueLog

QueueLog

Synopsis

Adds custom entry in queue_log.

Description

Syntax

```
Action: QueueLog
ActionID: <value>
Queue: <value>
Event: <value>
Uniqueid: <value>
Interface: <value>
Message: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.
- `Queue`
- `Event`
- `Uniqueid`
- `Interface`
- `Message`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_QueueMemberRingInUse

QueueMemberRingInUse

Synopsis

Set the ringinuse value for a queue member.

Description

Syntax

```
Action: QueueMemberRingInUse
ActionID: <value>
Interface: <value>
RingInUse: <value>
Queue: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.
- `Interface`
- `RingInUse`
- `Queue`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_QueuePause

QueuePause

Synopsis

Makes a queue member temporarily unavailable.

Description

Syntax

```
Action: QueuePause
ActionID: <value>
Interface: <value>
Paused: <value>
Queue: <value>
Reason: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.
- `Interface`
- `Paused`
- `Queue`
- `Reason`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_QueuePenalty

QueuePenalty

Synopsis

Set the penalty for a queue member.

Description

Syntax

```
Action: QueuePenalty
ActionID: <value>
Interface: <value>
Penalty: <value>
Queue: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.
- `Interface`
- `Penalty`
- `Queue`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_QueueReload

QueueReload

Synopsis

Reload a queue, queues, or any sub-section of a queue or queues.

Description

Syntax

```
Action: QueueReload
ActionID: <value>
Queue: <value>
Members: <value>
Rules: <value>
Parameters: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.
- `Queue`
- `Members`
    - `yes`
    - `no`
- `Rules`
    - `yes`
    - `no`
- `Parameters`
    - `yes`
    - `no`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_QueueRemove

QueueRemove

Synopsis

Remove interface from queue.

Description

Syntax

```
Action: QueueRemove
ActionID: <value>
Queue: <value>
Interface: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.
- `Queue`
- `Interface`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_QueueReset

QueueReset

Synopsis

Reset queue statistics.

Description

Syntax

```
Action: QueueReset
ActionID: <value>
Queue: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.
- `Queue`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_QueueRule

QueueRule

Synopsis

Queue Rules.

Description

Syntax

```
Action: QueueRule
ActionID: <value>
Rule: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.
- `Rule`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_Queues

Queues

Synopsis

Queues.

Description

Syntax

```
Action: Queues
```

Arguments

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_QueueStatus

QueueStatus

Synopsis

Show queue status.

Description

Syntax

```
Action: QueueStatus
ActionID: <value>
Queue: <value>
Member: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.
- `Queue`
- `Member`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_QueueSummary

QueueSummary

Synopsis

Show queue summary.

Description

Syntax

```
Action: QueueSummary
ActionID: <value>
Queue: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.
- `Queue`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_Redirect

Redirect

Synopsis

Redirect (transfer) a call.

Description

Redirect (transfer) a call.

Syntax

```
Action: Redirect
ActionID: <value>
Channel: <value>
ExtraChannel: <value>
Exten: <value>
ExtraExten: <value>
Context: <value>
ExtraContext: <value>
Priority: <value>
ExtraPriority: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.
- `Channel` - Channel to redirect.
- `ExtraChannel` - Second call leg to transfer (optional).
- `Exten` - Extension to transfer to.
- `ExtraExten` - Extension to transfer extrachannel to (optional).
- `Context` - Context to transfer to.
- `ExtraContext` - Context to transfer extrachannel to (optional).
- `Priority` - Priority to transfer to.
- `ExtraPriority` - Priority to transfer extrachannel to (optional).

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_Reload

Reload

Synopsis

Send a reload event.

Description

Send a reload event.

Syntax

```
Action: Reload
ActionID: <value>
Module: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.
- `Module` - Name of the module to reload.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_SendText

SendText

Synopsis

Send text message to channel.

Description

Sends A Text Message to a channel while in a call.

Syntax

```
Action: SendText
ActionID: <value>
Channel: <value>
Message: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.
- `Channel` - Channel to send message to.
- `Message` - Message to send.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_Setvar

Setvar

Synopsis

Set a channel variable.

Description

Set a global or local channel variable.

> **Note**
> If a channel name is not provided then the variable is global.

Syntax

```
Action: Setvar
ActionID: <value>
Channel: <value>
Variable: <value>
Value: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.
- `Channel` - Channel to set variable for.
- `Variable` - Variable name.
- `Value` - Variable value.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_ShowDialPlan

ShowDialPlan

Synopsis

Show dialplan contexts and extensions

Description

Show dialplan contexts and extensions. Be aware that showing the full dialplan may take a lot of capacity.

Syntax

```
Action: ShowDialPlan
ActionID: <value>
Extension: <value>
Context: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.
- `Extension` - Show a specific extension.
- `Context` - Show a specific context.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_SIPnotify

SIPnotify

Synopsis

Send a SIP notify.

Description

Sends a SIP Notify event.

All parameters for this event must be specified in the body of this request via multiple `Variable : name=value` sequences.

Syntax

```
Action: SIPnotify
ActionID: <value>
Channel: <value>
Variable: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.
- `Channel` - Peer to receive the notify.
- `Variable` - At least one variable pair must be specified. *name=value*

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_SIPpeers

SIPpeers

Synopsis

List SIP peers (text format).

Description

Lists SIP peers in text format with details on current status. `Peerlist` will follow as separate events, followed by a final event called `PeerlistComplete`.

Syntax

```
Action: SIPpeers
ActionID: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_SIPpeerstatus

SIPpeerstatus

Synopsis

Show the status of one or all of the sip peers.

Description

Retrieves the status of one or all of the sip peers. If no peer name is specified, status for all of the sip peers will be retrieved.

Syntax

```
Action: SIPpeerstatus
ActionID: <value>
[Peer:] <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.
- `Peer` - The peer name you want to check.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_SIPqualifypeer

SIPqualifypeer

Synopsis

Qualify SIP peers.

Description

Qualify a SIP peer.

Syntax

```
Action: SIPqualifypeer
ActionID: <value>
Peer: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.
- `Peer` - The peer name you want to qualify.

See Also

- Asterisk 12 ManagerEvent_SIPQualifyPeerDone

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_SIPshowpeer

SIPshowpeer

Synopsis

show SIP peer (text format).

Description

Show one SIP peer with details on current status.

Syntax

```
Action: SIPshowpeer
ActionID: <value>
Peer: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.
- `Peer` - The peer name you want to check.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_SIPshowregistry

SIPshowregistry

Synopsis

Show SIP registrations (text format).

Description

Lists all registration requests and status. Registrations will follow as separate events followed by a final event called `RegistrationsComplete`.

Syntax

```
Action: SIPshowregistry
ActionID: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_SKINNYdevices

SKINNYdevices

Synopsis

List SKINNY devices (text format).

Description

Lists Skinny devices in text format with details on current status. Devicelist will follow as separate events, followed by a final event called DevicelistComplete.

Syntax

```
Action: SKINNYdevices
ActionID: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_SKINNYlines

SKINNYlines

Synopsis

List SKINNY lines (text format).

Description

Lists Skinny lines in text format with details on current status. Linelist will follow as separate events, followed by a final event called LinelistComplete.

Syntax

```
Action: SKINNYlines
ActionID: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_SKINNYshowdevice

SKINNYshowdevice

Synopsis

Show SKINNY device (text format).

Description

Show one SKINNY device with details on current status.

Syntax

```
Action: SKINNYshowdevice
ActionID: <value>
Device: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.
- `Device` - The device name you want to check.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_SKINNYshowline

SKINNYshowline

Synopsis

Show SKINNY line (text format).

Description

Show one SKINNY line with details on current status.

Syntax

```
Action: SKINNYshowline
ActionID: <value>
Line: <value>
```

Arguments

* `ActionID` - ActionID for this transaction. Will be returned.
* `Line` - The line name you want to check.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_Status

Status

Synopsis

List channel status.

Description

Will return the status information of each channel along with the value for the specified channel variables.

Syntax

```
Action: Status
ActionID: <value>
Channel: <value>
Variables: <value>
```

Arguments

* `ActionID` - ActionID for this transaction. Will be returned.
* `Channel` - The name of the channel to query for status.
* `Variables` - Comma `,` separated list of variable to include.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_StopMixMonitor

StopMixMonitor

Synopsis

Stop recording a call through MixMonitor, and free the recording's file handle.

Description

This action stops the audio recording that was started with the `MixMonitor` action on the current channel.

Syntax

```
Action: StopMixMonitor
ActionID: <value>
Channel: <value>
[MixMonitorID:] <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.
- `Channel` - The name of the channel monitored.
- `MixMonitorID` - If a valid ID is provided, then this command will stop only that specific MixMonitor.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_StopMonitor

StopMonitor

Synopsis

Stop monitoring a channel.

Description

This action may be used to end a previously started 'Monitor' action.

Syntax

```
Action: StopMonitor
ActionID: <value>
Channel: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.
- `Channel` - The name of the channel monitored.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_UnpauseMonitor

UnpauseMonitor

Synopsis

Unpause monitoring of a channel.

Description

This action may be used to re-enable recording of a channel after calling PauseMonitor.

Syntax

```
Action: UnpauseMonitor
ActionID: <value>
Channel: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.
- `Channel` - Used to specify the channel to record.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_UpdateConfig

UpdateConfig

Synopsis

Update basic configuration.

Description

This action will modify, create, or delete configuration elements in Asterisk configuration files.

Syntax

```
Action: UpdateConfig
ActionID: <value>
SrcFilename: <value>
DstFilename: <value>
Reload: <value>
Action-XXXXXX: <value>
Cat-XXXXXX: <value>
Var-XXXXXX: <value>
Value-XXXXXX: <value>
Match-XXXXXX: <value>
Line-XXXXXX: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.
- `SrcFilename` - Configuration filename to read (e.g. `foo.conf`).
- `DstFilename` - Configuration filename to write (e.g. `foo.conf`)
- `Reload` - Whether or not a reload should take place (or name of specific module).
- `Action-XXXXXX` - Action to take.
  X's represent 6 digit number beginning with 000000.
  - `NewCat`
  - `RenameCat`

        - `DelCat`
        - `EmptyCat`
        - `Update`
        - `Delete`
        - `Append`
        - `Insert`
- `Cat-XXXXXX` - Category to operate on.
  X's represent 6 digit number beginning with 000000.
- `Var-XXXXXX` - Variable to work on.
  X's represent 6 digit number beginning with 000000.
- `Value-XXXXXX` - Value to work on.
  X's represent 6 digit number beginning with 000000.
- `Match-XXXXXX` - Extra match required to match line.
  X's represent 6 digit number beginning with 000000.
- `Line-XXXXXX` - Line in category to operate on (used with delete and insert actions).
  X's represent 6 digit number beginning with 000000.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_UserEvent

UserEvent

Synopsis

Send an arbitrary event.

Description

Send an event to manager sessions.

Syntax

```
Action: UserEvent
ActionID: <value>
UserEvent: <value>
Header1: <value>
HeaderN: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.
- `UserEvent` - Event string to send.
- `Header1` - Content1.
- `HeaderN` - ContentN.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_VoicemailRefresh

VoicemailRefresh

Synopsis

Tell Asterisk to poll mailboxes for a change

Description

Normally, MWI indicators are only sent when Asterisk itself changes a mailbox. With external programs that modify the content of a mailbox from outside the application, an option exists called `pollmailboxes` that will cause voicemail to continually scan all mailboxes on a system for changes. This can cause a large amount of load on a system. This command allows external applications to signal when a particular mailbox has changed, thus permitting external applications to modify mailboxes and MWI to work without introducing considerable CPU load.

If *Context* is not specified, all mailboxes on the system will be polled for changes. If *Context* is specified, but *Mailbox* is omitted, then all mailboxes within *Context* will be polled. Otherwise, only a single mailbox will be polled for changes.

Syntax

```
Action: VoicemailRefresh
ActionID: <value>
Context: <value>
Mailbox: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.
- `Context`
- `Mailbox`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_VoicemailUsersList

VoicemailUsersList

Synopsis

List All Voicemail User Information.

Description

Syntax

```
Action: VoicemailUsersList
ActionID: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerAction_WaitEvent

WaitEvent

Synopsis

Wait for an event to occur.

Description

This action will ellicit a `Success` response. Whenever a manager event is queued. Once WaitEvent has been called on an HTTP manager session, events will be generated and queued.

Syntax

```
Action: WaitEvent
ActionID: <value>
Timeout: <value>
```

Arguments

- `ActionID` - ActionID for this transaction. Will be returned.
- `Timeout` - Maximum time (in seconds) to wait for events, `-1` means forever.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 AMI Events

Asterisk 12 ManagerEvent_AgentCalled

AgentCalled

Synopsis

Raised when an queue member is notified of a caller in the queue.

Description

Syntax

```
Action:
Channel: <value>
ChannelState: <value>
ChannelStateDesc: <value>
CallerIDNum: <value>
CallerIDName: <value>
ConnectedLineNum: <value>
ConnectedLineName: <value>
AccountCode: <value>
Context: <value>
Exten: <value>
Priority: <value>
Uniqueid: <value>
DestChannel: <value>
DestChannelState: <value>
DestChannelStateDesc: <value>
DestCallerIDNum: <value>
DestCallerIDName: <value>
DestConnectedLineNum: <value>
DestConnectedLineName: <value>
DestAccountCode: <value>
DestContext: <value>
DestExten: <value>
DestPriority: <value>
DestUniqueid: <value>
Queue: <value>
MemberName: <value>
Interface: <value>
```

Arguments

- `Channel`
- `ChannelState` - A numeric code for the channel's current state, related to ChannelStateDesc
- `ChannelStateDesc`
    - `Down`
    - `Rsrvd`
    - `OffHook`
    - `Dialing`
    - `Ring`
    - `Ringing`
    - `Up`
    - `Busy`
    - `Dialing Offhook`
    - `Pre-ring`
    - `Unknown`
- `CallerIDNum`
- `CallerIDName`
- `ConnectedLineNum`
- `ConnectedLineName`
- `AccountCode`
- `Context`
- `Exten`
- `Priority`
- `Uniqueid`

* `DestChannel`
* `DestChannelState` - A numeric code for the channel's current state, related to DestChannelStateDesc
* `DestChannelStateDesc`
    * `Down`
    * `Rsrvd`
    * `OffHook`
    * `Dialing`
    * `Ring`
    * `Ringing`
    * `Up`
    * `Busy`
    * `Dialing Offhook`
    * `Pre-ring`
    * `Unknown`
* `DestCallerIDNum`
* `DestCallerIDName`
* `DestConnectedLineNum`
* `DestConnectedLineName`
* `DestAccountCode`
* `DestContext`
* `DestExten`
* `DestPriority`
* `DestUniqueid`
* `Queue` - The name of the queue.
* `MemberName` - The name of the queue member.
* `Interface` - The queue member's channel technology or location.

See Also

* Asterisk 12 ManagerEvent_AgentRingNoAnswer
* Asterisk 12 ManagerEvent_AgentComplete
* Asterisk 12 ManagerEvent_AgentConnect

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_AgentComplete

AgentComplete

Synopsis

Raised when a queue member has finished servicing a caller in the queue.

Description

Syntax

```
Action:
Channel: <value>
ChannelState: <value>
ChannelStateDesc: <value>
CallerIDNum: <value>
CallerIDName: <value>
ConnectedLineNum: <value>
ConnectedLineName: <value>
AccountCode: <value>
Context: <value>
Exten: <value>
Priority: <value>
Uniqueid: <value>
DestChannel: <value>
DestChannelState: <value>
DestChannelStateDesc: <value>
DestCallerIDNum: <value>
DestCallerIDName: <value>
DestConnectedLineNum: <value>
DestConnectedLineName: <value>
DestAccountCode: <value>
DestContext: <value>
DestExten: <value>
DestPriority: <value>
DestUniqueid: <value>
Queue: <value>
MemberName: <value>
Interface: <value>
HoldTime: <value>
TalkTime: <value>
Reason: <value>
```

Arguments

- `Channel`
- `ChannelState` - A numeric code for the channel's current state, related to ChannelStateDesc
- `ChannelStateDesc`
    - `Down`
    - `Rsrvd`
    - `OffHook`
    - `Dialing`
    - `Ring`
    - `Ringing`
    - `Up`
    - `Busy`
    - `Dialing Offhook`
    - `Pre-ring`
    - `Unknown`
- `CallerIDNum`
- `CallerIDName`
- `ConnectedLineNum`
- `ConnectedLineName`
- `AccountCode`

- Context
- Exten
- Priority
- Uniqueid
- DestChannel
- DestChannelState - A numeric code for the channel's current state, related to DestChannelStateDesc
- DestChannelStateDesc
    - Down
    - Rsrvd
    - OffHook
    - Dialing
    - Ring
    - Ringing
    - Up
    - Busy
    - Dialing Offhook
    - Pre-ring
    - Unknown
- DestCallerIDNum
- DestCallerIDName
- DestConnectedLineNum
- DestConnectedLineName
- DestAccountCode
- DestContext
- DestExten
- DestPriority
- DestUniqueid
- Queue - The name of the queue.
- MemberName - The name of the queue member.
- Interface - The queue member's channel technology or location.
- HoldTime - The time the channel was in the queue, expressed in seconds since 00:00, Jan 1, 1970 UTC.
- TalkTime - The time the queue member talked with the caller in the queue, expressed in seconds since 00:00, Jan 1, 1970 UTC.
- Reason
    - caller
    - agent
    - transfer

See Also

- Asterisk 12 ManagerEvent_AgentCalled
- Asterisk 12 ManagerEvent_AgentConnect

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_AgentConnect

AgentConnect

Synopsis

Raised when a queue member answers and is bridged to a caller in the queue.

Description

Syntax

```
Action:
Channel: <value>
ChannelState: <value>
ChannelStateDesc: <value>
CallerIDNum: <value>
CallerIDName: <value>
ConnectedLineNum: <value>
ConnectedLineName: <value>
AccountCode: <value>
Context: <value>
Exten: <value>
Priority: <value>
Uniqueid: <value>
DestChannel: <value>
DestChannelState: <value>
DestChannelStateDesc: <value>
DestCallerIDNum: <value>
DestCallerIDName: <value>
DestConnectedLineNum: <value>
DestConnectedLineName: <value>
DestAccountCode: <value>
DestContext: <value>
DestExten: <value>
DestPriority: <value>
DestUniqueid: <value>
Queue: <value>
MemberName: <value>
Interface: <value>
RingTime: <value>
HoldTime: <value>
```

Arguments

- `Channel`
- `ChannelState` - A numeric code for the channel's current state, related to ChannelStateDesc
- `ChannelStateDesc`
    - `Down`
    - `Rsrvd`
    - `OffHook`
    - `Dialing`
    - `Ring`
    - `Ringing`
    - `Up`
    - `Busy`
    - `Dialing Offhook`
    - `Pre-ring`
    - `Unknown`
- `CallerIDNum`
- `CallerIDName`
- `ConnectedLineNum`
- `ConnectedLineName`
- `AccountCode`
- `Context`

- `Exten`
- `Priority`
- `Uniqueid`
- `DestChannel`
- `DestChannelState` - A numeric code for the channel's current state, related to DestChannelStateDesc
- `DestChannelStateDesc`
  - `Down`
  - `Rsrvd`
  - `OffHook`
  - `Dialing`
  - `Ring`
  - `Ringing`
  - `Up`
  - `Busy`
  - `Dialing Offhook`
  - `Pre-ring`
  - `Unknown`
- `DestCallerIDNum`
- `DestCallerIDName`
- `DestConnectedLineNum`
- `DestConnectedLineName`
- `DestAccountCode`
- `DestContext`
- `DestExten`
- `DestPriority`
- `DestUniqueid`
- `Queue` - The name of the queue.
- `MemberName` - The name of the queue member.
- `Interface` - The queue member's channel technology or location.
- `RingTime` - The time the queue member was rung, expressed in seconds since 00:00, Jan 1, 1970 UTC.
- `HoldTime` - The time the channel was in the queue, expressed in seconds since 00:00, Jan 1, 1970 UTC.

See Also

- Asterisk 12 ManagerEvent_AgentCalled
- Asterisk 12 ManagerEvent_AgentComplete
- Asterisk 12 ManagerEvent_AgentDump

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_AgentDump

AgentDump

Synopsis

Raised when a queue member hangs up on a caller in the queue.

Description

Syntax

```
Action:
Channel: <value>
ChannelState: <value>
ChannelStateDesc: <value>
CallerIDNum: <value>
CallerIDName: <value>
ConnectedLineNum: <value>
ConnectedLineName: <value>
AccountCode: <value>
Context: <value>
Exten: <value>
Priority: <value>
Uniqueid: <value>
DestChannel: <value>
DestChannelState: <value>
DestChannelStateDesc: <value>
DestCallerIDNum: <value>
DestCallerIDName: <value>
DestConnectedLineNum: <value>
DestConnectedLineName: <value>
DestAccountCode: <value>
DestContext: <value>
DestExten: <value>
DestPriority: <value>
DestUniqueid: <value>
Queue: <value>
MemberName: <value>
Interface: <value>
```

Arguments

- `Channel`
- `ChannelState` - A numeric code for the channel's current state, related to ChannelStateDesc
- `ChannelStateDesc`
    - `Down`
    - `Rsrvd`
    - `OffHook`
    - `Dialing`
    - `Ring`
    - `Ringing`
    - `Up`
    - `Busy`
    - `Dialing Offhook`
    - `Pre-ring`
    - `Unknown`
- `CallerIDNum`
- `CallerIDName`
- `ConnectedLineNum`
- `ConnectedLineName`
- `AccountCode`
- `Context`
- `Exten`
- `Priority`
- `Uniqueid`

- `DestChannel`
- `DestChannelState` - A numeric code for the channel's current state, related to DestChannelStateDesc
- `DestChannelStateDesc`
    - `Down`
    - `Rsrvd`
    - `OffHook`
    - `Dialing`
    - `Ring`
    - `Ringing`
    - `Up`
    - `Busy`
    - `Dialing Offhook`
    - `Pre-ring`
    - `Unknown`
- `DestCallerIDNum`
- `DestCallerIDName`
- `DestConnectedLineNum`
- `DestConnectedLineName`
- `DestAccountCode`
- `DestContext`
- `DestExten`
- `DestPriority`
- `DestUniqueid`
- `Queue` - The name of the queue.
- `MemberName` - The name of the queue member.
- `Interface` - The queue member's channel technology or location.

See Also

- Asterisk 12 ManagerEvent_AgentCalled
- Asterisk 12 ManagerEvent_AgentConnect

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_AgentLogin

AgentLogin

Synopsis

Raised when an Agent has logged in.

Description

Syntax

```
Action:
Channel: <value>
ChannelState: <value>
ChannelStateDesc: <value>
CallerIDNum: <value>
CallerIDName: <value>
ConnectedLineNum: <value>
ConnectedLineName: <value>
AccountCode: <value>
Context: <value>
Exten: <value>
Priority: <value>
Uniqueid: <value>
Agent: <value>
```

Arguments

- `Channel`
- `ChannelState` - A numeric code for the channel's current state, related to ChannelStateDesc
- `ChannelStateDesc`
    - `Down`
    - `Rsrvd`
    - `OffHook`
    - `Dialing`
    - `Ring`
    - `Ringing`
    - `Up`
    - `Busy`
    - `Dialing Offhook`
    - `Pre-ring`
    - `Unknown`
- `CallerIDNum`
- `CallerIDName`
- `ConnectedLineNum`
- `ConnectedLineName`
- `AccountCode`
- `Context`
- `Exten`
- `Priority`
- `Uniqueid`
- `Agent` - Agent ID of the agent.

See Also

- Asterisk 12 Application_AgentLogin
- Asterisk 12 ManagerEvent_AgentLogoff

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_AgentLogoff

AgentLogoff

Synopsis

Raised when an Agent has logged off.

Description

Syntax

```
Action:
Agent: <value>
Logintime: <value>
```

Arguments

- `Agent` - Agent ID of the agent.
- `Logintime` - The number of seconds the agent was logged in.

See Also

- Asterisk 12 ManagerEvent_AgentLogin

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_AgentRingNoAnswer

AgentRingNoAnswer

Synopsis

Raised when a queue member is notified of a caller in the queue and fails to answer.

Description

Syntax

```
Action:
Channel: <value>
ChannelState: <value>
ChannelStateDesc: <value>
CallerIDNum: <value>
CallerIDName: <value>
ConnectedLineNum: <value>
ConnectedLineName: <value>
AccountCode: <value>
Context: <value>
Exten: <value>
Priority: <value>
Uniqueid: <value>
DestChannel: <value>
DestChannelState: <value>
DestChannelStateDesc: <value>
DestCallerIDNum: <value>
DestCallerIDName: <value>
DestConnectedLineNum: <value>
DestConnectedLineName: <value>
DestAccountCode: <value>
DestContext: <value>
DestExten: <value>
DestPriority: <value>
DestUniqueid: <value>
Queue: <value>
MemberName: <value>
Interface: <value>
RingTime: <value>
```

Arguments

- `Channel`
- `ChannelState` - A numeric code for the channel's current state, related to ChannelStateDesc
- `ChannelStateDesc`
    - `Down`
    - `Rsrvd`
    - `OffHook`
    - `Dialing`
    - `Ring`
    - `Ringing`
    - `Up`
    - `Busy`
    - `Dialing Offhook`
    - `Pre-ring`
    - `Unknown`
- `CallerIDNum`
- `CallerIDName`
- `ConnectedLineNum`
- `ConnectedLineName`
- `AccountCode`
- `Context`
- `Exten`

- `Priority`
- `Uniqueid`
- `DestChannel`
- `DestChannelState` - A numeric code for the channel's current state, related to DestChannelStateDesc
- `DestChannelStateDesc`
    - `Down`
    - `Rsrvd`
    - `OffHook`
    - `Dialing`
    - `Ring`
    - `Ringing`
    - `Up`
    - `Busy`
    - `Dialing Offhook`
    - `Pre-ring`
    - `Unknown`
- `DestCallerIDNum`
- `DestCallerIDName`
- `DestConnectedLineNum`
- `DestConnectedLineName`
- `DestAccountCode`
- `DestContext`
- `DestExten`
- `DestPriority`
- `DestUniqueid`
- `Queue` - The name of the queue.
- `MemberName` - The name of the queue member.
- `Interface` - The queue member's channel technology or location.
- `RingTime` - The time the queue member was rung, expressed in seconds since 00:00, Jan 1, 1970 UTC.

See Also

- Asterisk 12 ManagerEvent_AgentCalled

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_Agents

Agents

Synopsis

Response event in a series to the Agents AMI action containing information about a defined agent.

Description

The channel snapshot is present if the Status value is `AGENT_IDLE` or `AGENT_ONCALL`.

Syntax

```
Action:
Agent: <value>
Name: <value>
Status: <value>
TalkingToChan: <value>
CallStarted: <value>
LoggedInTime: <value>
Channel: <value>
ChannelState: <value>
ChannelStateDesc: <value>
CallerIDNum: <value>
CallerIDName: <value>
ConnectedLineNum: <value>
ConnectedLineName: <value>
AccountCode: <value>
Context: <value>
Exten: <value>
Priority: <value>
Uniqueid: <value>
ActionID: <value>
```

Arguments

- `Agent` - Agent ID of the agent.
- `Name` - User friendly name of the agent.
- `Status` - Current status of the agent.
  The valid values are:
  - `AGENT_LOGGEDOFF`
  - `AGENT_IDLE`
  - `AGENT_ONCALL`
- `TalkingToChan` - BRIDGEPEER value on agent channel.
  Present if Status value is `AGENT_ONCALL`.
- `CallStarted` - Epoche time when the agent started talking with the caller.
  Present if Status value is `AGENT_ONCALL`.
- `LoggedInTime` - Epoche time when the agent logged in.
  Present if Status value is `AGENT_IDLE` or `AGENT_ONCALL`.
- `Channel`
- `ChannelState` - A numeric code for the channel's current state, related to ChannelStateDesc
- `ChannelStateDesc`
  - `Down`
  - `Rsrvd`
  - `OffHook`
  - `Dialing`
  - `Ring`
  - `Ringing`
  - `Up`
  - `Busy`
  - `Dialing Offhook`
  - `Pre-ring`
  - `Unknown`
- `CallerIDNum`
- `CallerIDName`
- `ConnectedLineNum`
- `ConnectedLineName`

* `AccountCode`
* `Context`
* `Exten`
* `Priority`
* `Uniqueid`
* `ActionID` - ActionID for this transaction. Will be returned.

See Also

* Asterisk 12 ManagerAction_Agents

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_AgentsComplete

AgentsComplete

Synopsis

Final response event in a series of events to the Agents AMI action.

Description

Syntax

```
Action:
ActionID: <value>
```

Arguments

* `ActionID` - ActionID for this transaction. Will be returned.

See Also

* Asterisk 12 ManagerAction_Agents

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_AGIExecEnd

AGIExecEnd

Synopsis

Raised when a received AGI command completes processing.

Description

Syntax

```
Action:
Channel: <value>
ChannelState: <value>
ChannelStateDesc: <value>
CallerIDNum: <value>
CallerIDName: <value>
ConnectedLineNum: <value>
ConnectedLineName: <value>
AccountCode: <value>
Context: <value>
Exten: <value>
Priority: <value>
Uniqueid: <value>
Command: <value>
CommandId: <value>
ResultCode: <value>
Result: <value>
```

Arguments

- `Channel`
- `ChannelState` - A numeric code for the channel's current state, related to ChannelStateDesc
- `ChannelStateDesc`
    - `Down`
    - `Rsrvd`
    - `OffHook`
    - `Dialing`
    - `Ring`
    - `Ringing`
    - `Up`
    - `Busy`
    - `Dialing Offhook`
    - `Pre-ring`
    - `Unknown`
- `CallerIDNum`
- `CallerIDName`
- `ConnectedLineNum`
- `ConnectedLineName`
- `AccountCode`
- `Context`
- `Exten`
- `Priority`
- `Uniqueid`
- `Command` - The AGI command as received from the external source.
- `CommandId` - Random identification number assigned to the execution of this command.
- `ResultCode` - The numeric result code from AGI
- `Result` - The text result reason from AGI

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_AGIExecStart

AGIExecStart

Synopsis

Raised when a received AGI command starts processing.

Description

Syntax

```
Action:
Channel: <value>
ChannelState: <value>
ChannelStateDesc: <value>
CallerIDNum: <value>
CallerIDName: <value>
ConnectedLineNum: <value>
ConnectedLineName: <value>
AccountCode: <value>
Context: <value>
Exten: <value>
Priority: <value>
Uniqueid: <value>
Command: <value>
CommandId: <value>
```

Arguments

- `Channel`
- `ChannelState` - A numeric code for the channel's current state, related to ChannelStateDesc
- `ChannelStateDesc`
    - `Down`
    - `Rsrvd`
    - `OffHook`
    - `Dialing`
    - `Ring`
    - `Ringing`
    - `Up`
    - `Busy`
    - `Dialing Offhook`
    - `Pre-ring`
    - `Unknown`
- `CallerIDNum`
- `CallerIDName`
- `ConnectedLineNum`
- `ConnectedLineName`
- `AccountCode`
- `Context`
- `Exten`
- `Priority`
- `Uniqueid`
- `Command` - The AGI command as received from the external source.
- `CommandId` - Random identification number assigned to the execution of this command.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_Alarm

Alarm

Synopsis

Raised when an alarm is set on a DAHDI channel.

Description

Syntax

```
Action:
DAHDIChannel: <value>
Alarm: <value>
```

Arguments

- `DAHDIChannel` - The channel on which the alarm occurred.

> **Note**
> This is not an Asterisk channel identifier.

- `Alarm` - A textual description of the alarm that occurred.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_AlarmClear

AlarmClear

Synopsis

Raised when an alarm is cleared on a DAHDI channel.

Description

Syntax

```
Action:
DAHDIChannel: <value>
```

Arguments

- `DAHDIChannel` - The DAHDI channel on which the alarm was cleared.

> **Note**

This is not an Asterisk channel identifier.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_AOC-D

AOC-D

Synopsis

Raised when an Advice of Charge message is sent during a call.

Description

Syntax

```
Action: 
Channel: <value>
ChannelState: <value>
ChannelStateDesc: <value>
CallerIDNum: <value>
CallerIDName: <value>
ConnectedLineNum: <value>
ConnectedLineName: <value>
AccountCode: <value>
Context: <value>
Exten: <value>
Priority: <value>
Uniqueid: <value>
Charge: <value>
Type: <value>
BillingID: <value>
TotalType: <value>
Currency: <value>
Name: <value>
Cost: <value>
Multiplier: <value>
Units: <value>
NumberOf: <value>
TypeOf: <value>
```

Arguments

- `Channel`
- `ChannelState` - A numeric code for the channel's current state, related to ChannelStateDesc
- `ChannelStateDesc`
  - `Down`
  - `Rsrvd`
  - `OffHook`
  - `Dialing`
  - `Ring`

    - Ringing
    - Up
    - Busy
    - Dialing Offhook
    - Pre-ring
    - Unknown
- CallerIDNum
- CallerIDName
- ConnectedLineNum
- ConnectedLineName
- AccountCode
- Context
- Exten
- Priority
- Uniqueid
- Charge
- Type
    - NotAvailable
    - Free
    - Currency
    - Units
- BillingID
    - Normal
    - Reverse
    - CreditCard
    - CallForwardingUnconditional
    - CallForwardingBusy
    - CallForwardingNoReply
    - CallDeflection
    - CallTransfer
    - NotAvailable
- TotalType
    - SubTotal
    - Total
- Currency
- Name
- Cost
- Multiplier
    - 1/1000
    - 1/100
    - 1/10
    - 1
    - 10
    - 100
    - 1000
- Units
- NumberOf
- TypeOf

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_AOC-E

AOC-E

Synopsis

Raised when an Advice of Charge message is sent at the end of a call.

Description

Syntax

```
Action:
Channel: <value>
ChannelState: <value>
ChannelStateDesc: <value>
CallerIDNum: <value>
CallerIDName: <value>
ConnectedLineNum: <value>
ConnectedLineName: <value>
AccountCode: <value>
Context: <value>
Exten: <value>
Priority: <value>
Uniqueid: <value>
ChargingAssociation: <value>
Number: <value>
Plan: <value>
ID: <value>
Charge: <value>
Type: <value>
BillingID: <value>
TotalType: <value>
Currency: <value>
Name: <value>
Cost: <value>
Multiplier: <value>
Units: <value>
NumberOf: <value>
TypeOf: <value>
```

Arguments

- `Channel`
- `ChannelState` - A numeric code for the channel's current state, related to ChannelStateDesc
- `ChannelStateDesc`
    - `Down`
    - `Rsrvd`
    - `OffHook`
    - `Dialing`
    - `Ring`
    - `Ringing`
    - `Up`
    - `Busy`
    - `Dialing Offhook`
    - `Pre-ring`
    - `Unknown`
- `CallerIDNum`
- `CallerIDName`
- `ConnectedLineNum`

- ConnectedLineName
- AccountCode
- Context
- Exten
- Priority
- Uniqueid
- ChargingAssociation
- Number
- Plan
- ID
- Charge
- Type
    - NotAvailable
    - Free
    - Currency
    - Units
- BillingID
    - Normal
    - Reverse
    - CreditCard
    - CallForwardingUnconditional
    - CallForwardingBusy
    - CallForwardingNoReply
    - CallDeflection
    - CallTransfer
    - NotAvailable
- TotalType
    - SubTotal
    - Total
- Currency
- Name
- Cost
- Multiplier
    - 1/1000
    - 1/100
    - 1/10
    - 1
    - 10
    - 100
    - 1000
- Units
- NumberOf
- TypeOf

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_AOC-S

AOC-S

Synopsis

Raised when an Advice of Charge message is sent at the beginning of a call.

Description

Syntax

```
Action:
Channel: <value>
ChannelState: <value>
ChannelStateDesc: <value>
CallerIDNum: <value>
CallerIDName: <value>
ConnectedLineNum: <value>
ConnectedLineName: <value>
AccountCode: <value>
Context: <value>
Exten: <value>
Priority: <value>
Uniqueid: <value>
Chargeable: <value>
RateType: <value>
Currency: <value>
Name: <value>
Cost: <value>
Multiplier: <value>
ChargingType: <value>
StepFunction: <value>
Granularity: <value>
Length: <value>
Scale: <value>
Unit: <value>
SpecialCode: <value>
```

Arguments

- `Channel`
- `ChannelState` - A numeric code for the channel's current state, related to ChannelStateDesc
- `ChannelStateDesc`
    - `Down`
    - `Rsrvd`
    - `OffHook`
    - `Dialing`
    - `Ring`
    - `Ringing`
    - `Up`
    - `Busy`
    - `Dialing Offhook`
    - `Pre-ring`
    - `Unknown`
- `CallerIDNum`
- `CallerIDName`
- `ConnectedLineNum`
- `ConnectedLineName`
- `AccountCode`
- `Context`
- `Exten`
- `Priority`
- `Uniqueid`
- `Chargeable`
- `RateType`

        - NotAvailable
        - Free
        - FreeFromBeginning
        - Duration
        - Flag
        - Volume
        - SpecialCode
    - Currency
    - Name
    - Cost
    - Multiplier
        - 1/1000
        - 1/100
        - 1/10
        - 1
        - 10
        - 100
        - 1000
    - ChargingType
    - StepFunction
    - Granularity
    - Length
    - Scale
    - Unit
        - Octect
        - Segment
        - Message
    - SpecialCode

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_AsyncAGIEnd

AsyncAGIEnd

Synopsis

Raised when a channel stops AsyncAGI command processing.

Description

Syntax

```
Action:
Channel: <value>
ChannelState: <value>
ChannelStateDesc: <value>
CallerIDNum: <value>
CallerIDName: <value>
ConnectedLineNum: <value>
ConnectedLineName: <value>
AccountCode: <value>
Context: <value>
Exten: <value>
Priority: <value>
Uniqueid: <value>
```

Arguments

- `Channel`
- `ChannelState` - A numeric code for the channel's current state, related to ChannelStateDesc
- `ChannelStateDesc`
    - `Down`
    - `Rsrvd`
    - `OffHook`
    - `Dialing`
    - `Ring`
    - `Ringing`
    - `Up`
    - `Busy`
    - `Dialing Offhook`
    - `Pre-ring`
    - `Unknown`
- `CallerIDNum`
- `CallerIDName`
- `ConnectedLineNum`
- `ConnectedLineName`
- `AccountCode`
- `Context`
- `Exten`
- `Priority`
- `Uniqueid`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_AsyncAGIExec

AsyncAGIExec

Synopsis

Raised when AsyncAGI completes an AGI command.

Description

Syntax

```
Action:
Channel: <value>
ChannelState: <value>
ChannelStateDesc: <value>
CallerIDNum: <value>
CallerIDName: <value>
ConnectedLineNum: <value>
ConnectedLineName: <value>
AccountCode: <value>
Context: <value>
Exten: <value>
Priority: <value>
Uniqueid: <value>
[CommandID:] <value>
Result: <value>
```

Arguments

- `Channel`
- `ChannelState` - A numeric code for the channel's current state, related to ChannelStateDesc
- `ChannelStateDesc`
    - `Down`
    - `Rsrvd`
    - `OffHook`
    - `Dialing`
    - `Ring`
    - `Ringing`
    - `Up`
    - `Busy`
    - `Dialing Offhook`
    - `Pre-ring`
    - `Unknown`
- `CallerIDNum`
- `CallerIDName`
- `ConnectedLineNum`
- `ConnectedLineName`
- `AccountCode`
- `Context`
- `Exten`
- `Priority`
- `Uniqueid`
- `CommandID` - Optional command ID sent by the AsyncAGI server to identify the command.
- `Result` - URL encoded result string from the executed AGI command.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_AsyncAGIStart

AsyncAGIStart

Synopsis

Raised when a channel starts AsyncAGI command processing.

Description

Syntax

```
Action:
Channel: <value>
ChannelState: <value>
ChannelStateDesc: <value>
CallerIDNum: <value>
CallerIDName: <value>
ConnectedLineNum: <value>
ConnectedLineName: <value>
AccountCode: <value>
Context: <value>
Exten: <value>
Priority: <value>
Uniqueid: <value>
Env: <value>
```

Arguments

- `Channel`
- `ChannelState` - A numeric code for the channel's current state, related to ChannelStateDesc
- `ChannelStateDesc`
    - `Down`
    - `Rsrvd`
    - `OffHook`
    - `Dialing`
    - `Ring`
    - `Ringing`
    - `Up`
    - `Busy`
    - `Dialing Offhook`
    - `Pre-ring`
    - `Unknown`
- `CallerIDNum`
- `CallerIDName`
- `ConnectedLineNum`
- `ConnectedLineName`
- `AccountCode`
- `Context`
- `Exten`
- `Priority`
- `Uniqueid`
- `Env` - URL encoded string read from the AsyncAGI server.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_AttendedTransfer

AttendedTransfer

Synopsis

Raised when an attended transfer is complete.

Description

The headers in this event attempt to describe all the major details of the attended transfer. The two transferer channels and the two bridges are determined based on their chronological establishment. So consider that Alice calls Bob, and then Alice transfers the call to Voicemail. The transferer and bridge headers would be arranged as follows:

OrigTransfererChannel: Alice's channel in the bridge with Bob.

OrigBridgeUniqueid: The bridge between Alice and Bob.

SecondTransfererChannel: Alice's channel that called Voicemail.

SecondBridgeUniqueid: Not present, since a call to Voicemail has no bridge.

Now consider if the order were reversed; instead of having Alice call Bob and transfer him to Voicemail, Alice instead calls her Voicemail and transfers that to Bob. The transferer and bridge headers would be arranged as follows:

OrigTransfererChannel: Alice's channel that called Voicemail.

OrigBridgeUniqueid: Not present, since a call to Voicemail has no bridge.

SecondTransfererChannel: Alice's channel in the bridge with Bob.

SecondBridgeUniqueid: The bridge between Alice and Bob.

Syntax

```
Action:
Result: <value>
OrigTransfererChannel: <value>
OrigTransfererChannelState: <value>
OrigTransfererChannelStateDesc: <value>
OrigTransfererCallerIDNum: <value>
OrigTransfererCallerIDName: <value>
OrigTransfererConnectedLineNum: <value>
OrigTransfererConnectedLineName: <value>
OrigTransfererAccountCode: <value>
OrigTransfererContext: <value>
OrigTransfererExten: <value>
OrigTransfererPriority: <value>
OrigTransfererUniqueid: <value>
OrigBridgeUniqueid: <value>
OrigBridgeType: <value>
OrigBridgeTechnology: <value>
OrigBridgeNumChannels: <value>
SecondTransfererChannel: <value>
```

```
SecondTransfererChannelState: <value>
SecondTransfererChannelStateDesc: <value>
SecondTransfererCallerIDNum: <value>
SecondTransfererCallerIDName: <value>
SecondTransfererConnectedLineNum: <value>
SecondTransfererConnectedLineName: <value>
SecondTransfererAccountCode: <value>
SecondTransfererContext: <value>
SecondTransfererExten: <value>
SecondTransfererPriority: <value>
SecondTransfererUniqueid: <value>
SecondBridgeUniqueid: <value>
SecondBridgeType: <value>
SecondBridgeTechnology: <value>
SecondBridgeNumChannels: <value>
DestType: <value>
DestBridgeUniqueid: <value>
DestApp: <value>
LocalOneChannel: <value>
LocalOneChannelState: <value>
LocalOneChannelStateDesc: <value>
LocalOneCallerIDNum: <value>
LocalOneCallerIDName: <value>
LocalOneConnectedLineNum: <value>
LocalOneConnectedLineName: <value>
LocalOneAccountCode: <value>
LocalOneContext: <value>
LocalOneExten: <value>
LocalOnePriority: <value>
LocalOneUniqueid: <value>
LocalTwoChannel: <value>
LocalTwoChannelState: <value>
LocalTwoChannelStateDesc: <value>
LocalTwoCallerIDNum: <value>
LocalTwoCallerIDName: <value>
LocalTwoConnectedLineNum: <value>
LocalTwoConnectedLineName: <value>
LocalTwoAccountCode: <value>
LocalTwoContext: <value>
LocalTwoExten: <value>
LocalTwoPriority: <value>
```

```
LocalTwoUniqueid: <value>
DestTransfererChannel: <value>
```

Arguments

- `Result` - Indicates if the transfer was successful or if it failed.
    - `Fail` - An internal error occurred.
    - `Invalid` - Invalid configuration for transfer (e.g. Not bridged)
    - `Not Permitted` - Bridge does not permit transfers
    - `Success` - Transfer completed successfully

    > **Note**
    > A result of `Success` does not necessarily mean that a target was succesfully contacted. It means that a party was succesfully placed into the dialplan at the expected location.

- `OrigTransfererChannel`
- `OrigTransfererChannelState` - A numeric code for the channel's current state, related to OrigTransfererChannelStateDesc
- `OrigTransfererChannelStateDesc`
    - `Down`
    - `Rsrvd`
    - `OffHook`
    - `Dialing`
    - `Ring`
    - `Ringing`
    - `Up`
    - `Busy`
    - `Dialing Offhook`
    - `Pre-ring`
    - `Unknown`
- `OrigTransfererCallerIDNum`
- `OrigTransfererCallerIDName`
- `OrigTransfererConnectedLineNum`
- `OrigTransfererConnectedLineName`
- `OrigTransfererAccountCode`
- `OrigTransfererContext`
- `OrigTransfererExten`
- `OrigTransfererPriority`
- `OrigTransfererUniqueid`
- `OrigBridgeUniqueid`
- `OrigBridgeType` - The type of bridge
- `OrigBridgeTechnology` - Technology in use by the bridge
- `OrigBridgeNumChannels` - Number of channels in the bridge
- `SecondTransfererChannel`
- `SecondTransfererChannelState` - A numeric code for the channel's current state, related to SecondTransfererChannelStateDesc
- `SecondTransfererChannelStateDesc`
    - `Down`
    - `Rsrvd`
    - `OffHook`
    - `Dialing`
    - `Ring`
    - `Ringing`
    - `Up`
    - `Busy`
    - `Dialing Offhook`

        - `Pre-ring`
        - `Unknown`
- `SecondTransfererCallerIDNum`
- `SecondTransfererCallerIDName`
- `SecondTransfererConnectedLineNum`
- `SecondTransfererConnectedLineName`
- `SecondTransfererAccountCode`
- `SecondTransfererContext`
- `SecondTransfererExten`
- `SecondTransfererPriority`
- `SecondTransfererUniqueid`
- `SecondBridgeUniqueid`
- `SecondBridgeType` - The type of bridge
- `SecondBridgeTechnology` - Technology in use by the bridge
- `SecondBridgeNumChannels` - Number of channels in the bridge
- `DestType` - Indicates the method by which the attended transfer completed.
    - `Bridge` - The transfer was accomplished by merging two bridges into one.
    - `App` - The transfer was accomplished by having a channel or bridge run a dialplan application.
    - `Link` - The transfer was accomplished by linking two bridges together using a local channel pair.
    - `Threeway` - The transfer was accomplished by placing all parties into a threeway call.
    - `Fail` - The transfer failed.
- `DestBridgeUniqueid` - Indicates the surviving bridge when bridges were merged to complete the transfer

> **Note**
> This header is only present when *DestType* is `Bridge` or `Threeway`

- `DestApp` - Indicates the application that is running when the transfer completes

> **Note**
> This header is only present when *DestType* is `App`

- `LocalOneChannel`
- `LocalOneChannelState` - A numeric code for the channel's current state, related to LocalOneChannelStateDesc
- `LocalOneChannelStateDesc`
    - `Down`
    - `Rsrvd`
    - `OffHook`
    - `Dialing`
    - `Ring`
    - `Ringing`
    - `Up`
    - `Busy`
    - `Dialing Offhook`
    - `Pre-ring`
    - `Unknown`
- `LocalOneCallerIDNum`
- `LocalOneCallerIDName`
- `LocalOneConnectedLineNum`
- `LocalOneConnectedLineName`
- `LocalOneAccountCode`
- `LocalOneContext`
- `LocalOneExten`
- `LocalOnePriority`
- `LocalOneUniqueid`
- `LocalTwoChannel`
- `LocalTwoChannelState` - A numeric code for the channel's current state, related to LocalTwoChannelStateDesc
- `LocalTwoChannelStateDesc`

    - `Down`
    - `Rsrvd`
    - `OffHook`
    - `Dialing`
    - `Ring`
    - `Ringing`
    - `Up`
    - `Busy`
    - `Dialing Offhook`
    - `Pre-ring`
    - `Unknown`
- `LocalTwoCallerIDNum`
- `LocalTwoCallerIDName`
- `LocalTwoConnectedLineNum`
- `LocalTwoConnectedLineName`
- `LocalTwoAccountCode`
- `LocalTwoContext`
- `LocalTwoExten`
- `LocalTwoPriority`
- `LocalTwoUniqueid`
- `DestTransfererChannel` - The name of the surviving transferer channel when a transfer results in a threeway call

> **Note**
> This header is only present when *DestType* is `Threeway`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_BlindTransfer

BlindTransfer

Synopsis

Raised when a blind transfer is complete.

Description

Syntax

```
Action:
Result: <value>
TransfererChannel: <value>
TransfererChannelState: <value>
TransfererChannelStateDesc: <value>
TransfererCallerIDNum: <value>
TransfererCallerIDName: <value>
TransfererConnectedLineNum: <value>
TransfererConnectedLineName: <value>
TransfererAccountCode: <value>
TransfererContext: <value>
TransfererExten: <value>
TransfererPriority: <value>
TransfererUniqueid: <value>
BridgeUniqueid: <value>
BridgeType: <value>
BridgeTechnology: <value>
BridgeNumChannels: <value>
IsExternal: <value>
Context: <value>
Extension: <value>
```

Arguments

- `Result` - Indicates if the transfer was successful or if it failed.
    - `Fail` - An internal error occurred.
    - `Invalid` - Invalid configuration for transfer (e.g. Not bridged)
    - `Not Permitted` - Bridge does not permit transfers
    - `Success` - Transfer completed successfully

    > **Note**
    > A result of `Success` does not necessarily mean that a target was succesfully contacted. It means that a party was succesfully placed into the dialplan at the expected location.

- `TransfererChannel`
- `TransfererChannelState` - A numeric code for the channel's current state, related to TransfererChannelStateDesc
- `TransfererChannelStateDesc`
    - `Down`
    - `Rsrvd`
    - `OffHook`
    - `Dialing`
    - `Ring`
    - `Ringing`
    - `Up`
    - `Busy`
    - `Dialing Offhook`
    - `Pre-ring`
    - `Unknown`
- `TransfererCallerIDNum`
- `TransfererCallerIDName`
- `TransfererConnectedLineNum`

- `TransfererConnectedLineName`
- `TransfererAccountCode`
- `TransfererContext`
- `TransfererExten`
- `TransfererPriority`
- `TransfererUniqueid`
- `BridgeUniqueid`
- `BridgeType` - The type of bridge
- `BridgeTechnology` - Technology in use by the bridge
- `BridgeNumChannels` - Number of channels in the bridge
- `IsExternal` - Indicates if the transfer was performed outside of Asterisk. For instance, a channel protocol native transfer is external. A DTMF transfer is internal.
    - `Yes`
    - `No`
- `Context` - Destination context for the blind transfer.
- `Extension` - Destination extension for the blind transfer.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_BridgeCreate

BridgeCreate

Synopsis

Raised when a bridge is created.

Description

Syntax

```
Action:
BridgeUniqueid: <value>
BridgeType: <value>
BridgeTechnology: <value>
BridgeNumChannels: <value>
```

Arguments

- `BridgeUniqueid`
- `BridgeType` - The type of bridge
- `BridgeTechnology` - Technology in use by the bridge
- `BridgeNumChannels` - Number of channels in the bridge

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_BridgeDestroy

BridgeDestroy

Synopsis

Raised when a bridge is destroyed.

Description

Syntax

```
Action:
BridgeUniqueid: <value>
BridgeType: <value>
BridgeTechnology: <value>
BridgeNumChannels: <value>
```

Arguments

- `BridgeUniqueid`
- `BridgeType` - The type of bridge
- `BridgeTechnology` - Technology in use by the bridge
- `BridgeNumChannels` - Number of channels in the bridge

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_BridgeEnter

BridgeEnter

Synopsis

Raised when a channel enters a bridge.

Description

Syntax

```
Action:
BridgeUniqueid: <value>
BridgeType: <value>
BridgeTechnology: <value>
BridgeNumChannels: <value>
Channel: <value>
ChannelState: <value>
ChannelStateDesc: <value>
CallerIDNum: <value>
CallerIDName: <value>
ConnectedLineNum: <value>
ConnectedLineName: <value>
AccountCode: <value>
Context: <value>
Exten: <value>
Priority: <value>
Uniqueid: <value>
SwapUniqueid: <value>
```

Arguments

- `BridgeUniqueid`
- `BridgeType` - The type of bridge
- `BridgeTechnology` - Technology in use by the bridge
- `BridgeNumChannels` - Number of channels in the bridge
- `Channel`
- `ChannelState` - A numeric code for the channel's current state, related to ChannelStateDesc
- `ChannelStateDesc`
    - `Down`
    - `Rsrvd`
    - `OffHook`
    - `Dialing`
    - `Ring`
    - `Ringing`
    - `Up`
    - `Busy`
    - `Dialing Offhook`
    - `Pre-ring`
    - `Unknown`
- `CallerIDNum`
- `CallerIDName`
- `ConnectedLineNum`
- `ConnectedLineName`
- `AccountCode`
- `Context`
- `Exten`
- `Priority`
- `Uniqueid`
- `SwapUniqueid` - The uniqueid of the channel being swapped out of the bridge

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_BridgeLeave

BridgeLeave

Synopsis

Raised when a channel leaves a bridge.

Description

Syntax

```
Action:
BridgeUniqueid: <value>
BridgeType: <value>
BridgeTechnology: <value>
BridgeNumChannels: <value>
Channel: <value>
ChannelState: <value>
ChannelStateDesc: <value>
CallerIDNum: <value>
CallerIDName: <value>
ConnectedLineNum: <value>
ConnectedLineName: <value>
AccountCode: <value>
Context: <value>
Exten: <value>
Priority: <value>
Uniqueid: <value>
```

Arguments

- `BridgeUniqueid`
- `BridgeType` - The type of bridge
- `BridgeTechnology` - Technology in use by the bridge
- `BridgeNumChannels` - Number of channels in the bridge
- `Channel`
- `ChannelState` - A numeric code for the channel's current state, related to ChannelStateDesc
- `ChannelStateDesc`
    - `Down`
    - `Rsrvd`
    - `OffHook`
    - `Dialing`
    - `Ring`
    - `Ringing`
    - `Up`
    - `Busy`
    - `Dialing Offhook`
    - `Pre-ring`
    - `Unknown`
- `CallerIDNum`
- `CallerIDName`
- `ConnectedLineNum`
- `ConnectedLineName`
- `AccountCode`
- `Context`

* `Exten`
* `Priority`
* `Uniqueid`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_BridgeMerge

BridgeMerge

Synopsis

Raised when two bridges are merged.

Description

Syntax

```
Action:
ToBridgeUniqueid: <value>
ToBridgeType: <value>
ToBridgeTechnology: <value>
ToBridgeNumChannels: <value>
FromBridgeUniqueid: <value>
FromBridgeType: <value>
FromBridgeTechnology: <value>
FromBridgeNumChannels: <value>
```

Arguments

* `ToBridgeUniqueid`
* `ToBridgeType` - The type of bridge
* `ToBridgeTechnology` - Technology in use by the bridge
* `ToBridgeNumChannels` - Number of channels in the bridge
* `FromBridgeUniqueid`
* `FromBridgeType` - The type of bridge
* `FromBridgeTechnology` - Technology in use by the bridge
* `FromBridgeNumChannels` - Number of channels in the bridge

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_ChanSpyStart

ChanSpyStart

Synopsis

Raised when one channel begins spying on another channel.

Description

Syntax

```
Action: 
SpyerChannel: <value>
SpyerChannelState: <value>
SpyerChannelStateDesc: <value>
SpyerCallerIDNum: <value>
SpyerCallerIDName: <value>
SpyerConnectedLineNum: <value>
SpyerConnectedLineName: <value>
SpyerAccountCode: <value>
SpyerContext: <value>
SpyerExten: <value>
SpyerPriority: <value>
SpyerUniqueid: <value>
SpyeeChannel: <value>
SpyeeChannelState: <value>
SpyeeChannelStateDesc: <value>
SpyeeCallerIDNum: <value>
SpyeeCallerIDName: <value>
SpyeeConnectedLineNum: <value>
SpyeeConnectedLineName: <value>
SpyeeAccountCode: <value>
SpyeeContext: <value>
SpyeeExten: <value>
SpyeePriority: <value>
SpyeeUniqueid: <value>
```

Arguments

- `SpyerChannel`
- `SpyerChannelState` - A numeric code for the channel's current state, related to SpyerChannelStateDesc
- `SpyerChannelStateDesc`
  - `Down`
  - `Rsrvd`
  - `OffHook`
  - `Dialing`
  - `Ring`
  - `Ringing`
  - `Up`
  - `Busy`
  - `Dialing Offhook`
  - `Pre-ring`
  - `Unknown`
- `SpyerCallerIDNum`
- `SpyerCallerIDName`
- `SpyerConnectedLineNum`
- `SpyerConnectedLineName`
- `SpyerAccountCode`
- `SpyerContext`
- `SpyerExten`
- `SpyerPriority`
- `SpyerUniqueid`
- `SpyeeChannel`
- `SpyeeChannelState` - A numeric code for the channel's current state, related to SpyeeChannelStateDesc

- SpyeeChannelStateDesc
    - Down
    - Rsrvd
    - OffHook
    - Dialing
    - Ring
    - Ringing
    - Up
    - Busy
    - Dialing Offhook
    - Pre-ring
    - Unknown
- SpyeeCallerIDNum
- SpyeeCallerIDName
- SpyeeConnectedLineNum
- SpyeeConnectedLineName
- SpyeeAccountCode
- SpyeeContext
- SpyeeExten
- SpyeePriority
- SpyeeUniqueid

See Also

- Asterisk 12 Application_ChanSpyStop

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_ChanSpyStop

ChanSpyStop

Synopsis

Raised when a channel has stopped spying.

Description

Syntax

```
Action:
SpyerChannel: <value>
SpyerChannelState: <value>
SpyerChannelStateDesc: <value>
SpyerCallerIDNum: <value>
SpyerCallerIDName: <value>
SpyerConnectedLineNum: <value>
SpyerConnectedLineName: <value>
SpyerAccountCode: <value>
SpyerContext: <value>
SpyerExten: <value>
SpyerPriority: <value>
SpyerUniqueid: <value>
SpyeeChannel: <value>
SpyeeChannelState: <value>
SpyeeChannelStateDesc: <value>
SpyeeCallerIDNum: <value>
SpyeeCallerIDName: <value>
SpyeeConnectedLineNum: <value>
SpyeeConnectedLineName: <value>
SpyeeAccountCode: <value>
SpyeeContext: <value>
SpyeeExten: <value>
SpyeePriority: <value>
SpyeeUniqueid: <value>
```

Arguments

- `SpyerChannel`
- `SpyerChannelState` - A numeric code for the channel's current state, related to SpyerChannelStateDesc
- `SpyerChannelStateDesc`
  - `Down`
  - `Rsrvd`
  - `OffHook`
  - `Dialing`
  - `Ring`
  - `Ringing`
  - `Up`
  - `Busy`
  - `Dialing Offhook`
  - `Pre-ring`
  - `Unknown`
- `SpyerCallerIDNum`
- `SpyerCallerIDName`
- `SpyerConnectedLineNum`
- `SpyerConnectedLineName`
- `SpyerAccountCode`
- `SpyerContext`
- `SpyerExten`
- `SpyerPriority`
- `SpyerUniqueid`
- `SpyeeChannel`
- `SpyeeChannelState` - A numeric code for the channel's current state, related to SpyeeChannelStateDesc
- `SpyeeChannelStateDesc`

        - `Down`
        - `Rsrvd`
        - `OffHook`
        - `Dialing`
        - `Ring`
        - `Ringing`
        - `Up`
        - `Busy`
        - `Dialing Offhook`
        - `Pre-ring`
        - `Unknown`
    - `SpyeeCallerIDNum`
    - `SpyeeCallerIDName`
    - `SpyeeConnectedLineNum`
    - `SpyeeConnectedLineName`
    - `SpyeeAccountCode`
    - `SpyeeContext`
    - `SpyeeExten`
    - `SpyeePriority`
    - `SpyeeUniqueid`

See Also

- Asterisk 12 Application_ChanSpyStart

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_ConfbridgeEnd

ConfbridgeEnd

Synopsis

Raised when a conference ends.

Description

Syntax

```
Action:
Conference: <value>
BridgeUniqueid: <value>
BridgeType: <value>
BridgeTechnology: <value>
BridgeNumChannels: <value>
```

Arguments

- `Conference` - The name of the Confbridge conference.
- `BridgeUniqueid`
- `BridgeType` - The type of bridge
- `BridgeTechnology` - Technology in use by the bridge
- `BridgeNumChannels` - Number of channels in the bridge

See Also

- Asterisk 12 ManagerEvent_ConfbridgeStart

* Asterisk 12 Application_ConfBridge

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_ConfbridgeJoin

ConfbridgeJoin

Synopsis

Raised when a channel joins a Confbridge conference.

Description

Syntax

```
Action:
Conference: <value>
BridgeUniqueid: <value>
BridgeType: <value>
BridgeTechnology: <value>
BridgeNumChannels: <value>
Channel: <value>
ChannelState: <value>
ChannelStateDesc: <value>
CallerIDNum: <value>
CallerIDName: <value>
ConnectedLineNum: <value>
ConnectedLineName: <value>
AccountCode: <value>
Context: <value>
Exten: <value>
Priority: <value>
Uniqueid: <value>
```

Arguments

* `Conference` - The name of the Confbridge conference.
* `BridgeUniqueid`
* `BridgeType` - The type of bridge
* `BridgeTechnology` - Technology in use by the bridge
* `BridgeNumChannels` - Number of channels in the bridge
* `Channel`
* `ChannelState` - A numeric code for the channel's current state, related to ChannelStateDesc
* `ChannelStateDesc`
    * `Down`
    * `Rsrvd`
    * `OffHook`
    * `Dialing`
    * `Ring`
    * `Ringing`
    * `Up`
    * `Busy`
    * `Dialing Offhook`

        - `Pre-ring`
        - `Unknown`
- `CallerIDNum`
- `CallerIDName`
- `ConnectedLineNum`
- `ConnectedLineName`
- `AccountCode`
- `Context`
- `Exten`
- `Priority`
- `Uniqueid`

See Also

- Asterisk 12 ManagerEvent_ConfbridgeLeave
- Asterisk 12 Application_ConfBridge

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_ConfbridgeLeave

ConfbridgeLeave

Synopsis

Raised when a channel leaves a Confbridge conference.

Description

Syntax

```
Action:
Conference: <value>
BridgeUniqueid: <value>
BridgeType: <value>
BridgeTechnology: <value>
BridgeNumChannels: <value>
Channel: <value>
ChannelState: <value>
ChannelStateDesc: <value>
CallerIDNum: <value>
CallerIDName: <value>
ConnectedLineNum: <value>
ConnectedLineName: <value>
AccountCode: <value>
Context: <value>
Exten: <value>
Priority: <value>
Uniqueid: <value>
```

Arguments

- `Conference` - The name of the Confbridge conference.
- `BridgeUniqueid`
- `BridgeType` - The type of bridge

- `BridgeTechnology` - Technology in use by the bridge
- `BridgeNumChannels` - Number of channels in the bridge
- `Channel`
- `ChannelState` - A numeric code for the channel's current state, related to ChannelStateDesc
- `ChannelStateDesc`
    - `Down`
    - `Rsrvd`
    - `OffHook`
    - `Dialing`
    - `Ring`
    - `Ringing`
    - `Up`
    - `Busy`
    - `Dialing Offhook`
    - `Pre-ring`
    - `Unknown`
- `CallerIDNum`
- `CallerIDName`
- `ConnectedLineNum`
- `ConnectedLineName`
- `AccountCode`
- `Context`
- `Exten`
- `Priority`
- `Uniqueid`

See Also

- Asterisk 12 ManagerEvent_ConfbridgeJoin
- Asterisk 12 Application_ConfBridge

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_ConfbridgeMute

ConfbridgeMute

Synopsis

Raised when a Confbridge participant mutes.

Description

Syntax

```
Action:
Conference: <value>
BridgeUniqueid: <value>
BridgeType: <value>
BridgeTechnology: <value>
BridgeNumChannels: <value>
Channel: <value>
ChannelState: <value>
ChannelStateDesc: <value>
CallerIDNum: <value>
CallerIDName: <value>
ConnectedLineNum: <value>
ConnectedLineName: <value>
AccountCode: <value>
Context: <value>
Exten: <value>
Priority: <value>
Uniqueid: <value>
```

Arguments

- `Conference` - The name of the Confbridge conference.
- `BridgeUniqueid`
- `BridgeType` - The type of bridge
- `BridgeTechnology` - Technology in use by the bridge
- `BridgeNumChannels` - Number of channels in the bridge
- `Channel`
- `ChannelState` - A numeric code for the channel's current state, related to ChannelStateDesc
- `ChannelStateDesc`
    - `Down`
    - `Rsrvd`
    - `OffHook`
    - `Dialing`
    - `Ring`
    - `Ringing`
    - `Up`
    - `Busy`
    - `Dialing Offhook`
    - `Pre-ring`
    - `Unknown`
- `CallerIDNum`
- `CallerIDName`
- `ConnectedLineNum`
- `ConnectedLineName`
- `AccountCode`
- `Context`
- `Exten`
- `Priority`
- `Uniqueid`

See Also

- Asterisk 12 ManagerEvent_ConfbridgeUnmute
- Asterisk 12 Application_ConfBridge

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_ConfbridgeRecord

ConfbridgeRecord

Synopsis

Raised when a conference starts recording.

Description

Syntax

```
Action:
Conference: <value>
BridgeUniqueid: <value>
BridgeType: <value>
BridgeTechnology: <value>
BridgeNumChannels: <value>
```

Arguments

- `Conference` - The name of the Confbridge conference.
- `BridgeUniqueid`
- `BridgeType` - The type of bridge
- `BridgeTechnology` - Technology in use by the bridge
- `BridgeNumChannels` - Number of channels in the bridge

See Also

- Asterisk 12 ManagerEvent_ConfbridgeStopRecord
- Asterisk 12 Application_ConfBridge

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_ConfbridgeStart

ConfbridgeStart

Synopsis

Raised when a conference starts.

Description

Syntax

```
Action:
Conference: <value>
BridgeUniqueid: <value>
BridgeType: <value>
BridgeTechnology: <value>
BridgeNumChannels: <value>
```

Arguments

- `Conference` - The name of the Confbridge conference.
- `BridgeUniqueid`
- `BridgeType` - The type of bridge
- `BridgeTechnology` - Technology in use by the bridge
- `BridgeNumChannels` - Number of channels in the bridge

See Also

- Asterisk 12 ManagerEvent_ConfbridgeEnd
- Asterisk 12 Application_ConfBridge

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_ConfbridgeStopRecord

ConfbridgeStopRecord

Synopsis

Raised when a conference that was recording stops recording.

Description

Syntax

```
Action:
Conference: <value>
BridgeUniqueid: <value>
BridgeType: <value>
BridgeTechnology: <value>
BridgeNumChannels: <value>
```

Arguments

- `Conference` - The name of the Confbridge conference.
- `BridgeUniqueid`
- `BridgeType` - The type of bridge
- `BridgeTechnology` - Technology in use by the bridge
- `BridgeNumChannels` - Number of channels in the bridge

See Also

- Asterisk 12 ManagerEvent_ConfbridgeRecord
- Asterisk 12 Application_ConfBridge

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_ConfbridgeTalking

ConfbridgeTalking

Synopsis

Raised when a confbridge participant unmutes.

Description

Syntax

```
Action:
Conference: <value>
BridgeUniqueid: <value>
BridgeType: <value>
BridgeTechnology: <value>
BridgeNumChannels: <value>
Channel: <value>
ChannelState: <value>
ChannelStateDesc: <value>
CallerIDNum: <value>
CallerIDName: <value>
ConnectedLineNum: <value>
ConnectedLineName: <value>
AccountCode: <value>
Context: <value>
Exten: <value>
Priority: <value>
Uniqueid: <value>
TalkingStatus: <value>
```

Arguments

- `Conference` - The name of the Confbridge conference.
- `BridgeUniqueid`
- `BridgeType` - The type of bridge
- `BridgeTechnology` - Technology in use by the bridge
- `BridgeNumChannels` - Number of channels in the bridge
- `Channel`
- `ChannelState` - A numeric code for the channel's current state, related to ChannelStateDesc
- `ChannelStateDesc`
    - `Down`
    - `Rsrvd`
    - `OffHook`
    - `Dialing`
    - `Ring`
    - `Ringing`
    - `Up`
    - `Busy`
    - `Dialing Offhook`
    - `Pre-ring`

        - `Unknown`
- `CallerIDNum`
- `CallerIDName`
- `ConnectedLineNum`
- `ConnectedLineName`
- `AccountCode`
- `Context`
- `Exten`
- `Priority`
- `Uniqueid`
- `TalkingStatus`
    - `on`
    - `off`

See Also

- Asterisk 12 Application_ConfBridge

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_ConfbridgeUnmute

ConfbridgeUnmute

Synopsis

Raised when a confbridge participant unmutes.

Description

Syntax

```
Action: 
Conference: <value>
BridgeUniqueid: <value>
BridgeType: <value>
BridgeTechnology: <value>
BridgeNumChannels: <value>
Channel: <value>
ChannelState: <value>
ChannelStateDesc: <value>
CallerIDNum: <value>
CallerIDName: <value>
ConnectedLineNum: <value>
ConnectedLineName: <value>
AccountCode: <value>
Context: <value>
Exten: <value>
Priority: <value>
Uniqueid: <value>
```

Arguments

- `Conference` - The name of the Confbridge conference.
- `BridgeUniqueid`

* `BridgeType` - The type of bridge
* `BridgeTechnology` - Technology in use by the bridge
* `BridgeNumChannels` - Number of channels in the bridge
* `Channel`
* `ChannelState` - A numeric code for the channel's current state, related to ChannelStateDesc
* `ChannelStateDesc`
    * `Down`
    * `Rsrvd`
    * `OffHook`
    * `Dialing`
    * `Ring`
    * `Ringing`
    * `Up`
    * `Busy`
    * `Dialing Offhook`
    * `Pre-ring`
    * `Unknown`
* `CallerIDNum`
* `CallerIDName`
* `ConnectedLineNum`
* `ConnectedLineName`
* `AccountCode`
* `Context`
* `Exten`
* `Priority`
* `Uniqueid`

See Also

* Asterisk 12 ManagerEvent_ConfbridgeMute
* Asterisk 12 Application_ConfBridge

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_DAHDIChannel

DAHDIChannel

Synopsis

Raised when a DAHDI channel is created or an underlying technology is associated with a DAHDI channel.

Description

Syntax

```
Action:
Channel: <value>
ChannelState: <value>
ChannelStateDesc: <value>
CallerIDNum: <value>
CallerIDName: <value>
ConnectedLineNum: <value>
ConnectedLineName: <value>
AccountCode: <value>
Context: <value>
Exten: <value>
Priority: <value>
Uniqueid: <value>
DAHDISpan: <value>
DAHDIChannel: <value>
```

Arguments

- `Channel`
- `ChannelState` - A numeric code for the channel's current state, related to ChannelStateDesc
- `ChannelStateDesc`
    - `Down`
    - `Rsrvd`
    - `OffHook`
    - `Dialing`
    - `Ring`
    - `Ringing`
    - `Up`
    - `Busy`
    - `Dialing Offhook`
    - `Pre-ring`
    - `Unknown`
- `CallerIDNum`
- `CallerIDName`
- `ConnectedLineNum`
- `ConnectedLineName`
- `AccountCode`
- `Context`
- `Exten`
- `Priority`
- `Uniqueid`
- `DAHDISpan` - The DAHDI span associated with this channel.
- `DAHDIChannel` - The DAHDI channel associated with this channel.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_DialBegin

DialBegin

Synopsis

Raised when a dial action has started.

Description

Syntax

```
Action:
Channel: <value>
ChannelState: <value>
ChannelStateDesc: <value>
CallerIDNum: <value>
CallerIDName: <value>
ConnectedLineNum: <value>
ConnectedLineName: <value>
AccountCode: <value>
Context: <value>
Exten: <value>
Priority: <value>
Uniqueid: <value>
DestChannel: <value>
DestChannelState: <value>
DestChannelStateDesc: <value>
DestCallerIDNum: <value>
DestCallerIDName: <value>
DestConnectedLineNum: <value>
DestConnectedLineName: <value>
DestAccountCode: <value>
DestContext: <value>
DestExten: <value>
DestPriority: <value>
DestUniqueid: <value>
DialString: <value>
```

Arguments

- `Channel`
- `ChannelState` - A numeric code for the channel's current state, related to ChannelStateDesc
- `ChannelStateDesc`
    - `Down`
    - `Rsrvd`
    - `OffHook`
    - `Dialing`
    - `Ring`
    - `Ringing`
    - `Up`
    - `Busy`
    - `Dialing Offhook`
    - `Pre-ring`
    - `Unknown`
- `CallerIDNum`
- `CallerIDName`
- `ConnectedLineNum`
- `ConnectedLineName`
- `AccountCode`

- `Context`
- `Exten`
- `Priority`
- `Uniqueid`
- `DestChannel`
- `DestChannelState` - A numeric code for the channel's current state, related to DestChannelStateDesc
- `DestChannelStateDesc`
  - `Down`
  - `Rsrvd`
  - `OffHook`
  - `Dialing`
  - `Ring`
  - `Ringing`
  - `Up`
  - `Busy`
  - `Dialing Offhook`
  - `Pre-ring`
  - `Unknown`
- `DestCallerIDNum`
- `DestCallerIDName`
- `DestConnectedLineNum`
- `DestConnectedLineName`
- `DestAccountCode`
- `DestContext`
- `DestExten`
- `DestPriority`
- `DestUniqueid`
- `DialString` - The non-technology specific device being dialed.

See Also

- Asterisk 12 Application_Dial

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_DialEnd

DialEnd

Synopsis

Raised when a dial action has completed.

Description

Syntax

```
Action:
Channel: <value>
ChannelState: <value>
ChannelStateDesc: <value>
CallerIDNum: <value>
CallerIDName: <value>
ConnectedLineNum: <value>
ConnectedLineName: <value>
AccountCode: <value>
Context: <value>
Exten: <value>
Priority: <value>
Uniqueid: <value>
DestChannel: <value>
DestChannelState: <value>
DestChannelStateDesc: <value>
DestCallerIDNum: <value>
DestCallerIDName: <value>
DestConnectedLineNum: <value>
DestConnectedLineName: <value>
DestAccountCode: <value>
DestContext: <value>
DestExten: <value>
DestPriority: <value>
DestUniqueid: <value>
DialStatus: <value>
```

Arguments

- `Channel`
- `ChannelState` - A numeric code for the channel's current state, related to ChannelStateDesc
- `ChannelStateDesc`
    - `Down`
    - `Rsrvd`
    - `OffHook`
    - `Dialing`
    - `Ring`
    - `Ringing`
    - `Up`
    - `Busy`
    - `Dialing Offhook`
    - `Pre-ring`
    - `Unknown`
- `CallerIDNum`
- `CallerIDName`
- `ConnectedLineNum`
- `ConnectedLineName`
- `AccountCode`
- `Context`
- `Exten`
- `Priority`
- `Uniqueid`
- `DestChannel`
- `DestChannelState` - A numeric code for the channel's current state, related to DestChannelStateDesc

- `DestChannelStateDesc`
    - `Down`
    - `Rsrvd`
    - `OffHook`
    - `Dialing`
    - `Ring`
    - `Ringing`
    - `Up`
    - `Busy`
    - `Dialing Offhook`
    - `Pre-ring`
    - `Unknown`
- `DestCallerIDNum`
- `DestCallerIDName`
- `DestConnectedLineNum`
- `DestConnectedLineName`
- `DestAccountCode`
- `DestContext`
- `DestExten`
- `DestPriority`
- `DestUniqueid`
- `DialStatus` - The result of the dial operation.
    - `ANSWER`
    - `BUSY`
    - `CANCEL`
    - `CHANUNAVAIL`
    - `CONGESTION`
    - `NOANSWER`

See Also

- Asterisk 12 Application_Dial

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_DNDState

DNDState

Synopsis

Raised when the Do Not Disturb state is changed on a DAHDI channel.

Description

Syntax

```
Action:
DAHDIChannel: <value>
Status: <value>
```

Arguments

- `DAHDIChannel` - The DAHDI channel on which DND status changed.

> **Note**

This is not an Asterisk channel identifier.

- `Status`
  - `enabled`
  - `disabled`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_DTMFBegin

DTMFBegin

Synopsis

Raised when a DTMF digit has started on a channel.

Description

Syntax

```
Action:
Channel: <value>
ChannelState: <value>
ChannelStateDesc: <value>
CallerIDNum: <value>
CallerIDName: <value>
ConnectedLineNum: <value>
ConnectedLineName: <value>
AccountCode: <value>
Context: <value>
Exten: <value>
Priority: <value>
Uniqueid: <value>
Digit: <value>
Direction: <value>
```

Arguments

- `Channel`
- `ChannelState` - A numeric code for the channel's current state, related to ChannelStateDesc
- `ChannelStateDesc`
  - `Down`
  - `Rsrvd`
  - `OffHook`
  - `Dialing`
  - `Ring`
  - `Ringing`
  - `Up`
  - `Busy`
  - `Dialing Offhook`
  - `Pre-ring`
  - `Unknown`
- `CallerIDNum`

- `CallerIDName`
- `ConnectedLineNum`
- `ConnectedLineName`
- `AccountCode`
- `Context`
- `Exten`
- `Priority`
- `Uniqueid`
- `Digit` - DTMF digit received or transmitted (0-9, A-E, # or *
- `Direction`
    - `Received`
    - `Sent`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_DTMFEnd

DTMFEnd

Synopsis

Raised when a DTMF digit has ended on a channel.

Description

Syntax

```
Action:
Channel: <value>
ChannelState: <value>
ChannelStateDesc: <value>
CallerIDNum: <value>
CallerIDName: <value>
ConnectedLineNum: <value>
ConnectedLineName: <value>
AccountCode: <value>
Context: <value>
Exten: <value>
Priority: <value>
Uniqueid: <value>
Digit: <value>
DurationMs: <value>
Direction: <value>
```

Arguments

- `Channel`
- `ChannelState` - A numeric code for the channel's current state, related to ChannelStateDesc
- `ChannelStateDesc`
    - `Down`
    - `Rsrvd`
    - `OffHook`
    - `Dialing`

        - `Ring`
        - `Ringing`
        - `Up`
        - `Busy`
        - `Dialing Offhook`
        - `Pre-ring`
        - `Unknown`
- `CallerIDNum`
- `CallerIDName`
- `ConnectedLineNum`
- `ConnectedLineName`
- `AccountCode`
- `Context`
- `Exten`
- `Priority`
- `Uniqueid`
- `Digit` - DTMF digit received or transmitted (0-9, A-E, # or *
- `DurationMs` - Duration (in milliseconds) DTMF was sent/received
- `Direction`
    - `Received`
    - `Sent`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_ExtensionStatus

ExtensionStatus

Synopsis

Raised when an extension state has changed.

Description

Syntax

```
Action:
Exten: <value>
Context: <value>
Hint: <value>
Status: <value>
```

Arguments

- `Exten`
- `Context`
- `Hint`
- `Status`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_FAXStatus

FAXStatus

Synopsis

Raised periodically during a fax transmission.

Description

Syntax

```
Action:
Channel: <value>
ChannelState: <value>
ChannelStateDesc: <value>
CallerIDNum: <value>
CallerIDName: <value>
ConnectedLineNum: <value>
ConnectedLineName: <value>
AccountCode: <value>
Context: <value>
Exten: <value>
Priority: <value>
Uniqueid: <value>
Operation: <value>
Status: <value>
LocalStationID: <value>
FileName: <value>
```

Arguments

- `Channel`
- `ChannelState` - A numeric code for the channel's current state, related to ChannelStateDesc
- `ChannelStateDesc`
    - `Down`
    - `Rsrvd`
    - `OffHook`
    - `Dialing`
    - `Ring`
    - `Ringing`
    - `Up`
    - `Busy`
    - `Dialing Offhook`
    - `Pre-ring`
    - `Unknown`
- `CallerIDNum`
- `CallerIDName`
- `ConnectedLineNum`
- `ConnectedLineName`
- `AccountCode`
- `Context`
- `Exten`
- `Priority`
- `Uniqueid`
- `Operation`

        - `gateway`
        - `receive`
        - `send`
- `Status` - A text message describing the current status of the fax
- `LocalStationID` - The value of the `LOCALSTATIONID` channel variable
- `FileName` - The files being affected by the fax operation

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_FullyBooted

FullyBooted

Synopsis

Raised when all Asterisk initialization procedures have finished.

Description

Syntax

```
Action:
Status: <value>
```

Arguments

- `Status` - Informational message

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_Hangup

Hangup

Synopsis

Raised when a channel is hung up.

Description

Syntax

```
Action:
Channel: <value>
ChannelState: <value>
ChannelStateDesc: <value>
CallerIDNum: <value>
CallerIDName: <value>
ConnectedLineNum: <value>
ConnectedLineName: <value>
AccountCode: <value>
Context: <value>
Exten: <value>
Priority: <value>
Uniqueid: <value>
Cause: <value>
Cause-txt: <value>
```

Arguments

- `Channel`
- `ChannelState` - A numeric code for the channel's current state, related to ChannelStateDesc
- `ChannelStateDesc`
    - `Down`
    - `Rsrvd`
    - `OffHook`
    - `Dialing`
    - `Ring`
    - `Ringing`
    - `Up`
    - `Busy`
    - `Dialing Offhook`
    - `Pre-ring`
    - `Unknown`
- `CallerIDNum`
- `CallerIDName`
- `ConnectedLineNum`
- `ConnectedLineName`
- `AccountCode`
- `Context`
- `Exten`
- `Priority`
- `Uniqueid`
- `Cause` - A numeric cause code for why the channel was hung up.
- `Cause-txt` - A description of why the channel was hung up.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_HangupHandlerPop

HangupHandlerPop

Synopsis

Raised when a hangup handler is removed from the handler stack by the CHANNEL() function.

Description

Syntax

```
Action:
Channel: <value>
ChannelState: <value>
ChannelStateDesc: <value>
CallerIDNum: <value>
CallerIDName: <value>
ConnectedLineNum: <value>
ConnectedLineName: <value>
AccountCode: <value>
Context: <value>
Exten: <value>
Priority: <value>
Uniqueid: <value>
Handler: <value>
```

Arguments

- `Channel`
- `ChannelState` - A numeric code for the channel's current state, related to ChannelStateDesc
- `ChannelStateDesc`
    - `Down`
    - `Rsrvd`
    - `OffHook`
    - `Dialing`
    - `Ring`
    - `Ringing`
    - `Up`
    - `Busy`
    - `Dialing Offhook`
    - `Pre-ring`
    - `Unknown`
- `CallerIDNum`
- `CallerIDName`
- `ConnectedLineNum`
- `ConnectedLineName`
- `AccountCode`
- `Context`
- `Exten`
- `Priority`
- `Uniqueid`
- `Handler` - Hangup handler parameter string passed to the Gosub application.

See Also

- Asterisk 12 ManagerEvent_HangupHandlerPush
- Asterisk 12 Function_CHANNEL

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_HangupHandlerPush

HangupHandlerPush

Synopsis

Raised when a hangup handler is added to the handler stack by the CHANNEL() function.

Description

Syntax

```
Action:
Channel: <value>
ChannelState: <value>
ChannelStateDesc: <value>
CallerIDNum: <value>
CallerIDName: <value>
ConnectedLineNum: <value>
ConnectedLineName: <value>
AccountCode: <value>
Context: <value>
Exten: <value>
Priority: <value>
Uniqueid: <value>
Handler: <value>
```

Arguments

- `Channel`
- `ChannelState` - A numeric code for the channel's current state, related to ChannelStateDesc
- `ChannelStateDesc`
  - `Down`
  - `Rsrvd`
  - `OffHook`
  - `Dialing`
  - `Ring`
  - `Ringing`
  - `Up`
  - `Busy`
  - `Dialing Offhook`
  - `Pre-ring`
  - `Unknown`
- `CallerIDNum`
- `CallerIDName`
- `ConnectedLineNum`
- `ConnectedLineName`
- `AccountCode`
- `Context`
- `Exten`
- `Priority`
- `Uniqueid`
- `Handler` - Hangup handler parameter string passed to the Gosub application.

See Also

- Asterisk 12 ManagerEvent_HangupHandlerPop

* Asterisk 12 Function_CHANNEL

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_HangupHandlerRun

HangupHandlerRun

Synopsis

Raised when a hangup handler is about to be called.

Description

Syntax

```
Action:
Channel: <value>
ChannelState: <value>
ChannelStateDesc: <value>
CallerIDNum: <value>
CallerIDName: <value>
ConnectedLineNum: <value>
ConnectedLineName: <value>
AccountCode: <value>
Context: <value>
Exten: <value>
Priority: <value>
Uniqueid: <value>
Handler: <value>
```

Arguments

* `Channel`
* `ChannelState` - A numeric code for the channel's current state, related to ChannelStateDesc
* `ChannelStateDesc`
    * `Down`
    * `Rsrvd`
    * `OffHook`
    * `Dialing`
    * `Ring`
    * `Ringing`
    * `Up`
    * `Busy`
    * `Dialing Offhook`
    * `Pre-ring`
    * `Unknown`
* `CallerIDNum`
* `CallerIDName`
* `ConnectedLineNum`
* `ConnectedLineName`
* `AccountCode`
* `Context`
* `Exten`
* `Priority`

* `Uniqueid`
* `Handler` - Hangup handler parameter string passed to the Gosub application.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_HangupRequest

HangupRequest

Synopsis

Raised when a hangup is requested.

Description

Syntax

```
Action:
Channel: <value>
ChannelState: <value>
ChannelStateDesc: <value>
CallerIDNum: <value>
CallerIDName: <value>
ConnectedLineNum: <value>
ConnectedLineName: <value>
AccountCode: <value>
Context: <value>
Exten: <value>
Priority: <value>
Uniqueid: <value>
Cause: <value>
```

Arguments

* `Channel`
* `ChannelState` - A numeric code for the channel's current state, related to ChannelStateDesc
* `ChannelStateDesc`
    * `Down`
    * `Rsrvd`
    * `OffHook`
    * `Dialing`
    * `Ring`
    * `Ringing`
    * `Up`
    * `Busy`
    * `Dialing Offhook`
    * `Pre-ring`
    * `Unknown`
* `CallerIDNum`
* `CallerIDName`
* `ConnectedLineNum`
* `ConnectedLineName`
* `AccountCode`

* `Context`
* `Exten`
* `Priority`
* `Uniqueid`
* `Cause` - A numeric cause code for why the channel was hung up.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_Hold

Hold

Synopsis

Raised when a channel goes on hold.

Description

Syntax

```
Action:
Channel: <value>
ChannelState: <value>
ChannelStateDesc: <value>
CallerIDNum: <value>
CallerIDName: <value>
ConnectedLineNum: <value>
ConnectedLineName: <value>
AccountCode: <value>
Context: <value>
Exten: <value>
Priority: <value>
Uniqueid: <value>
MusicClass: <value>
```

Arguments

* `Channel`
* `ChannelState` - A numeric code for the channel's current state, related to ChannelStateDesc
* `ChannelStateDesc`
    * `Down`
    * `Rsrvd`
    * `OffHook`
    * `Dialing`
    * `Ring`
    * `Ringing`
    * `Up`
    * `Busy`
    * `Dialing Offhook`
    * `Pre-ring`
    * `Unknown`
* `CallerIDNum`
* `CallerIDName`

- `ConnectedLineNum`
- `ConnectedLineName`
- `AccountCode`
- `Context`
- `Exten`
- `Priority`
- `Uniqueid`
- `MusicClass` - The suggested MusicClass, if provided.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_LocalBridge

LocalBridge

Synopsis

Raised when two halves of a Local Channel form a bridge.

Description

Syntax

```
Action:
LocalOneChannel: <value>
LocalOneChannelState: <value>
LocalOneChannelStateDesc: <value>
LocalOneCallerIDNum: <value>
LocalOneCallerIDName: <value>
LocalOneConnectedLineNum: <value>
LocalOneConnectedLineName: <value>
LocalOneAccountCode: <value>
LocalOneContext: <value>
LocalOneExten: <value>
LocalOnePriority: <value>
LocalOneUniqueid: <value>
LocalTwoChannel: <value>
LocalTwoChannelState: <value>
LocalTwoChannelStateDesc: <value>
LocalTwoCallerIDNum: <value>
LocalTwoCallerIDName: <value>
LocalTwoConnectedLineNum: <value>
LocalTwoConnectedLineName: <value>
LocalTwoAccountCode: <value>
LocalTwoContext: <value>
LocalTwoExten: <value>
LocalTwoPriority: <value>
LocalTwoUniqueid: <value>
Context: <value>
Exten: <value>
LocalOptimization: <value>
```

Arguments

- `LocalOneChannel`
- `LocalOneChannelState` - A numeric code for the channel's current state, related to LocalOneChannelStateDesc
- `LocalOneChannelStateDesc`
  - `Down`
  - `Rsrvd`
  - `OffHook`
  - `Dialing`
  - `Ring`
  - `Ringing`
  - `Up`
  - `Busy`
  - `Dialing Offhook`
  - `Pre-ring`
  - `Unknown`
- `LocalOneCallerIDNum`
- `LocalOneCallerIDName`
- `LocalOneConnectedLineNum`
- `LocalOneConnectedLineName`
- `LocalOneAccountCode`
- `LocalOneContext`
- `LocalOneExten`
- `LocalOnePriority`
- `LocalOneUniqueid`

- `LocalTwoChannel`
- `LocalTwoChannelState` - A numeric code for the channel's current state, related to LocalTwoChannelStateDesc
- `LocalTwoChannelStateDesc`
  - `Down`
  - `Rsrvd`
  - `OffHook`
  - `Dialing`
  - `Ring`
  - `Ringing`
  - `Up`
  - `Busy`
  - `Dialing Offhook`
  - `Pre-ring`
  - `Unknown`
- `LocalTwoCallerIDNum`
- `LocalTwoCallerIDName`
- `LocalTwoConnectedLineNum`
- `LocalTwoConnectedLineName`
- `LocalTwoAccountCode`
- `LocalTwoContext`
- `LocalTwoExten`
- `LocalTwoPriority`
- `LocalTwoUniqueid`
- `Context` - The context in the dialplan that Channel2 starts in.
- `Exten` - The extension in the dialplan that Channel2 starts in.
- `LocalOptimization`
  - `Yes`
  - `No`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_LocalOptimizationBegin

LocalOptimizationBegin

Synopsis

Raised when two halves of a Local Channel begin to optimize themselves out of the media path.

Description

Syntax

```
Action:
LocalOneChannel: <value>
LocalOneChannelState: <value>
LocalOneChannelStateDesc: <value>
LocalOneCallerIDNum: <value>
LocalOneCallerIDName: <value>
LocalOneConnectedLineNum: <value>
LocalOneConnectedLineName: <value>
LocalOneAccountCode: <value>
LocalOneContext: <value>
LocalOneExten: <value>
LocalOnePriority: <value>
LocalOneUniqueid: <value>
LocalTwoChannel: <value>
LocalTwoChannelState: <value>
LocalTwoChannelStateDesc: <value>
LocalTwoCallerIDNum: <value>
LocalTwoCallerIDName: <value>
LocalTwoConnectedLineNum: <value>
LocalTwoConnectedLineName: <value>
LocalTwoAccountCode: <value>
LocalTwoContext: <value>
LocalTwoExten: <value>
LocalTwoPriority: <value>
LocalTwoUniqueid: <value>
SourceChannel: <value>
SourceChannelState: <value>
SourceChannelStateDesc: <value>
SourceCallerIDNum: <value>
SourceCallerIDName: <value>
SourceConnectedLineNum: <value>
SourceConnectedLineName: <value>
SourceAccountCode: <value>
SourceContext: <value>
SourceExten: <value>
SourcePriority: <value>
SourceUniqueid: <value>
DestUniqueId: <value>
Id: <value>
```

Arguments

- `LocalOneChannel`
- `LocalOneChannelState` - A numeric code for the channel's current state, related to LocalOneChannelStateDesc
- `LocalOneChannelStateDesc`
    - `Down`
    - `Rsrvd`
    - `OffHook`
    - `Dialing`
    - `Ring`
    - `Ringing`
    - `Up`

    - Busy
    - Dialing Offhook
    - Pre-ring
    - Unknown
- `LocalOneCallerIDNum`
- `LocalOneCallerIDName`
- `LocalOneConnectedLineNum`
- `LocalOneConnectedLineName`
- `LocalOneAccountCode`
- `LocalOneContext`
- `LocalOneExten`
- `LocalOnePriority`
- `LocalOneUniqueid`
- `LocalTwoChannel`
- `LocalTwoChannelState` - A numeric code for the channel's current state, related to LocalTwoChannelStateDesc
- `LocalTwoChannelStateDesc`
    - Down
    - Rsrvd
    - OffHook
    - Dialing
    - Ring
    - Ringing
    - Up
    - Busy
    - Dialing Offhook
    - Pre-ring
    - Unknown
- `LocalTwoCallerIDNum`
- `LocalTwoCallerIDName`
- `LocalTwoConnectedLineNum`
- `LocalTwoConnectedLineName`
- `LocalTwoAccountCode`
- `LocalTwoContext`
- `LocalTwoExten`
- `LocalTwoPriority`
- `LocalTwoUniqueid`
- `SourceChannel`
- `SourceChannelState` - A numeric code for the channel's current state, related to SourceChannelStateDesc
- `SourceChannelStateDesc`
    - Down
    - Rsrvd
    - OffHook
    - Dialing
    - Ring
    - Ringing
    - Up
    - Busy
    - Dialing Offhook
    - Pre-ring
    - Unknown
- `SourceCallerIDNum`
- `SourceCallerIDName`
- `SourceConnectedLineNum`
- `SourceConnectedLineName`
- `SourceAccountCode`
- `SourceContext`
- `SourceExten`
- `SourcePriority`
- `SourceUniqueid`
- `DestUniqueId` - The unique ID of the bridge into which the local channel is optimizing.

* `Id` - Identification for the optimization operation.

See Also

* Asterisk 12 ManagerEvent_LocalOptimizationEnd
* Asterisk 12 ManagerAction_LocalOptimizeAway

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_LocalOptimizationEnd

LocalOptimizationEnd

Synopsis

Raised when two halves of a Local Channel have finished optimizing themselves out of the media path.

Description

Syntax

```
Action: 
LocalOneChannel: <value>
LocalOneChannelState: <value>
LocalOneChannelStateDesc: <value>
LocalOneCallerIDNum: <value>
LocalOneCallerIDName: <value>
LocalOneConnectedLineNum: <value>
LocalOneConnectedLineName: <value>
LocalOneAccountCode: <value>
LocalOneContext: <value>
LocalOneExten: <value>
LocalOnePriority: <value>
LocalOneUniqueid: <value>
LocalTwoChannel: <value>
LocalTwoChannelState: <value>
LocalTwoChannelStateDesc: <value>
LocalTwoCallerIDNum: <value>
LocalTwoCallerIDName: <value>
LocalTwoConnectedLineNum: <value>
LocalTwoConnectedLineName: <value>
LocalTwoAccountCode: <value>
LocalTwoContext: <value>
LocalTwoExten: <value>
LocalTwoPriority: <value>
LocalTwoUniqueid: <value>
Success: <value>
Id: <value>
```

Arguments

* `LocalOneChannel`

- `LocalOneChannelState` - A numeric code for the channel's current state, related to LocalOneChannelStateDesc
- `LocalOneChannelStateDesc`
    - `Down`
    - `Rsrvd`
    - `OffHook`
    - `Dialing`
    - `Ring`
    - `Ringing`
    - `Up`
    - `Busy`
    - `Dialing Offhook`
    - `Pre-ring`
    - `Unknown`
- `LocalOneCallerIDNum`
- `LocalOneCallerIDName`
- `LocalOneConnectedLineNum`
- `LocalOneConnectedLineName`
- `LocalOneAccountCode`
- `LocalOneContext`
- `LocalOneExten`
- `LocalOnePriority`
- `LocalOneUniqueid`
- `LocalTwoChannel`
- `LocalTwoChannelState` - A numeric code for the channel's current state, related to LocalTwoChannelStateDesc
- `LocalTwoChannelStateDesc`
    - `Down`
    - `Rsrvd`
    - `OffHook`
    - `Dialing`
    - `Ring`
    - `Ringing`
    - `Up`
    - `Busy`
    - `Dialing Offhook`
    - `Pre-ring`
    - `Unknown`
- `LocalTwoCallerIDNum`
- `LocalTwoCallerIDName`
- `LocalTwoConnectedLineNum`
- `LocalTwoConnectedLineName`
- `LocalTwoAccountCode`
- `LocalTwoContext`
- `LocalTwoExten`
- `LocalTwoPriority`
- `LocalTwoUniqueid`
- `Success` - Indicates whether the local optimization succeeded.
- `Id` - Identification for the optimization operation. Matches the *Id* from a previous `LocalOptimizationBegin`

See Also

- Asterisk 12 ManagerEvent_LocalOptimizationBegin
- Asterisk 12 ManagerAction_LocalOptimizeAway

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_LogChannel

LogChannel

Synopsis

Raised when a logging channel is re-enabled after a reload operation.

Description

Syntax

```
Action:
Channel: <value>
Enabled: <value>
```

Arguments

- `Channel` - The name of the logging channel.
- `Enabled`

See Also

Synopsis

Raised when a logging channel is disabled.

Description

Syntax

```
Action:
Channel: <value>
Enabled: <value>
Reason: <value>
```

Arguments

- `Channel` - The name of the logging channel.
- `Enabled`
- `Reason`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_MCID

MCID

Synopsis

Published when a malicious call ID request arrives.

Description

Syntax

```
Action:
Channel: <value>
ChannelState: <value>
ChannelStateDesc: <value>
CallerIDNum: <value>
CallerIDName: <value>
ConnectedLineNum: <value>
ConnectedLineName: <value>
AccountCode: <value>
Context: <value>
Exten: <value>
Priority: <value>
Uniqueid: <value>
MCallerIDNumValid: <value>
MCallerIDNum: <value>
MCallerIDton: <value>
MCallerIDNumPlan: <value>
MCallerIDNumPres: <value>
MCallerIDNameValid: <value>
MCallerIDName: <value>
MCallerIDNameCharSet: <value>
MCallerIDNamePres: <value>
MCallerIDSubaddr: <value>
MCallerIDSubaddrType: <value>
MCallerIDSubaddrOdd: <value>
MCallerIDPres: <value>
MConnectedIDNumValid: <value>
MConnectedIDNum: <value>
MConnectedIDton: <value>
MConnectedIDNumPlan: <value>
MConnectedIDNumPres: <value>
MConnectedIDNameValid: <value>
MConnectedIDName: <value>
MConnectedIDNameCharSet: <value>
MConnectedIDNamePres: <value>
MConnectedIDSubaddr: <value>
MConnectedIDSubaddrType: <value>
MConnectedIDSubaddrOdd: <value>
MConnectedIDPres: <value>
```

Arguments

- `Channel`
- `ChannelState` - A numeric code for the channel's current state, related to ChannelStateDesc
- `ChannelStateDesc`
    - `Down`
    - `Rsrvd`
    - `OffHook`
    - `Dialing`
    - `Ring`
    - `Ringing`
    - `Up`

    - Busy
    - Dialing Offhook
    - Pre-ring
    - Unknown
- CallerIDNum
- CallerIDName
- ConnectedLineNum
- ConnectedLineName
- AccountCode
- Context
- Exten
- Priority
- Uniqueid
- MCallerIDNumValid
- MCallerIDNum
- MCallerIDton
- MCallerIDNumPlan
- MCallerIDNumPres
- MCallerIDNameValid
- MCallerIDName
- MCallerIDNameCharSet
- MCallerIDNamePres
- MCallerIDSubaddr
- MCallerIDSubaddrType
- MCallerIDSubaddrOdd
- MCallerIDPres
- MConnectedIDNumValid
- MConnectedIDNum
- MConnectedIDton
- MConnectedIDNumPlan
- MConnectedIDNumPres
- MConnectedIDNameValid
- MConnectedIDName
- MConnectedIDNameCharSet
- MConnectedIDNamePres
- MConnectedIDSubaddr
- MConnectedIDSubaddrType
- MConnectedIDSubaddrOdd
- MConnectedIDPres

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_MeetmeEnd

MeetmeEnd

Synopsis

Raised when a MeetMe conference ends.

Description

Syntax

```
Action:
Meetme: <value>
```

Arguments

- `Meetme` - The identifier for the MeetMe conference.

See Also

- Asterisk 12 ManagerEvent_MeetmeJoin

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_MeetmeJoin

MeetmeJoin

Synopsis

Raised when a user joins a MeetMe conference.

Description

Syntax

```
Action:
Meetme: <value>
Usernum: <value>
Channel: <value>
ChannelState: <value>
ChannelStateDesc: <value>
CallerIDNum: <value>
CallerIDName: <value>
ConnectedLineNum: <value>
ConnectedLineName: <value>
AccountCode: <value>
Context: <value>
Exten: <value>
Priority: <value>
Uniqueid: <value>
```

Arguments

- `Meetme` - The identifier for the MeetMe conference.
- `Usernum` - The identifier of the MeetMe user who joined.
- `Channel`
- `ChannelState` - A numeric code for the channel's current state, related to ChannelStateDesc
- `ChannelStateDesc`
    - `Down`
    - `Rsrvd`
    - `OffHook`
    - `Dialing`

    - Ring
    - Ringing
    - Up
    - Busy
    - Dialing Offhook
    - Pre-ring
    - Unknown
- CallerIDNum
- CallerIDName
- ConnectedLineNum
- ConnectedLineName
- AccountCode
- Context
- Exten
- Priority
- Uniqueid

See Also

- Asterisk 12 ManagerEvent_MeetmeLeave
- Asterisk 12 Application_MeetMe

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_MeetmeLeave

MeetmeLeave

Synopsis

Raised when a user leaves a MeetMe conference.

Description

Syntax

```
Action:
Meetme: <value>
Usernum: <value>
Channel: <value>
ChannelState: <value>
ChannelStateDesc: <value>
CallerIDNum: <value>
CallerIDName: <value>
ConnectedLineNum: <value>
ConnectedLineName: <value>
AccountCode: <value>
Context: <value>
Exten: <value>
Priority: <value>
Uniqueid: <value>
Duration: <value>
```

Arguments

* `Meetme` - The identifier for the MeetMe conference.
* `Usernum` - The identifier of the MeetMe user who joined.
* `Channel`
* `ChannelState` - A numeric code for the channel's current state, related to ChannelStateDesc
* `ChannelStateDesc`
    * `Down`
    * `Rsrvd`
    * `OffHook`
    * `Dialing`
    * `Ring`
    * `Ringing`
    * `Up`
    * `Busy`
    * `Dialing Offhook`
    * `Pre-ring`
    * `Unknown`
* `CallerIDNum`
* `CallerIDName`
* `ConnectedLineNum`
* `ConnectedLineName`
* `AccountCode`
* `Context`
* `Exten`
* `Priority`
* `Uniqueid`
* `Duration` - The length of time in seconds that the Meetme user was in the conference.

See Also

* Asterisk 12 ManagerEvent_MeetmeJoin

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_MeetmeMute

MeetmeMute

Synopsis

Raised when a MeetMe user is muted or unmuted.

Description

Syntax

```
Action:
Meetme: <value>
Usernum: <value>
Channel: <value>
ChannelState: <value>
ChannelStateDesc: <value>
CallerIDNum: <value>
CallerIDName: <value>
ConnectedLineNum: <value>
ConnectedLineName: <value>
AccountCode: <value>
Context: <value>
Exten: <value>
Priority: <value>
Uniqueid: <value>
Duration: <value>
Status: <value>
```

Arguments

- `Meetme` - The identifier for the MeetMe conference.
- `Usernum` - The identifier of the MeetMe user who joined.
- `Channel`
- `ChannelState` - A numeric code for the channel's current state, related to ChannelStateDesc
- `ChannelStateDesc`
    - `Down`
    - `Rsrvd`
    - `OffHook`
    - `Dialing`
    - `Ring`
    - `Ringing`
    - `Up`
    - `Busy`
    - `Dialing Offhook`
    - `Pre-ring`
    - `Unknown`
- `CallerIDNum`
- `CallerIDName`
- `ConnectedLineNum`
- `ConnectedLineName`
- `AccountCode`
- `Context`
- `Exten`
- `Priority`
- `Uniqueid`
- `Duration` - The length of time in seconds that the Meetme user has been in the conference at the time of this event.
- `Status`
    - `on`
    - `off`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_MeetmeTalking

MeetmeTalking

Synopsis

Raised when a MeetMe user begins or ends talking.

Description

Syntax

```
Action:
Meetme: <value>
Usernum: <value>
Channel: <value>
ChannelState: <value>
ChannelStateDesc: <value>
CallerIDNum: <value>
CallerIDName: <value>
ConnectedLineNum: <value>
ConnectedLineName: <value>
AccountCode: <value>
Context: <value>
Exten: <value>
Priority: <value>
Uniqueid: <value>
Duration: <value>
Status: <value>
```

Arguments

- `Meetme` - The identifier for the MeetMe conference.
- `Usernum` - The identifier of the MeetMe user who joined.
- `Channel`
- `ChannelState` - A numeric code for the channel's current state, related to ChannelStateDesc
- `ChannelStateDesc`
    - `Down`
    - `Rsrvd`
    - `OffHook`
    - `Dialing`
    - `Ring`
    - `Ringing`
    - `Up`
    - `Busy`
    - `Dialing Offhook`
    - `Pre-ring`
    - `Unknown`
- `CallerIDNum`
- `CallerIDName`
- `ConnectedLineNum`
- `ConnectedLineName`
- `AccountCode`
- `Context`
- `Exten`
- `Priority`

- Uniqueid
- Duration - The length of time in seconds that the Meetme user has been in the conference at the time of this event.
- Status
    - on
    - off

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_MeetmeTalkRequest

MeetmeTalkRequest

Synopsis

Raised when a MeetMe user has started talking.

Description

Syntax

```
Action:
Meetme: <value>
Usernum: <value>
Channel: <value>
ChannelState: <value>
ChannelStateDesc: <value>
CallerIDNum: <value>
CallerIDName: <value>
ConnectedLineNum: <value>
ConnectedLineName: <value>
AccountCode: <value>
Context: <value>
Exten: <value>
Priority: <value>
Uniqueid: <value>
Duration: <value>
Status: <value>
```

Arguments

- Meetme - The identifier for the MeetMe conference.
- Usernum - The identifier of the MeetMe user who joined.
- Channel
- ChannelState - A numeric code for the channel's current state, related to ChannelStateDesc
- ChannelStateDesc
    - Down
    - Rsrvd
    - OffHook
    - Dialing
    - Ring
    - Ringing
    - Up
    - Busy

        - `Dialing Offhook`
        - `Pre-ring`
        - `Unknown`
- `CallerIDNum`
- `CallerIDName`
- `ConnectedLineNum`
- `ConnectedLineName`
- `AccountCode`
- `Context`
- `Exten`
- `Priority`
- `Uniqueid`
- `Duration` - The length of time in seconds that the Meetme user has been in the conference at the time of this event.
- `Status`
    - `on`
    - `off`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_MessageWaiting

MessageWaiting

Synopsis

Raised when the state of messages in a voicemail mailbox has changed or when a channel has finished interacting with a mailbox.

Description

> **Note**
> The Channel related parameters are only present if a channel was involved in the manipulation of a mailbox. If no channel is involved, the parameters are not included with the event.

Syntax

```
Action:
Channel: <value>
ChannelState: <value>
ChannelStateDesc: <value>
CallerIDNum: <value>
CallerIDName: <value>
ConnectedLineNum: <value>
ConnectedLineName: <value>
AccountCode: <value>
Context: <value>
Exten: <value>
Priority: <value>
Uniqueid: <value>
Mailbox: <value>
Waiting: <value>
New: <value>
Old: <value>
```

Arguments

- `Channel`
- `ChannelState` - A numeric code for the channel's current state, related to ChannelStateDesc
- `ChannelStateDesc`
  - `Down`
  - `Rsrvd`
  - `OffHook`
  - `Dialing`
  - `Ring`
  - `Ringing`
  - `Up`
  - `Busy`
  - `Dialing Offhook`
  - `Pre-ring`
  - `Unknown`
- `CallerIDNum`
- `CallerIDName`
- `ConnectedLineNum`
- `ConnectedLineName`
- `AccountCode`
- `Context`
- `Exten`
- `Priority`
- `Uniqueid`
- `Mailbox` - The mailbox with the new message, specified as `mailbox@context`
- `Waiting` - Whether or not the mailbox has messages waiting for it.
- `New` - The number of new messages.
- `Old` - The number of old messages.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_MiniVoiceMail

MiniVoiceMail

Synopsis

Raised when a notification is sent out by a MiniVoiceMail application

Description

Syntax

```
Action: 
Channel: <value>
ChannelState: <value>
ChannelStateDesc: <value>
CallerIDNum: <value>
CallerIDName: <value>
ConnectedLineNum: <value>
ConnectedLineName: <value>
AccountCode: <value>
Context: <value>
Exten: <value>
Priority: <value>
Uniqueid: <value>
Action: <value>
Mailbox: <value>
Counter: <value>
```

Arguments

- `Channel`
- `ChannelState` - A numeric code for the channel's current state, related to ChannelStateDesc
- `ChannelStateDesc`
    - `Down`
    - `Rsrvd`
    - `OffHook`
    - `Dialing`
    - `Ring`
    - `Ringing`
    - `Up`
    - `Busy`
    - `Dialing Offhook`
    - `Pre-ring`
    - `Unknown`
- `CallerIDNum`
- `CallerIDName`
- `ConnectedLineNum`
- `ConnectedLineName`
- `AccountCode`
- `Context`
- `Exten`
- `Priority`
- `Uniqueid`
- `Action` - What action was taken. Currently, this will always be `SentNotification`
- `Mailbox` - The mailbox that the notification was about, specified as `mailbox@context`
- `Counter` - A message counter derived from the `MVM_COUNTER` channel variable.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_MonitorStart

MonitorStart

Synopsis

Raised when monitoring has started on a channel.

Description

Syntax

```
Action:
Channel: <value>
ChannelState: <value>
ChannelStateDesc: <value>
CallerIDNum: <value>
CallerIDName: <value>
ConnectedLineNum: <value>
ConnectedLineName: <value>
AccountCode: <value>
Context: <value>
Exten: <value>
Priority: <value>
Uniqueid: <value>
```

Arguments

- `Channel`
- `ChannelState` - A numeric code for the channel's current state, related to ChannelStateDesc
- `ChannelStateDesc`
    - `Down`
    - `Rsrvd`
    - `OffHook`
    - `Dialing`
    - `Ring`
    - `Ringing`
    - `Up`
    - `Busy`
    - `Dialing Offhook`
    - `Pre-ring`
    - `Unknown`
- `CallerIDNum`
- `CallerIDName`
- `ConnectedLineNum`
- `ConnectedLineName`
- `AccountCode`
- `Context`
- `Exten`
- `Priority`
- `Uniqueid`

See Also

- Asterisk 12 ManagerEvent_MonitorStop
- Asterisk 12 Application_Monitor
- Asterisk 12 ManagerAction_Monitor

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_MonitorStop

MonitorStop

Synopsis

Raised when monitoring has stopped on a channel.

Description

Syntax

```
Action:
Channel: <value>
ChannelState: <value>
ChannelStateDesc: <value>
CallerIDNum: <value>
CallerIDName: <value>
ConnectedLineNum: <value>
ConnectedLineName: <value>
AccountCode: <value>
Context: <value>
Exten: <value>
Priority: <value>
Uniqueid: <value>
```

Arguments

- `Channel`
- `ChannelState` - A numeric code for the channel's current state, related to ChannelStateDesc
- `ChannelStateDesc`
    - `Down`
    - `Rsrvd`
    - `OffHook`
    - `Dialing`
    - `Ring`
    - `Ringing`
    - `Up`
    - `Busy`
    - `Dialing Offhook`
    - `Pre-ring`
    - `Unknown`
- `CallerIDNum`
- `CallerIDName`
- `ConnectedLineNum`
- `ConnectedLineName`
- `AccountCode`
- `Context`

* Exten
* Priority
* Uniqueid

See Also

* Asterisk 12 ManagerEvent_MonitorStart
* Asterisk 12 Application_StopMonitor
* Asterisk 12 ManagerAction_StopMonitor

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_MusicOnHoldStart

MusicOnHoldStart

Synopsis

Raised when music on hold has started on a channel.

Description

Syntax

```
Action:
Channel: <value>
ChannelState: <value>
ChannelStateDesc: <value>
CallerIDNum: <value>
CallerIDName: <value>
ConnectedLineNum: <value>
ConnectedLineName: <value>
AccountCode: <value>
Context: <value>
Exten: <value>
Priority: <value>
Uniqueid: <value>
Class: <value>
```

Arguments

* `Channel`
* `ChannelState` - A numeric code for the channel's current state, related to ChannelStateDesc
* `ChannelStateDesc`
    * `Down`
    * `Rsrvd`
    * `OffHook`
    * `Dialing`
    * `Ring`
    * `Ringing`
    * `Up`
    * `Busy`
    * `Dialing Offhook`
    * `Pre-ring`
    * `Unknown`
* `CallerIDNum`

- `CallerIDName`
- `ConnectedLineNum`
- `ConnectedLineName`
- `AccountCode`
- `Context`
- `Exten`
- `Priority`
- `Uniqueid`
- `Class` - The class of music being played on the channel

See Also

- Asterisk 12 ManagerEvent_MusicOnHoldStop
- Asterisk 12 Application_MusicOnHold

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_MusicOnHoldStop

MusicOnHoldStop

Synopsis

Raised when music on hold has stopped on a channel.

Description

Syntax

```
Action:
Channel: <value>
ChannelState: <value>
ChannelStateDesc: <value>
CallerIDNum: <value>
CallerIDName: <value>
ConnectedLineNum: <value>
ConnectedLineName: <value>
AccountCode: <value>
Context: <value>
Exten: <value>
Priority: <value>
Uniqueid: <value>
```

Arguments

- `Channel`
- `ChannelState` - A numeric code for the channel's current state, related to ChannelStateDesc
- `ChannelStateDesc`
    - `Down`
    - `Rsrvd`
    - `OffHook`
    - `Dialing`
    - `Ring`
    - `Ringing`
    - `Up`
    - `Busy`

        - `Dialing Offhook`
        - `Pre-ring`
        - `Unknown`
- `CallerIDNum`
- `CallerIDName`
- `ConnectedLineNum`
- `ConnectedLineName`
- `AccountCode`
- `Context`
- `Exten`
- `Priority`
- `Uniqueid`

See Also

- Asterisk 12 ManagerEvent_MusicOnHoldStart
- Asterisk 12 Application_StopMusicOnHold

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_NewAccountCode

NewAccountCode

Synopsis

Raised when a Channel's AccountCode is changed.

Description

Syntax

```
Action:
Channel: <value>
ChannelState: <value>
ChannelStateDesc: <value>
CallerIDNum: <value>
CallerIDName: <value>
ConnectedLineNum: <value>
ConnectedLineName: <value>
AccountCode: <value>
Context: <value>
Exten: <value>
Priority: <value>
Uniqueid: <value>
OldAccountCode: <value>
```

Arguments

- `Channel`
- `ChannelState` - A numeric code for the channel's current state, related to ChannelStateDesc
- `ChannelStateDesc`
    - `Down`
    - `Rsrvd`
    - `OffHook`
    - `Dialing`

        - Ring
        - Ringing
        - Up
        - Busy
        - Dialing Offhook
        - Pre-ring
        - Unknown
- CallerIDNum
- CallerIDName
- ConnectedLineNum
- ConnectedLineName
- AccountCode
- Context
- Exten
- Priority
- Uniqueid
- OldAccountCode - The channel's previous account code

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_NewCallerid

NewCallerid

Synopsis

Raised when a channel receives new Caller ID information.

Description

Syntax

```
Action: 
Channel: <value>
ChannelState: <value>
ChannelStateDesc: <value>
CallerIDNum: <value>
CallerIDName: <value>
ConnectedLineNum: <value>
ConnectedLineName: <value>
AccountCode: <value>
Context: <value>
Exten: <value>
Priority: <value>
Uniqueid: <value>
CID-CallingPres: <value>
```

Arguments

- Channel
- ChannelState - A numeric code for the channel's current state, related to ChannelStateDesc
- ChannelStateDesc
    - Down

    - Rsrvd
    - OffHook
    - Dialing
    - Ring
    - Ringing
    - Up
    - Busy
    - Dialing Offhook
    - Pre-ring
    - Unknown
- CallerIDNum
- CallerIDName
- ConnectedLineNum
- ConnectedLineName
- AccountCode
- Context
- Exten
- Priority
- Uniqueid
- CID-CallingPres - A description of the Caller ID presentation.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_Newchannel

Newchannel

Synopsis

Raised when a new channel is created.

Description

Syntax

```
Action:
Channel: <value>
ChannelState: <value>
ChannelStateDesc: <value>
CallerIDNum: <value>
CallerIDName: <value>
ConnectedLineNum: <value>
ConnectedLineName: <value>
AccountCode: <value>
Context: <value>
Exten: <value>
Priority: <value>
Uniqueid: <value>
```

Arguments

- Channel
- ChannelState - A numeric code for the channel's current state, related to ChannelStateDesc

- `ChannelStateDesc`
  - `Down`
  - `Rsrvd`
  - `OffHook`
  - `Dialing`
  - `Ring`
  - `Ringing`
  - `Up`
  - `Busy`
  - `Dialing Offhook`
  - `Pre-ring`
  - `Unknown`
- `CallerIDNum`
- `CallerIDName`
- `ConnectedLineNum`
- `ConnectedLineName`
- `AccountCode`
- `Context`
- `Exten`
- `Priority`
- `Uniqueid`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_NewExten

NewExten

Synopsis

Raised when a channel enters a new context, extension, priority.

Description

Syntax

```
Action:
Channel: <value>
ChannelState: <value>
ChannelStateDesc: <value>
CallerIDNum: <value>
CallerIDName: <value>
ConnectedLineNum: <value>
ConnectedLineName: <value>
AccountCode: <value>
Context: <value>
Exten: <value>
Priority: <value>
Uniqueid: <value>
Extension: <value>
Application: <value>
AppData: <value>
```

Arguments

- `Channel`
- `ChannelState` - A numeric code for the channel's current state, related to ChannelStateDesc
- `ChannelStateDesc`
    - `Down`
    - `Rsrvd`
    - `OffHook`
    - `Dialing`
    - `Ring`
    - `Ringing`
    - `Up`
    - `Busy`
    - `Dialing Offhook`
    - `Pre-ring`
    - `Unknown`
- `CallerIDNum`
- `CallerIDName`
- `ConnectedLineNum`
- `ConnectedLineName`
- `AccountCode`
- `Context`
- `Exten`
- `Priority`
- `Uniqueid`
- `Extension` - Deprecated in 12, but kept for backward compatability. Please use 'Exten' instead.
- `Application` - The application about to be executed.
- `AppData` - The data to be passed to the application.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_Newstate

Newstate

Synopsis

Raised when a channel's state changes.

Description

Syntax

```
Action:
Channel: <value>
ChannelState: <value>
ChannelStateDesc: <value>
CallerIDNum: <value>
CallerIDName: <value>
ConnectedLineNum: <value>
ConnectedLineName: <value>
AccountCode: <value>
Context: <value>
Exten: <value>
Priority: <value>
Uniqueid: <value>
```

Arguments

- `Channel`
- `ChannelState` - A numeric code for the channel's current state, related to ChannelStateDesc
- `ChannelStateDesc`
  - `Down`
  - `Rsrvd`
  - `OffHook`
  - `Dialing`
  - `Ring`
  - `Ringing`
  - `Up`
  - `Busy`
  - `Dialing Offhook`
  - `Pre-ring`
  - `Unknown`
- `CallerIDNum`
- `CallerIDName`
- `ConnectedLineNum`
- `ConnectedLineName`
- `AccountCode`
- `Context`
- `Exten`
- `Priority`
- `Uniqueid`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_OriginateResponse

OriginateResponse

Synopsis

Raised in response to an Originate command.

Description

Syntax

```
Action:
[ActionID:] <value>
Resonse: <value>
Channel: <value>
Context: <value>
Exten: <value>
Reason: <value>
Uniqueid: <value>
CallerIDNum: <value>
CallerIDName: <value>
```

Arguments

- `ActionID`
- `Resonse`
    - `Failure`
    - `Success`
- `Channel`
- `Context`
- `Exten`
- `Reason`
- `Uniqueid`
- `CallerIDNum`
- `CallerIDName`

See Also

- Asterisk 12 ManagerAction_Originate

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_ParkedCall

ParkedCall

Synopsis

Raised when a channel is parked.

Description

Syntax

```
Action:
ParkeeChannel: <value>
ParkeeChannelState: <value>
ParkeeChannelStateDesc: <value>
ParkeeCallerIDNum: <value>
ParkeeCallerIDName: <value>
ParkeeConnectedLineNum: <value>
ParkeeConnectedLineName: <value>
ParkeeAccountCode: <value>
ParkeeContext: <value>
ParkeeExten: <value>
ParkeePriority: <value>
ParkeeUniqueid: <value>
ParkerDialString: <value>
Parkinglot: <value>
ParkingSpace: <value>
ParkingTimeout: <value>
ParkingDuration: <value>
```

Arguments

- `ParkeeChannel`
- `ParkeeChannelState` - A numeric code for the channel's current state, related to ParkeeChannelStateDesc
- `ParkeeChannelStateDesc`
    - `Down`
    - `Rsrvd`
    - `OffHook`
    - `Dialing`
    - `Ring`
    - `Ringing`
    - `Up`
    - `Busy`
    - `Dialing Offhook`
    - `Pre-ring`
    - `Unknown`
- `ParkeeCallerIDNum`
- `ParkeeCallerIDName`
- `ParkeeConnectedLineNum`
- `ParkeeConnectedLineName`
- `ParkeeAccountCode`
- `ParkeeContext`
- `ParkeeExten`
- `ParkeePriority`
- `ParkeeUniqueid`
- `ParkerDialString` - Dial String that can be used to call back the parker on ParkingTimeout.
- `Parkinglot` - Name of the parking lot that the parkee is parked in
- `ParkingSpace` - Parking Space that the parkee is parked in
- `ParkingTimeout` - Time remaining until the parkee is forcefully removed from parking in seconds
- `ParkingDuration` - Time the parkee has been in the parking bridge (in seconds)

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_ParkedCallGiveUp

ParkedCallGiveUp

Synopsis

Raised when a channel leaves a parking lot because it hung up without being answered.

Description

Syntax

```
Action:
ParkeeChannel: <value>
ParkeeChannelState: <value>
ParkeeChannelStateDesc: <value>
ParkeeCallerIDNum: <value>
ParkeeCallerIDName: <value>
ParkeeConnectedLineNum: <value>
ParkeeConnectedLineName: <value>
ParkeeAccountCode: <value>
ParkeeContext: <value>
ParkeeExten: <value>
ParkeePriority: <value>
ParkeeUniqueid: <value>
ParkerChannel: <value>
ParkerChannelState: <value>
ParkerChannelStateDesc: <value>
ParkerCallerIDNum: <value>
ParkerCallerIDName: <value>
ParkerConnectedLineNum: <value>
ParkerConnectedLineName: <value>
ParkerAccountCode: <value>
ParkerContext: <value>
ParkerExten: <value>
ParkerPriority: <value>
ParkerUniqueid: <value>
ParkerDialString: <value>
Parkinglot: <value>
ParkingSpace: <value>
ParkingTimeout: <value>
ParkingDuration: <value>
```

Arguments

- `ParkeeChannel`
- `ParkeeChannelState` - A numeric code for the channel's current state, related to ParkeeChannelStateDesc
- `ParkeeChannelStateDesc`
  - `Down`
  - `Rsrvd`
  - `OffHook`
  - `Dialing`
  - `Ring`
  - `Ringing`

        - Up
        - Busy
        - Dialing Offhook
        - Pre-ring
        - Unknown
- ParkeeCallerIDNum
- ParkeeCallerIDName
- ParkeeConnectedLineNum
- ParkeeConnectedLineName
- ParkeeAccountCode
- ParkeeContext
- ParkeeExten
- ParkeePriority
- ParkeeUniqueid
- ParkerChannel
- ParkerChannelState - A numeric code for the channel's current state, related to ParkerChannelStateDesc
- ParkerChannelStateDesc
    - Down
    - Rsrvd
    - OffHook
    - Dialing
    - Ring
    - Ringing
    - Up
    - Busy
    - Dialing Offhook
    - Pre-ring
    - Unknown
- ParkerCallerIDNum
- ParkerCallerIDName
- ParkerConnectedLineNum
- ParkerConnectedLineName
- ParkerAccountCode
- ParkerContext
- ParkerExten
- ParkerPriority
- ParkerUniqueid
- ParkerDialString - Dial String that can be used to call back the parker on ParkingTimeout.
- Parkinglot - Name of the parking lot that the parkee is parked in
- ParkingSpace - Parking Space that the parkee is parked in
- ParkingTimeout - Time remaining until the parkee is forcefully removed from parking in seconds
- ParkingDuration - Time the parkee has been in the parking bridge (in seconds)

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_ParkedCallTimeOut

ParkedCallTimeOut

Synopsis

Raised when a channel leaves a parking lot due to reaching the time limit of being parked.

Description

Syntax

```
Action:
ParkeeChannel: <value>
ParkeeChannelState: <value>
ParkeeChannelStateDesc: <value>
ParkeeCallerIDNum: <value>
ParkeeCallerIDName: <value>
ParkeeConnectedLineNum: <value>
ParkeeConnectedLineName: <value>
ParkeeAccountCode: <value>
ParkeeContext: <value>
ParkeeExten: <value>
ParkeePriority: <value>
ParkeeUniqueid: <value>
ParkerChannel: <value>
ParkerChannelState: <value>
ParkerChannelStateDesc: <value>
ParkerCallerIDNum: <value>
ParkerCallerIDName: <value>
ParkerConnectedLineNum: <value>
ParkerConnectedLineName: <value>
ParkerAccountCode: <value>
ParkerContext: <value>
ParkerExten: <value>
ParkerPriority: <value>
ParkerUniqueid: <value>
ParkerDialString: <value>
Parkinglot: <value>
ParkingSpace: <value>
ParkingTimeout: <value>
ParkingDuration: <value>
```

Arguments

- `ParkeeChannel`
- `ParkeeChannelState` - A numeric code for the channel's current state, related to ParkeeChannelStateDesc
- `ParkeeChannelStateDesc`
    - `Down`
    - `Rsrvd`
    - `OffHook`
    - `Dialing`
    - `Ring`
    - `Ringing`
    - `Up`
    - `Busy`
    - `Dialing Offhook`
    - `Pre-ring`
    - `Unknown`
- `ParkeeCallerIDNum`
- `ParkeeCallerIDName`
- `ParkeeConnectedLineNum`
- `ParkeeConnectedLineName`
- `ParkeeAccountCode`
- `ParkeeContext`

- `ParkeeExten`
- `ParkeePriority`
- `ParkeeUniqueid`
- `ParkerChannel`
- `ParkerChannelState` - A numeric code for the channel's current state, related to ParkerChannelStateDesc
- `ParkerChannelStateDesc`
    - `Down`
    - `Rsrvd`
    - `OffHook`
    - `Dialing`
    - `Ring`
    - `Ringing`
    - `Up`
    - `Busy`
    - `Dialing Offhook`
    - `Pre-ring`
    - `Unknown`
- `ParkerCallerIDNum`
- `ParkerCallerIDName`
- `ParkerConnectedLineNum`
- `ParkerConnectedLineName`
- `ParkerAccountCode`
- `ParkerContext`
- `ParkerExten`
- `ParkerPriority`
- `ParkerUniqueid`
- `ParkerDialString` - Dial String that can be used to call back the parker on ParkingTimeout.
- `Parkinglot` - Name of the parking lot that the parkee is parked in
- `ParkingSpace` - Parking Space that the parkee is parked in
- `ParkingTimeout` - Time remaining until the parkee is forcefully removed from parking in seconds
- `ParkingDuration` - Time the parkee has been in the parking bridge (in seconds)

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_PeerStatus

PeerStatus

Synopsis

Raised when the state of a peer changes.

Description

Syntax

```
Action:
ChannelType: <value>
Peer: <value>
PeerStatus: <value>
Cause: <value>
Address: <value>
Port: <value>
Time: <value>
```

Arguments

- `ChannelType` - The channel technology of the peer.
- `Peer` - The name of the peer (including channel technology).
- `PeerStatus` - New status of the peer.
    - `Unknown`
    - `Registered`
    - `Unregistered`
    - `Rejected`
    - `Lagged`
- `Cause` - The reason the status has changed.
- `Address` - New address of the peer.
- `Port` - New port for the peer.
- `Time` - Time it takes to reach the peer and receive a response.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_Pickup

Pickup

Synopsis

Raised when a call pickup occurs.

Description

Syntax

```
Action:
Channel: <value>
ChannelState: <value>
ChannelStateDesc: <value>
CallerIDNum: <value>
CallerIDName: <value>
ConnectedLineNum: <value>
ConnectedLineName: <value>
AccountCode: <value>
Context: <value>
Exten: <value>
Priority: <value>
Uniqueid: <value>
TargetChannel: <value>
TargetChannelState: <value>
TargetChannelStateDesc: <value>
TargetCallerIDNum: <value>
TargetCallerIDName: <value>
TargetConnectedLineNum: <value>
TargetConnectedLineName: <value>
TargetAccountCode: <value>
TargetContext: <value>
TargetExten: <value>
TargetPriority: <value>
TargetUniqueid: <value>
```

Arguments

- `Channel`
- `ChannelState` - A numeric code for the channel's current state, related to ChannelStateDesc
- `ChannelStateDesc`
    - `Down`
    - `Rsrvd`
    - `OffHook`
    - `Dialing`
    - `Ring`
    - `Ringing`
    - `Up`
    - `Busy`
    - `Dialing Offhook`
    - `Pre-ring`
    - `Unknown`
- `CallerIDNum`
- `CallerIDName`
- `ConnectedLineNum`
- `ConnectedLineName`
- `AccountCode`
- `Context`
- `Exten`
- `Priority`
- `Uniqueid`
- `TargetChannel`
- `TargetChannelState` - A numeric code for the channel's current state, related to TargetChannelStateDesc
- `TargetChannelStateDesc`

    - Down
    - Rsrvd
    - OffHook
    - Dialing
    - Ring
    - Ringing
    - Up
    - Busy
    - Dialing Offhook
    - Pre-ring
    - Unknown
- TargetCallerIDNum
- TargetCallerIDName
- TargetConnectedLineNum
- TargetConnectedLineName
- TargetAccountCode
- TargetContext
- TargetExten
- TargetPriority
- TargetUniqueid

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_QueueCallerAbandon

QueueCallerAbandon

Synopsis

Raised when a caller abandons the queue.

Description

Syntax

```
Action:
Channel: <value>
ChannelState: <value>
ChannelStateDesc: <value>
CallerIDNum: <value>
CallerIDName: <value>
ConnectedLineNum: <value>
ConnectedLineName: <value>
AccountCode: <value>
Context: <value>
Exten: <value>
Priority: <value>
Uniqueid: <value>
Queue: <value>
Position: <value>
OriginalPosition: <value>
HoldTime: <value>
```

Arguments

- `Channel`
- `ChannelState` - A numeric code for the channel's current state, related to ChannelStateDesc
- `ChannelStateDesc`
    - `Down`
    - `Rsrvd`
    - `OffHook`
    - `Dialing`
    - `Ring`
    - `Ringing`
    - `Up`
    - `Busy`
    - `Dialing Offhook`
    - `Pre-ring`
    - `Unknown`
- `CallerIDNum`
- `CallerIDName`
- `ConnectedLineNum`
- `ConnectedLineName`
- `AccountCode`
- `Context`
- `Exten`
- `Priority`
- `Uniqueid`
- `Queue` - The name of the queue.
- `Position` - This channel's current position in the queue.
- `OriginalPosition` - The channel's original position in the queue.
- `HoldTime` - The time the channel was in the queue, expressed in seconds since 00:00, Jan 1, 1970 UTC.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_QueueCallerJoin

QueueCallerJoin

Synopsis

Raised when a caller joins a Queue.

Description

Syntax

```
Action:
Channel: <value>
ChannelState: <value>
ChannelStateDesc: <value>
CallerIDNum: <value>
CallerIDName: <value>
ConnectedLineNum: <value>
ConnectedLineName: <value>
AccountCode: <value>
Context: <value>
Exten: <value>
Priority: <value>
Uniqueid: <value>
Queue: <value>
Position: <value>
Count: <value>
```

Arguments

- `Channel`
- `ChannelState` - A numeric code for the channel's current state, related to ChannelStateDesc
- `ChannelStateDesc`
    - `Down`
    - `Rsrvd`
    - `OffHook`
    - `Dialing`
    - `Ring`
    - `Ringing`
    - `Up`
    - `Busy`
    - `Dialing Offhook`
    - `Pre-ring`
    - `Unknown`
- `CallerIDNum`
- `CallerIDName`
- `ConnectedLineNum`
- `ConnectedLineName`
- `AccountCode`
- `Context`
- `Exten`
- `Priority`
- `Uniqueid`
- `Queue` - The name of the queue.
- `Position` - This channel's current position in the queue.
- `Count` - The total number of channels in the queue.

See Also

- Asterisk 12 ManagerEvent_QueueCallerLeave
- Asterisk 12 Application_Queue

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_QueueCallerLeave

QueueCallerLeave

Synopsis

Raised when a caller leaves a Queue.

Description

Syntax

```
Action:
Channel: <value>
ChannelState: <value>
ChannelStateDesc: <value>
CallerIDNum: <value>
CallerIDName: <value>
ConnectedLineNum: <value>
ConnectedLineName: <value>
AccountCode: <value>
Context: <value>
Exten: <value>
Priority: <value>
Uniqueid: <value>
Queue: <value>
Count: <value>
Position: <value>
```

Arguments

- `Channel`
- `ChannelState` - A numeric code for the channel's current state, related to ChannelStateDesc
- `ChannelStateDesc`
    - `Down`
    - `Rsrvd`
    - `OffHook`
    - `Dialing`
    - `Ring`
    - `Ringing`
    - `Up`
    - `Busy`
    - `Dialing Offhook`
    - `Pre-ring`
    - `Unknown`
- `CallerIDNum`
- `CallerIDName`
- `ConnectedLineNum`
- `ConnectedLineName`
- `AccountCode`
- `Context`
- `Exten`
- `Priority`
- `Uniqueid`
- `Queue` - The name of the queue.
- `Count` - The total number of channels in the queue.
- `Position` - This channel's current position in the queue.

See Also

- Asterisk 12 ManagerEvent_QueueCallerJoin

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_QueueMemberAdded

QueueMemberAdded

Synopsis

Raised when a member is added to the queue.

Description

Syntax

```
Action:
Queue: <value>
MemberName: <value>
Interface: <value>
StateInterface: <value>
Membership: <value>
Penalty: <value>
CallsTaken: <value>
LastCall: <value>
Status: <value>
Paused: <value>
Ringinuse: <value>
```

Arguments

- `Queue` - The name of the queue.
- `MemberName` - The name of the queue member.
- `Interface` - The queue member's channel technology or location.
- `StateInterface` - Channel technology or location from which to read device state changes.
- `Membership`
  - `dynamic`
  - `realtime`
  - `static`
- `Penalty` - The penalty associated with the queue member.
- `CallsTaken` - The number of calls this queue member has serviced.
- `LastCall` - The time this member last took a call, expressed in seconds since 00:00, Jan 1, 1970 UTC.
- `Status` - The numeric device state status of the queue member.
  - `0` - AST_DEVICE_UNKNOWN
  - `1` - AST_DEVICE_NOT_INUSE
  - `2` - AST_DEVICE_INUSE
  - `3` - AST_DEVICE_BUSY
  - `4` - AST_DEVICE_INVALID
  - `5` - AST_DEVICE_UNAVAILABLE
  - `6` - AST_DEVICE_RINGING
  - `7` - AST_DEVICE_RINGINUSE
  - `8` - AST_DEVICE_ONHOLD
- `Paused`

    - `0`
    - `1`
- `Ringinuse`
    - `0`
    - `1`

See Also

- Asterisk 12 ManagerEvent_QueueMemberRemoved
- Asterisk 12 Application_AddQueueMember

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_QueueMemberPaused

QueueMemberPaused

Synopsis

Raised when a member is paused/unpaused in the queue.

Description

Syntax

```
Action:
Queue: <value>
MemberName: <value>
Interface: <value>
StateInterface: <value>
Membership: <value>
Penalty: <value>
CallsTaken: <value>
LastCall: <value>
Status: <value>
Paused: <value>
Ringinuse: <value>
Reason: <value>
```

Arguments

- `Queue` - The name of the queue.
- `MemberName` - The name of the queue member.
- `Interface` - The queue member's channel technology or location.
- `StateInterface` - Channel technology or location from which to read device state changes.
- `Membership`
    - `dynamic`
    - `realtime`
    - `static`
- `Penalty` - The penalty associated with the queue member.
- `CallsTaken` - The number of calls this queue member has serviced.
- `LastCall` - The time this member last took a call, expressed in seconds since 00:00, Jan 1, 1970 UTC.
- `Status` - The numeric device state status of the queue member.
    - `0` - AST_DEVICE_UNKNOWN
    - `1` - AST_DEVICE_NOT_INUSE

        - 2 - AST_DEVICE_INUSE
        - 3 - AST_DEVICE_BUSY
        - 4 - AST_DEVICE_INVALID
        - 5 - AST_DEVICE_UNAVAILABLE
        - 6 - AST_DEVICE_RINGING
        - 7 - AST_DEVICE_RINGINUSE
        - 8 - AST_DEVICE_ONHOLD
- `Paused`
    - `0`
    - `1`
- `Ringinuse`
    - `0`
    - `1`
- `Reason` - The reason a member was paused.

See Also

- Asterisk 12 Application_PauseQueueMember
- Asterisk 12 Application_UnPauseQueueMember

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_QueueMemberPenalty

QueueMemberPenalty

Synopsis

Raised when a member's penalty is changed.

Description

Syntax

```
Action:
Queue: <value>
MemberName: <value>
Interface: <value>
StateInterface: <value>
Membership: <value>
Penalty: <value>
CallsTaken: <value>
LastCall: <value>
Status: <value>
Paused: <value>
Ringinuse: <value>
```

Arguments

- `Queue` - The name of the queue.
- `MemberName` - The name of the queue member.
- `Interface` - The queue member's channel technology or location.
- `StateInterface` - Channel technology or location from which to read device state changes.
- `Membership`
    - `dynamic`

        - `realtime`
        - `static`
- `Penalty` - The penalty associated with the queue member.
- `CallsTaken` - The number of calls this queue member has serviced.
- `LastCall` - The time this member last took a call, expressed in seconds since 00:00, Jan 1, 1970 UTC.
- `Status` - The numeric device state status of the queue member.
    - `0` - AST_DEVICE_UNKNOWN
    - `1` - AST_DEVICE_NOT_INUSE
    - `2` - AST_DEVICE_INUSE
    - `3` - AST_DEVICE_BUSY
    - `4` - AST_DEVICE_INVALID
    - `5` - AST_DEVICE_UNAVAILABLE
    - `6` - AST_DEVICE_RINGING
    - `7` - AST_DEVICE_RINGINUSE
    - `8` - AST_DEVICE_ONHOLD
- `Paused`
    - `0`
    - `1`
- `Ringinuse`
    - `0`
    - `1`

See Also

- Asterisk 12 Function_QUEUE_MEMBER

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_QueueMemberRemoved

QueueMemberRemoved

Synopsis

Raised when a member is removed from the queue.

Description

Syntax

```
Action:
Queue: <value>
MemberName: <value>
Interface: <value>
StateInterface: <value>
Membership: <value>
Penalty: <value>
CallsTaken: <value>
LastCall: <value>
Status: <value>
Paused: <value>
Ringinuse: <value>
```

Arguments

- `Queue` - The name of the queue.
- `MemberName` - The name of the queue member.
- `Interface` - The queue member's channel technology or location.
- `StateInterface` - Channel technology or location from which to read device state changes.
- `Membership`
    - `dynamic`
    - `realtime`
    - `static`
- `Penalty` - The penalty associated with the queue member.
- `CallsTaken` - The number of calls this queue member has serviced.
- `LastCall` - The time this member last took a call, expressed in seconds since 00:00, Jan 1, 1970 UTC.
- `Status` - The numeric device state status of the queue member.
    - `0` - AST_DEVICE_UNKNOWN
    - `1` - AST_DEVICE_NOT_INUSE
    - `2` - AST_DEVICE_INUSE
    - `3` - AST_DEVICE_BUSY
    - `4` - AST_DEVICE_INVALID
    - `5` - AST_DEVICE_UNAVAILABLE
    - `6` - AST_DEVICE_RINGING
    - `7` - AST_DEVICE_RINGINUSE
    - `8` - AST_DEVICE_ONHOLD
- `Paused`
    - `0`
    - `1`
- `Ringinuse`
    - `0`
    - `1`

See Also

- Asterisk 12 ManagerEvent_QueueMemberAdded
- Asterisk 12 Application_RemoveQueueMember

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_QueueMemberRinginuse

QueueMemberRinginuse

Synopsis

Raised when a member's ringinuse setting is changed.

Description

Syntax

```
Action:
Queue: <value>
MemberName: <value>
Interface: <value>
StateInterface: <value>
Membership: <value>
Penalty: <value>
CallsTaken: <value>
LastCall: <value>
Status: <value>
Paused: <value>
Ringinuse: <value>
```

Arguments

- `Queue` - The name of the queue.
- `MemberName` - The name of the queue member.
- `Interface` - The queue member's channel technology or location.
- `StateInterface` - Channel technology or location from which to read device state changes.
- `Membership`
    - `dynamic`
    - `realtime`
    - `static`
- `Penalty` - The penalty associated with the queue member.
- `CallsTaken` - The number of calls this queue member has serviced.
- `LastCall` - The time this member last took a call, expressed in seconds since 00:00, Jan 1, 1970 UTC.
- `Status` - The numeric device state status of the queue member.
    - `0` - AST_DEVICE_UNKNOWN
    - `1` - AST_DEVICE_NOT_INUSE
    - `2` - AST_DEVICE_INUSE
    - `3` - AST_DEVICE_BUSY
    - `4` - AST_DEVICE_INVALID
    - `5` - AST_DEVICE_UNAVAILABLE
    - `6` - AST_DEVICE_RINGING
    - `7` - AST_DEVICE_RINGINUSE
    - `8` - AST_DEVICE_ONHOLD
- `Paused`
    - `0`
    - `1`
- `Ringinuse`
    - `0`
    - `1`

See Also

- Asterisk 12 Function_QUEUE_MEMBER

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_QueueMemberStatus

QueueMemberStatus

Synopsis

Raised when a Queue member's status has changed.

Description

Syntax

```
Action:
Queue: <value>
MemberName: <value>
Interface: <value>
StateInterface: <value>
Membership: <value>
Penalty: <value>
CallsTaken: <value>
LastCall: <value>
Status: <value>
Paused: <value>
Ringinuse: <value>
```

Arguments

- `Queue` - The name of the queue.
- `MemberName` - The name of the queue member.
- `Interface` - The queue member's channel technology or location.
- `StateInterface` - Channel technology or location from which to read device state changes.
- `Membership`
    - `dynamic`
    - `realtime`
    - `static`
- `Penalty` - The penalty associated with the queue member.
- `CallsTaken` - The number of calls this queue member has serviced.
- `LastCall` - The time this member last took a call, expressed in seconds since 00:00, Jan 1, 1970 UTC.
- `Status` - The numeric device state status of the queue member.
    - `0` - AST_DEVICE_UNKNOWN
    - `1` - AST_DEVICE_NOT_INUSE
    - `2` - AST_DEVICE_INUSE
    - `3` - AST_DEVICE_BUSY
    - `4` - AST_DEVICE_INVALID
    - `5` - AST_DEVICE_UNAVAILABLE
    - `6` - AST_DEVICE_RINGING
    - `7` - AST_DEVICE_RINGINUSE
    - `8` - AST_DEVICE_ONHOLD
- `Paused`
    - `0`
    - `1`
- `Ringinuse`
    - `0`
    - `1`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_ReceiveFAX

ReceiveFAX

Synopsis

Raised when a receive fax operation has completed.

Description

Syntax

```
Action: 
Channel: <value>
ChannelState: <value>
ChannelStateDesc: <value>
CallerIDNum: <value>
CallerIDName: <value>
ConnectedLineNum: <value>
ConnectedLineName: <value>
AccountCode: <value>
Context: <value>
Exten: <value>
Priority: <value>
Uniqueid: <value>
LocalStationID: <value>
RemoteStationID: <value>
PagesTransferred: <value>
Resolution: <value>
TransferRate: <value>
FileName: <value>
```

Arguments

- `Channel`
- `ChannelState` - A numeric code for the channel's current state, related to ChannelStateDesc
- `ChannelStateDesc`
    - `Down`
    - `Rsrvd`
    - `OffHook`
    - `Dialing`
    - `Ring`
    - `Ringing`
    - `Up`
    - `Busy`
    - `Dialing Offhook`
    - `Pre-ring`
    - `Unknown`
- `CallerIDNum`
- `CallerIDName`
- `ConnectedLineNum`
- `ConnectedLineName`
- `AccountCode`
- `Context`
- `Exten`
- `Priority`
- `Uniqueid`

* `LocalStationID` - The value of the `LOCALSTATIONID` channel variable
* `RemoteStationID` - The value of the `REMOTESTATIONID` channel variable
* `PagesTransferred` - The number of pages that have been transferred
* `Resolution` - The negotiated resolution
* `TransferRate` - The negotiated transfer rate
* `FileName` - The files being affected by the fax operation

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_Registry

Registry

Synopsis

Raised when an outbound registration completes.

Description

Syntax

```
Action:
ChannelType: <value>
Username: <value>
Domain: <value>
Status: <value>
Cause: <value>
```

Arguments

* `ChannelType` - The type of channel that was registered (or not).
* `Username` - The username portion of the registration.
* `Domain` - The address portion of the registration.
* `Status` - The status of the registration request.
    * `Registered`
    * `Unregistered`
    * `Rejected`
    * `Failed`
* `Cause` - What caused the rejection of the request, if available.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_Reload

Reload

Synopsis

Raised when a module has been reloaded in Asterisk.

Description

Syntax

```
Action: 
Module: <value>
Status: <value>
```

Arguments

- `Module` - The name of the module that was reloaded, or `All` if all modules were reloaded
- `Status` - The numeric status code denoting the success or failure of the reload request.
  - `0` - Success
  - `1` - Request queued
  - `2` - Module not found
  - `3` - Error
  - `4` - Reload already in progress
  - `5` - Module uninitialized
  - `6` - Reload not supported

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_Rename

Rename

Synopsis

Raised when the name of a channel is changed.

Description

Syntax

```
Action: 
Channel: <value>
Newname: <value>
Uniqueid: <value>
```

Arguments

- `Channel`
- `Newname`
- `Uniqueid`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_RTCPReceived

RTCPReceived

Synopsis

Raised when an RTCP packet is received.

Description

Syntax

```
Action: 
Channel: <value>
ChannelState: <value>
ChannelStateDesc: <value>
CallerIDNum: <value>
CallerIDName: <value>
ConnectedLineNum: <value>
ConnectedLineName: <value>
AccountCode: <value>
Context: <value>
Exten: <value>
Priority: <value>
Uniqueid: <value>
SSRC: <value>
PT: <value>
From: <value>
RTT: <value>
ReportCount: <value>
[SentNTP:] <value>
[SentRTP:] <value>
[SentPackets:] <value>
[SentOctets:] <value>
ReportXSourceSSRC: <value>
ReportXFractionLost: <value>
ReportXCumulativeLost: <value>
ReportXHighestSequence: <value>
ReportXSequenceNumberCycles: <value>
ReportXIAJitter: <value>
ReportXLSR: <value>
ReportXDLSR: <value>
```

Arguments

- `Channel`
- `ChannelState` - A numeric code for the channel's current state, related to ChannelStateDesc
- `ChannelStateDesc`
    - `Down`
    - `Rsrvd`
    - `OffHook`
    - `Dialing`
    - `Ring`
    - `Ringing`
    - `Up`
    - `Busy`
    - `Dialing Offhook`
    - `Pre-ring`

- Unknown
- CallerIDNum
- CallerIDName
- ConnectedLineNum
- ConnectedLineName
- AccountCode
- Context
- Exten
- Priority
- Uniqueid
- SSRC - The SSRC identifier for the remote system
- PT - The type of packet for this RTCP report.
  - 200(SR)
  - 201(RR)
- From - The address the report was received from.
- RTT - Calculated Round-Trip Time in seconds
- ReportCount - The number of reports that were received.
  The report count determines the number of ReportX headers in the message. The X for each set of report headers will range from 0 to ReportCount - 1.
- SentNTP - The time the sender generated the report. Only valid when PT is 200(SR).
- SentRTP - The sender's last RTP timestamp. Only valid when PT is 200(SR).
- SentPackets - The number of packets the sender has sent. Only valid when PT is 200(SR).
- SentOctets - The number of bytes the sender has sent. Only valid when PT is 200(SR).
- ReportXSourceSSRC - The SSRC for the source of this report block.
- ReportXFractionLost - The fraction of RTP data packets from ReportXSourceSSRC lost since the previous SR or RR report was sent.
- ReportXCumulativeLost - The total number of RTP data packets from ReportXSourceSSRC lost since the beginning of reception.
- ReportXHighestSequence - The highest sequence number received in an RTP data packet from ReportXSourceSSRC.
- ReportXSequenceNumberCycles - The number of sequence number cycles seen for the RTP data received from ReportXSourceSSRC.
- ReportXIAJitter - An estimate of the statistical variance of the RTP data packet interarrival time, measured in timestamp units.
- ReportXLSR - The last SR timestamp received from ReportXSourceSSRC. If no SR has been received from ReportXSourceSSRC, then 0.
- ReportXDLSR - The delay, expressed in units of 1/65536 seconds, between receiving the last SR packet from ReportXSourceSSRC and sending this report.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_RTCPSent

RTCPSent

Synopsis

Raised when an RTCP packet is sent.

Description

Syntax

```
Action:
Channel: <value>
ChannelState: <value>
ChannelStateDesc: <value>
CallerIDNum: <value>
CallerIDName: <value>
ConnectedLineNum: <value>
ConnectedLineName: <value>
AccountCode: <value>
Context: <value>
Exten: <value>
Priority: <value>
Uniqueid: <value>
SSRC: <value>
PT: <value>
To: <value>
ReportCount: <value>
[SentNTP:] <value>
[SentRTP:] <value>
[SentPackets:] <value>
[SentOctets:] <value>
ReportXSourceSSRC: <value>
ReportXFractionLost: <value>
ReportXCumulativeLost: <value>
ReportXHighestSequence: <value>
ReportXSequenceNumberCycles: <value>
ReportXIAJitter: <value>
ReportXLSR: <value>
ReportXDLSR: <value>
```

Arguments

- `Channel`
- `ChannelState` - A numeric code for the channel's current state, related to ChannelStateDesc
- `ChannelStateDesc`
  - `Down`
  - `Rsrvd`
  - `OffHook`
  - `Dialing`
  - `Ring`
  - `Ringing`
  - `Up`
  - `Busy`
  - `Dialing Offhook`
  - `Pre-ring`
  - `Unknown`
- `CallerIDNum`
- `CallerIDName`
- `ConnectedLineNum`
- `ConnectedLineName`
- `AccountCode`
- `Context`
- `Exten`

- `Priority`
- `Uniqueid`
- `SSRC` - The SSRC identifier for our stream
- `PT` - The type of packet for this RTCP report.
    - `200(SR)`
    - `201(RR)`
- `To` - The address the report is sent to.
- `ReportCount` - The number of reports that were sent.
  The report count determines the number of ReportX headers in the message. The `X` for each set of report headers will range from `0` to `ReportCount - 1`.
- `SentNTP` - The time the sender generated the report. Only valid when PT is `200(SR)`.
- `SentRTP` - The sender's last RTP timestamp. Only valid when PT is `200(SR)`.
- `SentPackets` - The number of packets the sender has sent. Only valid when PT is `200(SR)`.
- `SentOctets` - The number of bytes the sender has sent. Only valid when PT is `200(SR)`.
- `ReportXSourceSSRC` - The SSRC for the source of this report block.
- `ReportXFractionLost` - The fraction of RTP data packets from `ReportXSourceSSRC` lost since the previous SR or RR report was sent.
- `ReportXCumulativeLost` - The total number of RTP data packets from `ReportXSourceSSRC` lost since the beginning of reception.
- `ReportXHighestSequence` - The highest sequence number received in an RTP data packet from `ReportXSourceSSRC`.
- `ReportXSequenceNumberCycles` - The number of sequence number cycles seen for the RTP data received from `ReportXSourceSSRC`.
- `ReportXIAJitter` - An estimate of the statistical variance of the RTP data packet interarrival time, measured in timestamp units.
- `ReportXLSR` - The last SR timestamp received from `ReportXSourceSSRC`. If no SR has been received from `ReportXSourceSSRC`, then `0`.
- `ReportXDLSR` - The delay, expressed in units of 1/65536 seconds, between receiving the last SR packet from `ReportXSourceSSRC` and sending this report.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_SendFAX

SendFAX

Synopsis

Raised when a send fax operation has completed.

Description

Syntax

```
Action:
Channel: <value>
ChannelState: <value>
ChannelStateDesc: <value>
CallerIDNum: <value>
CallerIDName: <value>
ConnectedLineNum: <value>
ConnectedLineName: <value>
AccountCode: <value>
Context: <value>
Exten: <value>
Priority: <value>
Uniqueid: <value>
LocalStationID: <value>
RemoteStationID: <value>
PagesTransferred: <value>
Resolution: <value>
TransferRate: <value>
FileName: <value>
```

Arguments

- `Channel`
- `ChannelState` - A numeric code for the channel's current state, related to ChannelStateDesc
- `ChannelStateDesc`
    - `Down`
    - `Rsrvd`
    - `OffHook`
    - `Dialing`
    - `Ring`
    - `Ringing`
    - `Up`
    - `Busy`
    - `Dialing Offhook`
    - `Pre-ring`
    - `Unknown`
- `CallerIDNum`
- `CallerIDName`
- `ConnectedLineNum`
- `ConnectedLineName`
- `AccountCode`
- `Context`
- `Exten`
- `Priority`
- `Uniqueid`
- `LocalStationID` - The value of the `LOCALSTATIONID` channel variable
- `RemoteStationID` - The value of the `REMOTESTATIONID` channel variable
- `PagesTransferred` - The number of pages that have been transferred
- `Resolution` - The negotiated resolution
- `TransferRate` - The negotiated transfer rate
- `FileName` - The files being affected by the fax operation

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_SessionTimeout

SessionTimeout

Synopsis

Raised when a SIP session times out.

Description

Syntax

```
Action:
Channel: <value>
ChannelState: <value>
ChannelStateDesc: <value>
CallerIDNum: <value>
CallerIDName: <value>
ConnectedLineNum: <value>
ConnectedLineName: <value>
AccountCode: <value>
Context: <value>
Exten: <value>
Priority: <value>
Uniqueid: <value>
Source: <value>
```

Arguments

- `Channel`
- `ChannelState` - A numeric code for the channel's current state, related to ChannelStateDesc
- `ChannelStateDesc`
    - `Down`
    - `Rsrvd`
    - `OffHook`
    - `Dialing`
    - `Ring`
    - `Ringing`
    - `Up`
    - `Busy`
    - `Dialing Offhook`
    - `Pre-ring`
    - `Unknown`
- `CallerIDNum`
- `CallerIDName`
- `ConnectedLineNum`
- `ConnectedLineName`
- `AccountCode`
- `Context`
- `Exten`
- `Priority`
- `Uniqueid`
- `Source` - The source of the session timeout.
    - `RTPTimeout`

- SIPSessionTimer

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_Shutdown

Shutdown

Synopsis

Raised when Asterisk is shutdown or restarted.

Description

Syntax

```
Action:
Shutdown: <value>
Restart: <value>
```

Arguments

- `Shutdown` - Whether the shutdown is proceeding cleanly (all channels were hungup successfully) or uncleanly (channels will be terminated)
  - `Uncleanly`
  - `Cleanly`
- `Restart` - Whether or not a restart will occur.
  - `True`
  - `False`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_SIPQualifyPeerDone

SIPQualifyPeerDone

Synopsis

Raised when SIPQualifyPeer has finished qualifying the specified peer.

Description

Syntax

```
Action:
Peer: <value>
ActionID: <value>
```

Arguments

* `Peer` - The name of the peer.
* `ActionID` - This is only included if an ActionID Header was sent with the action request, in which case it will be that ActionID.

See Also

* Asterisk 12 ManagerAction_SIPqualifypeer

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_SoftHangupRequest

SoftHangupRequest

Synopsis

Raised when a soft hangup is requested with a specific cause code.

Description

Syntax

```
Action:
Channel: <value>
ChannelState: <value>
ChannelStateDesc: <value>
CallerIDNum: <value>
CallerIDName: <value>
ConnectedLineNum: <value>
ConnectedLineName: <value>
AccountCode: <value>
Context: <value>
Exten: <value>
Priority: <value>
Uniqueid: <value>
Cause: <value>
```

Arguments

* `Channel`
* `ChannelState` - A numeric code for the channel's current state, related to ChannelStateDesc
* `ChannelStateDesc`
    * `Down`
    * `Rsrvd`
    * `OffHook`
    * `Dialing`
    * `Ring`
    * `Ringing`
    * `Up`
    * `Busy`
    * `Dialing Offhook`
    * `Pre-ring`
    * `Unknown`
* `CallerIDNum`
* `CallerIDName`
* `ConnectedLineNum`

* `ConnectedLineName`
* `AccountCode`
* `Context`
* `Exten`
* `Priority`
* `Uniqueid`
* `Cause` - A numeric cause code for why the channel was hung up.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_SpanAlarm

SpanAlarm

Synopsis

Raised when an alarm is set on a DAHDI span.

Description

Syntax

```
Action:
Span: <value>
Alarm: <value>
```

Arguments

* `Span` - The span on which the alarm occurred.
* `Alarm` - A textual description of the alarm that occurred.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_SpanAlarmClear

SpanAlarmClear

Synopsis

Raised when an alarm is cleared on a DAHDI span.

Description

Syntax

```
Action:
Span: <value>
```

Arguments

- `Span` - The span on which the alarm was cleared.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_Status

Status

Synopsis

Raised in response to a Status command.

Description

Syntax

```
Action:
[ActionID:] <value>
Channel: <value>
ChannelState: <value>
ChannelStateDesc: <value>
CallerIDNum: <value>
CallerIDName: <value>
ConnectedLineNum: <value>
ConnectedLineName: <value>
AccountCode: <value>
Context: <value>
Exten: <value>
Priority: <value>
Uniqueid: <value>
Type: <value>
DNID: <value>
TimeToHangup: <value>
BridgeID: <value>
Linkedid: <value>
Application: <value>
Data: <value>
Nativeformats: <value>
Readformat: <value>
Readtrans: <value>
Writeformat: <value>
Writetrans: <value>
Callgroup: <value>
Pickupgroup: <value>
Seconds: <value>
```

Arguments

- `ActionID`
- `Channel`
- `ChannelState` - A numeric code for the channel's current state, related to ChannelStateDesc

- `ChannelStateDesc`
    - `Down`
    - `Rsrvd`
    - `OffHook`
    - `Dialing`
    - `Ring`
    - `Ringing`
    - `Up`
    - `Busy`
    - `Dialing Offhook`
    - `Pre-ring`
    - `Unknown`
- `CallerIDNum`
- `CallerIDName`
- `ConnectedLineNum`
- `ConnectedLineName`
- `AccountCode`
- `Context`
- `Exten`
- `Priority`
- `Uniqueid`
- `Type` - Type of channel
- `DNID` - Dialed number identifier
- `TimeToHangup` - Absolute lifetime of the channel
- `BridgeID` - Identifier of the bridge the channel is in, may be empty if not in one
- `Linkedid`
- `Application` - Application currently executing on the channel
- `Data` - Data given to the currently executing channel
- `Nativeformats` - Media formats the connected party is willing to send or receive
- `Readformat` - Media formats that frames from the channel are received in
- `Readtrans` - Translation path for media received in native formats
- `Writeformat` - Media formats that frames to the channel are accepted in
- `Writetrans` - Translation path for media sent to the connected party
- `Callgroup` - Configured call group on the channel
- `Pickupgroup` - Configured pickup group on the channel
- `Seconds` - Number of seconds the channel has been active

See Also

- Asterisk 12 ManagerAction_Status

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_Unhold

Unhold

Synopsis

Raised when a channel goes off hold.

Description

Syntax

```
Action:
Channel: <value>
ChannelState: <value>
ChannelStateDesc: <value>
CallerIDNum: <value>
CallerIDName: <value>
ConnectedLineNum: <value>
ConnectedLineName: <value>
AccountCode: <value>
Context: <value>
Exten: <value>
Priority: <value>
Uniqueid: <value>
```

Arguments

- `Channel`
- `ChannelState` - A numeric code for the channel's current state, related to ChannelStateDesc
- `ChannelStateDesc`
  - `Down`
  - `Rsrvd`
  - `OffHook`
  - `Dialing`
  - `Ring`
  - `Ringing`
  - `Up`
  - `Busy`
  - `Dialing Offhook`
  - `Pre-ring`
  - `Unknown`
- `CallerIDNum`
- `CallerIDName`
- `ConnectedLineNum`
- `ConnectedLineName`
- `AccountCode`
- `Context`
- `Exten`
- `Priority`
- `Uniqueid`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_UnParkedCall

UnParkedCall

Synopsis

Raised when a channel leaves a parking lot because it was retrieved from the parking lot and reconnected.

Description

Syntax

```
Action:
ParkeeChannel: <value>
ParkeeChannelState: <value>
ParkeeChannelStateDesc: <value>
ParkeeCallerIDNum: <value>
ParkeeCallerIDName: <value>
ParkeeConnectedLineNum: <value>
ParkeeConnectedLineName: <value>
ParkeeAccountCode: <value>
ParkeeContext: <value>
ParkeeExten: <value>
ParkeePriority: <value>
ParkeeUniqueid: <value>
ParkerChannel: <value>
ParkerChannelState: <value>
ParkerChannelStateDesc: <value>
ParkerCallerIDNum: <value>
ParkerCallerIDName: <value>
ParkerConnectedLineNum: <value>
ParkerConnectedLineName: <value>
ParkerAccountCode: <value>
ParkerContext: <value>
ParkerExten: <value>
ParkerPriority: <value>
ParkerUniqueid: <value>
ParkerDialString: <value>
Parkinglot: <value>
ParkingSpace: <value>
ParkingTimeout: <value>
ParkingDuration: <value>
RetrieverChannel: <value>
RetrieverChannelState: <value>
RetrieverChannelStateDesc: <value>
RetrieverCallerIDNum: <value>
RetrieverCallerIDName: <value>
RetrieverConnectedLineNum: <value>
RetrieverConnectedLineName: <value>
RetrieverAccountCode: <value>
RetrieverContext: <value>
RetrieverExten: <value>
RetrieverPriority: <value>
RetrieverUniqueid: <value>
```

Arguments

- `ParkeeChannel`
- `ParkeeChannelState` - A numeric code for the channel's current state, related to ParkeeChannelStateDesc
- `ParkeeChannelStateDesc`
    - `Down`
    - `Rsrvd`

    - `OffHook`
    - `Dialing`
    - `Ring`
    - `Ringing`
    - `Up`
    - `Busy`
    - `Dialing Offhook`
    - `Pre-ring`
    - `Unknown`
- `ParkeeCallerIDNum`
- `ParkeeCallerIDName`
- `ParkeeConnectedLineNum`
- `ParkeeConnectedLineName`
- `ParkeeAccountCode`
- `ParkeeContext`
- `ParkeeExten`
- `ParkeePriority`
- `ParkeeUniqueid`
- `ParkerChannel`
- `ParkerChannelState` - A numeric code for the channel's current state, related to ParkerChannelStateDesc
- `ParkerChannelStateDesc`
    - `Down`
    - `Rsrvd`
    - `OffHook`
    - `Dialing`
    - `Ring`
    - `Ringing`
    - `Up`
    - `Busy`
    - `Dialing Offhook`
    - `Pre-ring`
    - `Unknown`
- `ParkerCallerIDNum`
- `ParkerCallerIDName`
- `ParkerConnectedLineNum`
- `ParkerConnectedLineName`
- `ParkerAccountCode`
- `ParkerContext`
- `ParkerExten`
- `ParkerPriority`
- `ParkerUniqueid`
- `ParkerDialString` - Dial String that can be used to call back the parker on ParkingTimeout.
- `Parkinglot` - Name of the parking lot that the parkee is parked in
- `ParkingSpace` - Parking Space that the parkee is parked in
- `ParkingTimeout` - Time remaining until the parkee is forcefully removed from parking in seconds
- `ParkingDuration` - Time the parkee has been in the parking bridge (in seconds)
- `RetrieverChannel`
- `RetrieverChannelState` - A numeric code for the channel's current state, related to RetrieverChannelStateDesc
- `RetrieverChannelStateDesc`
    - `Down`
    - `Rsrvd`
    - `OffHook`
    - `Dialing`
    - `Ring`
    - `Ringing`
    - `Up`
    - `Busy`
    - `Dialing Offhook`
    - `Pre-ring`
    - `Unknown`

* `RetrieverCallerIDNum`
* `RetrieverCallerIDName`
* `RetrieverConnectedLineNum`
* `RetrieverConnectedLineName`
* `RetrieverAccountCode`
* `RetrieverContext`
* `RetrieverExten`
* `RetrieverPriority`
* `RetrieverUniqueid`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_UserEvent

UserEvent

Synopsis

A user defined event raised from the dialplan.

Description

Syntax

```
Action:
Channel: <value>
ChannelState: <value>
ChannelStateDesc: <value>
CallerIDNum: <value>
CallerIDName: <value>
ConnectedLineNum: <value>
ConnectedLineName: <value>
AccountCode: <value>
Context: <value>
Exten: <value>
Priority: <value>
Uniqueid: <value>
UserEvent: <value>
```

Arguments

* `Channel`
* `ChannelState` - A numeric code for the channel's current state, related to ChannelStateDesc
* `ChannelStateDesc`
    * `Down`
    * `Rsrvd`
    * `OffHook`
    * `Dialing`
    * `Ring`
    * `Ringing`
    * `Up`
    * `Busy`
    * `Dialing Offhook`

    - Pre-ring
    - Unknown
- CallerIDNum
- CallerIDName
- ConnectedLineNum
- ConnectedLineName
- AccountCode
- Context
- Exten
- Priority
- Uniqueid
- UserEvent - The event name, as specified in the dialplan.

See Also

- Asterisk 12 Application_UserEvent

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ManagerEvent_VarSet

VarSet

Synopsis

Raised when a variable local to the gosub stack frame is set due to a subroutine call.

Description

Syntax

```
Action:
Channel: <value>
ChannelState: <value>
ChannelStateDesc: <value>
CallerIDNum: <value>
CallerIDName: <value>
ConnectedLineNum: <value>
ConnectedLineName: <value>
AccountCode: <value>
Context: <value>
Exten: <value>
Priority: <value>
Uniqueid: <value>
Variable: <value>
Value: <value>
```

Arguments

- Channel
- ChannelState - A numeric code for the channel's current state, related to ChannelStateDesc
- ChannelStateDesc
    - Down
    - Rsrvd
    - OffHook

    - `Dialing`
    - `Ring`
    - `Ringing`
    - `Up`
    - `Busy`
    - `Dialing Offhook`
    - `Pre-ring`
    - `Unknown`
- `CallerIDNum`
- `CallerIDName`
- `ConnectedLineNum`
- `ConnectedLineName`
- `AccountCode`
- `Context`
- `Exten`
- `Priority`
- `Uniqueid`
- `Variable` - The LOCAL variable being set.

  > **Note**
  > The variable name will always be enclosed with `LOCAL()`

- `Value` - The new value of the variable.

See Also

- Asterisk 12 Application_GoSub
- Asterisk 12 AGICommand_gosub
- Asterisk 12 Function_LOCAL
- Asterisk 12 Function_LOCAL_PEEK

Synopsis

Raised when a variable is shared between channels.

Description

Syntax

```
Action:
Channel: <value>
ChannelState: <value>
ChannelStateDesc: <value>
CallerIDNum: <value>
CallerIDName: <value>
ConnectedLineNum: <value>
ConnectedLineName: <value>
AccountCode: <value>
Context: <value>
Exten: <value>
Priority: <value>
Uniqueid: <value>
Variable: <value>
Value: <value>
```

Arguments

- `Channel`
- `ChannelState` - A numeric code for the channel's current state, related to ChannelStateDesc
- `ChannelStateDesc`
    - `Down`
    - `Rsrvd`
    - `OffHook`
    - `Dialing`
    - `Ring`
    - `Ringing`
    - `Up`
    - `Busy`
    - `Dialing Offhook`
    - `Pre-ring`
    - `Unknown`
- `CallerIDNum`
- `CallerIDName`
- `ConnectedLineNum`
- `ConnectedLineName`
- `AccountCode`
- `Context`
- `Exten`
- `Priority`
- `Uniqueid`
- `Variable` - The SHARED variable being set.

> **Note**
> The variable name will always be enclosed with `SHARED()`

- `Value` - The new value of the variable.

See Also

- Asterisk 12 Function_SHARED

Synopsis

Raised when a variable is set to a particular value.

Description

Syntax

```
Action:
Channel: <value>
ChannelState: <value>
ChannelStateDesc: <value>
CallerIDNum: <value>
CallerIDName: <value>
ConnectedLineNum: <value>
ConnectedLineName: <value>
AccountCode: <value>
Context: <value>
Exten: <value>
Priority: <value>
Uniqueid: <value>
Variable: <value>
Value: <value>
```

Arguments

- `Channel`
- `ChannelState` - A numeric code for the channel's current state, related to ChannelStateDesc
- `ChannelStateDesc`
  - `Down`
  - `Rsrvd`
  - `OffHook`
  - `Dialing`
  - `Ring`
  - `Ringing`
  - `Up`
  - `Busy`
  - `Dialing Offhook`
  - `Pre-ring`
  - `Unknown`
- `CallerIDNum`
- `CallerIDName`
- `ConnectedLineNum`
- `ConnectedLineName`
- `AccountCode`
- `Context`
- `Exten`
- `Priority`
- `Uniqueid`
- `Variable` - The variable being set.
- `Value` - The new value of the variable.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 ARI

Asterisk 12 REST Data Models

- AsteriskInfo
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- ConfigInfo

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AsteriskInfo

Asterisk system information

```
{
  "properties": {
    "status": {
      "required": false,
      "type": "StatusInfo",
      "description": "Info about Asterisk status"
    },
    "config": {
      "required": false,
      "type": "ConfigInfo",
      "description": "Info about Asterisk configuration"
    },
    "build": {
      "required": false,
      "type": "BuildInfo",
      "description": "Info about how Asterisk was built"
    },
    "system": {
      "required": false,
      "type": "SystemInfo",
      "description": "Info about the system running Asterisk"
    }
  },
  "id": "AsteriskInfo",
  "description": "Asterisk system information"
}
```

- build: BuildInfo *(optional)* - Info about how Asterisk was built
- config: ConfigInfo *(optional)* - Info about Asterisk configuration
- status: StatusInfo *(optional)* - Info about Asterisk status
- system: SystemInfo *(optional)* - Info about the system running Asterisk

BuildInfo

Info about how Asterisk was built

```
{
  "properties": {
    "kernel": {
      "required": true,
      "type": "string",
      "description": "Kernel version Asterisk was built on."
    },
    "machine": {
      "required": true,
      "type": "string",
      "description": "Machine architecture (x86_64, i686, ppc, etc.)"
    },
    "user": {
      "required": true,
      "type": "string",
      "description": "Username that build Asterisk"
    },
    "date": {
      "required": true,
      "type": "string",
      "description": "Date and time when Asterisk was built."
    },
    "os": {
      "required": true,
      "type": "string",
      "description": "OS Asterisk was built on."
    },
    "options": {
      "required": true,
      "type": "string",
      "description": "Compile time options, or empty string if default."
    }
  },
  "id": "BuildInfo",
  "description": "Info about how Asterisk was built"
}
```

- date: string - Date and time when Asterisk was built.
- kernel: string - Kernel version Asterisk was built on.
- machine: string - Machine architecture (x86_64, i686, ppc, etc.)
- options: string - Compile time options, or empty string if default.
- os: string - OS Asterisk was built on.
- user: string - Username that build Asterisk

ConfigInfo

Info about Asterisk configuration

```
{
  "properties": {
    "name": {
      "required": true,
      "type": "string",
      "description": "Asterisk system name."
    },
    "default_language": {
      "required": true,
      "type": "string",
      "description": "Default language for media playback."
    },
    "max_load": {
      "required": false,
      "type": "double",
      "description": "Maximum load avg on system."
    },
    "setid": {
      "required": true,
      "type": "SetId",
      "description": "Effective user/group id for running Asterisk."
    },
    "max_open_files": {
      "required": false,
      "type": "int",
      "description": "Maximum number of open file handles (files, sockets)."
    },
    "max_channels": {
      "required": false,
      "type": "int",
      "description": "Maximum number of simultaneous channels."
    }
  },
  "id": "ConfigInfo",
  "description": "Info about Asterisk configuration"
}
```

- default_language: string - Default language for media playback.
- max_channels: int *(optional)* - Maximum number of simultaneous channels.
- max_load: double *(optional)* - Maximum load avg on system.
- max_open_files: int *(optional)* - Maximum number of open file handles (files, sockets).
- name: string - Asterisk system name.
- setid: SetId - Effective user/group id for running Asterisk.

SetId

Effective user/group id

```
{
  "properties": {
    "group": {
      "required": true,
      "type": "string",
      "description": "Effective group id."
    },
    "user": {
      "required": true,
      "type": "string",
      "description": "Effective user id."
    }
  },
  "id": "SetId",
  "description": "Effective user/group id"
}
```

- group: string - Effective group id.
- user: string - Effective user id.

StatusInfo

Info about Asterisk status

```
{
  "properties": {
    "last_reload_time": {
      "required": true,
      "type": "Date",
      "description": "Time when Asterisk was last reloaded."
    },
    "startup_time": {
      "required": true,
      "type": "Date",
      "description": "Time when Asterisk was started."
    }
  },
  "id": "StatusInfo",
  "description": "Info about Asterisk status"
}
```

- last_reload_time: Date - Time when Asterisk was last reloaded.
- startup_time: Date - Time when Asterisk was started.

SystemInfo

Info about Asterisk

```
{
  "properties": {
    "entity_id": {
      "required": true,
      "type": "string",
      "description": ""
    },
    "version": {
      "required": true,
      "type": "string",
      "description": "Asterisk version."
    }
  },
  "id": "SystemInfo",
  "description": "Info about Asterisk"
}
```

- entity_id: string
- version: string - Asterisk version.

Variable

The value of a channel variable

```
{
  "properties": {
    "value": {
      "required": true,
      "type": "string",
      "description": "The value of the variable requested"
    }
  },
  "id": "Variable",
  "description": "The value of a channel variable"
}
```

- value: string - The value of the variable requested

Endpoint

An external device that may offer/accept calls to/from Asterisk.

Unlike most resources, which have a single unique identifier, an endpoint is uniquely identified by the technology/resource pair.

```
{
  "properties": {
    "resource": {
      "required": true,
      "type": "string",
      "description": "Identifier of the endpoint, specific to the given technology."
    },
    "state": {
      "allowableValues": {
        "valueType": "LIST",
        "values": [
          "unknown",
          "offline",
          "online"
        ]
      },
      "required": false,
      "type": "string",
      "description": "Endpoint's state"
    },
    "technology": {
      "required": true,
      "type": "string",
      "description": "Technology of the endpoint"
    },
    "channel_ids": {
      "required": true,
      "type": "List[string]",
      "description": "Id's of channels associated with this endpoint"
    }
  },
  "id": "Endpoint",
  "description": "An external device that may offer/accept calls to/from
Asterisk.\n\nUnlike most resources, which have a single unique identifier, an endpoint
is uniquely identified by the technology/resource pair."
}
```

- channel_ids: List[string] - Id's of channels associated with this endpoint
- resource: string - Identifier of the endpoint, specific to the given technology.
- state: string *(optional)* - Endpoint's state
- technology: string - Technology of the endpoint

CallerID

Caller identification

```
{
  "properties": {
    "name": {
      "required": true,
      "type": "string"
    },
    "number": {
      "required": true,
      "type": "string"
    }
  },
  "id": "CallerID",
  "description": "Caller identification"
}
```

- name: string
- number: string

Channel

A specific communication connection between Asterisk and an Endpoint.

```
{
  "properties": {
    "accountcode": {
      "required": true,
      "type": "string"
    },
    "name": {
      "required": true,
      "type": "string",
      "description": "Name of the channel (i.e. SIP/foo-0000a7e3)"
    },
    "caller": {
      "required": true,
      "type": "CallerID"
    },
    "creationtime": {
      "required": true,
      "type": "Date",
      "description": "Timestamp when channel was created"
    },
    "state": {
      "allowableValues": {
        "valueType": "LIST",
        "values": [
          "Down",
          "Rsrved",
          "OffHook",
          "Dialing",
          "Ring",
          "Ringing",
          "Up",
          "Busy",
```

```
          "Dialing Offhook",
          "Pre-ring",
          "Unknown"
        ]
      },
      "required": true,
      "type": "string"
    },
    "connected": {
      "required": true,
      "type": "CallerID"
    },
    "dialplan": {
      "required": true,
      "type": "DialplanCEP",
      "description": "Current location in the dialplan"
    },
    "id": {
      "required": true,
      "type": "string",
      "description": "Unique identifier of the channel.\n\nThis is the same as the
Uniqueid field in AMI."
    }
  },
  "id": "Channel",
  "description": "A specific communication connection between Asterisk and an
```

```
Endpoint."
}
```

- accountcode: string
- caller: CallerID
- connected: CallerID
- creationtime: Date - Timestamp when channel was created
- dialplan: DialplanCEP - Current location in the dialplan
- id: string - Unique identifier of the channel.

This is the same as the Uniqueid field in AMI.

- name: string - Name of the channel (i.e. SIP/foo-0000a7e3)
- state: string

Dialed

Dialed channel information.

```
{
  "properties": {},
  "id": "Dialed",
  "description": "Dialed channel information."
}
```

DialplanCEP

Dialplan location (context/extension/priority)

```
{
  "properties": {
    "priority": {
      "required": true,
      "type": "long",
      "description": "Priority in the dialplan"
    },
    "exten": {
      "required": true,
      "type": "string",
      "description": "Extension in the dialplan"
    },
    "context": {
      "required": true,
      "type": "string",
      "description": "Context in the dialplan"
    }
  },
  "id": "DialplanCEP",
  "description": "Dialplan location (context/extension/priority)"
}
```

- context: string - Context in the dialplan

- exten: string - Extension in the dialplan
- priority: long - Priority in the dialplan

Bridge

The merging of media from one or more channels.

Everyone on the bridge receives the same audio.


```
{
  "properties": {
    "channels": {
      "required": true,
      "type": "List[string]",
      "description": "Ids of channels participating in this bridge"
    },
    "bridge_type": {
      "allowableValues": {
        "valueType": "LIST",
        "values": [
          "mixing",
          "holding"
        ]
      },
      "required": true,
      "type": "string",
      "description": "Type of bridge technology"
    },
    "technology": {
      "required": true,
      "type": "string",
      "description": "Name of the current bridging technology"
    },
    "id": {
      "required": true,
      "type": "string",
      "description": "Unique identifier for this bridge"
    },
    "bridge_class": {
      "required": true,
      "type": "string",
      "description": "Bridging class"
    }
  },
  "id": "Bridge",
  "description": "The merging of media from one or more channels.\n\nEveryone on the
bridge receives the same audio."
}
```


- bridge_class: string - Bridging class
- bridge_type: string - Type of bridge technology
- channels: List[string] - Ids of channels participating in this bridge
- id: string - Unique identifier for this bridge
- technology: string - Name of the current bridging technology

LiveRecording

A recording that is in progress

```
{
  "properties": {
    "state": {
      "required": true,
      "type": "string"
    },
    "name": {
      "required": true,
      "type": "string"
    },
    "format": {
      "required": true,
      "type": "string"
    }
  },
  "id": "LiveRecording",
  "description": "A recording that is in progress"
}
```

- format: string
- name: string
- state: string

StoredRecording

A past recording that may be played back.

```
{
  "properties": {
    "time": {
      "required": false,
      "type": "Date",
      "description": "Time recording was started"
    },
    "id": {
      "required": true,
      "type": "string"
    },
    "duration_seconds": {
      "required": false,
      "type": "int"
    },
    "formats": {
      "required": true,
      "type": "List[string]"
    }
  },
  "id": "StoredRecording",
  "description": "A past recording that may be played back."
}
```

- duration_seconds: int *(optional)*
- formats: List[string]
- id: string
- time: Date *(optional)* - Time recording was started

FormatLangPair

Identifies the format and language of a sound file

```
{
  "properties": {
    "language": {
      "required": true,
      "type": "string"
    },
    "format": {
      "required": true,
      "type": "string"
    }
  },
  "id": "FormatLangPair",
  "description": "Identifies the format and language of a sound file"
}
```

- format: string
- language: string

Sound

A media file that may be played back.

```
{
  "properties": {
    "text": {
      "required": false,
      "type": "string",
      "description": "Text description of the sound, usually the words spoken."
    },
    "id": {
      "required": true,
      "type": "string",
      "description": "Sound's identifier."
    },
    "formats": {
      "required": true,
      "type": "List[FormatLangPair]",
      "description": "The formats and languages in which this sound is available."
    }
  },
  "id": "Sound",
  "description": "A media file that may be played back."
}
```

- formats: List[FormatLangPair] - The formats and languages in which this sound is available.

- id: string - Sound's identifier.
- text: string *(optional)* - Text description of the sound, usually the words spoken.

Playback

Object representing the playback of media to a channel


```
{
  "properties": {
    "language": {
      "type": "string",
      "description": "For media types that support multiple languages, the language requested for playback."
    },
    "media_uri": {
      "required": true,
      "type": "string",
      "description": "URI for the media to play back."
    },
    "id": {
      "required": true,
      "type": "string",
      "description": "ID for this playback operation"
    },
    "target_uri": {
      "required": true,
      "type": "string",
      "description": "URI for the channel or bridge to play the media on"
    },
    "state": {
      "allowableValues": {
        "valueType": "LIST",
        "values": [
          "queued",
          "playing",
          "complete"
        ]
      },
      "required": true,
      "type": "string",
      "description": "Current state of the playback operation."
    }
  },
  "id": "Playback",
  "description": "Object representing the playback of media to a channel"
}
```


- id: string - ID for this playback operation
- language: string *(optional)* - For media types that support multiple languages, the language requested for playback.
- media_uri: string - URI for the media to play back.
- state: string - Current state of the playback operation.
- target_uri: string - URI for the channel or bridge to play the media on

ApplicationReplaced

Base type: Event

Notification that another WebSocket has taken over for an application.

An application may only be subscribed to by a single WebSocket at a time. If multiple WebSockets attempt to subscribe to the same application, the newer WebSocket wins, and the older one receives this event.


```
{
  "properties": {},
  "extends": "Event",
  "id": "ApplicationReplaced",
  "description": "Notification that another WebSocket has taken over for an
application.\n\nAn application may only be subscribed to by a single WebSocket at a
time. If multiple WebSockets attempt to subscribe to the same application, the newer
WebSocket wins, and the older one receives this event."
}
```


- type: string - Indicates the type of this message.
- application: string - Name of the application receiving the event.
- timestamp: Date *(optional)* - Time at which this event was created.

BridgeCreated

Base type: Event

Notification that a bridge has been created.


```
{
  "properties": {
    "bridge": {
      "required": true,
      "type": "Bridge"
    }
  },
  "extends": "Event",
  "id": "BridgeCreated",
  "description": "Notification that a bridge has been created."
}
```


- type: string - Indicates the type of this message.
- application: string - Name of the application receiving the event.
- timestamp: Date *(optional)* - Time at which this event was created.
- bridge: Bridge

BridgeDestroyed

Base type: Event

Notification that a bridge has been destroyed.

```
{
  "properties": {
    "bridge": {
      "required": true,
      "type": "Bridge"
    }
  },
  "extends": "Event",
  "id": "BridgeDestroyed",
  "description": "Notification that a bridge has been destroyed."
}
```

- type: string - Indicates the type of this message.
- application: string - Name of the application receiving the event.
- timestamp: Date *(optional)* - Time at which this event was created.
- bridge: Bridge

BridgeMerged

Base type: Event

Notification that one bridge has merged into another.

```
{
  "properties": {
    "bridge": {
      "required": true,
      "type": "Bridge"
    },
    "bridge_from": {
      "required": true,
      "type": "Bridge"
    }
  },
  "extends": "Event",
  "id": "BridgeMerged",
  "description": "Notification that one bridge has merged into another."
}
```

- type: string - Indicates the type of this message.
- application: string - Name of the application receiving the event.
- timestamp: Date *(optional)* - Time at which this event was created.
- bridge: Bridge
- bridge_from: Bridge

ChannelCallerId

Base type: Event

Channel changed Caller ID.

```
{
  "properties": {
    "caller_presentation_txt": {
      "required": true,
      "type": "string",
      "description": "The text representation of the Caller Presentation value."
    },
    "caller_presentation": {
      "required": true,
      "type": "int",
      "description": "The integer representation of the Caller Presentation value."
    },
    "channel": {
      "required": true,
      "type": "Channel",
      "description": "The channel that changed Caller ID."
    }
  },
  "extends": "Event",
  "id": "ChannelCallerId",
  "description": "Channel changed Caller ID."
}
```

- type: string - Indicates the type of this message.
- application: string - Name of the application receiving the event.
- timestamp: Date *(optional)* - Time at which this event was created.
- caller_presentation: int - The integer representation of the Caller Presentation value.
- caller_presentation_txt: string - The text representation of the Caller Presentation value.
- channel: Channel - The channel that changed Caller ID.

ChannelCreated

Base type: Event

Notification that a channel has been created.

```
{
  "properties": {
    "channel": {
      "required": true,
      "type": "Channel"
    }
  },
  "extends": "Event",
  "id": "ChannelCreated",
  "description": "Notification that a channel has been created."
}
```

- type: string - Indicates the type of this message.
- application: string - Name of the application receiving the event.
- timestamp: Date *(optional)* - Time at which this event was created.
- channel: Channel

ChannelDestroyed

Base type: Event

Notification that a channel has been destroyed.


```
{
  "properties": {
    "cause": {
      "required": true,
      "type": "int",
      "description": "Integer representation of the cause of the hangup"
    },
    "cause_txt": {
      "required": true,
      "type": "string",
      "description": "Text representation of the cause of the hangup"
    },
    "channel": {
      "required": true,
      "type": "Channel"
    }
  },
  "extends": "Event",
  "id": "ChannelDestroyed",
  "description": "Notification that a channel has been destroyed."
}
```


- type: string - Indicates the type of this message.
- application: string - Name of the application receiving the event.
- timestamp: Date *(optional)* - Time at which this event was created.
- cause: int - Integer representation of the cause of the hangup
- cause_txt: string - Text representation of the cause of the hangup
- channel: Channel

ChannelDialplan

Base type: Event

Channel changed location in the dialplan.

```
{
  "properties": {
    "dialplan_app_data": {
      "required": true,
      "type": "string",
      "description": "The data to be passed to the application."
    },
    "channel": {
      "required": true,
      "type": "Channel",
      "description": "The channel that changed dialplan location."
    },
    "dialplan_app": {
      "required": true,
      "type": "string",
      "description": "The application about to be executed."
    }
  },
  "extends": "Event",
  "id": "ChannelDialplan",
  "description": "Channel changed location in the dialplan."
}
```

- type: string - Indicates the type of this message.
- application: string - Name of the application receiving the event.
- timestamp: Date *(optional)* - Time at which this event was created.
- channel: Channel - The channel that changed dialplan location.
- dialplan_app: string - The application about to be executed.
- dialplan_app_data: string - The data to be passed to the application.

ChannelDtmfReceived

Base type: Event

DTMF received on a channel.

This event is sent when the DTMF ends. There is no notification about the start of DTMF

```
{
  "properties": {
    "duration_ms": {
      "required": true,
      "type": "int",
      "description": "Number of milliseconds DTMF was received"
    },
    "digit": {
      "required": true,
      "type": "string",
      "description": "DTMF digit received (0-9, A-E, # or *)"
    },
    "channel": {
      "required": true,
      "type": "Channel",
      "description": "The channel on which DTMF was received"
    }
  },
  "extends": "Event",
  "id": "ChannelDtmfReceived",
  "description": "DTMF received on a channel.\n\nThis event is sent when the DTMF
ends. There is no notification about the start of DTMF"
}
```

- type: string - Indicates the type of this message.
- application: string - Name of the application receiving the event.
- timestamp: Date *(optional)* - Time at which this event was created.
- channel: Channel - The channel on which DTMF was received
- digit: string - DTMF digit received (0-9, A-E, # or *)
- duration_ms: int - Number of milliseconds DTMF was received

ChannelEnteredBridge

Base type: Event

Notification that a channel has entered a bridge.

```
{
  "properties": {
    "bridge": {
      "required": true,
      "type": "Bridge"
    },
    "channel": {
      "type": "Channel"
    }
  },
  "extends": "Event",
  "id": "ChannelEnteredBridge",
  "description": "Notification that a channel has entered a bridge."
}
```

- type: string - Indicates the type of this message.

- application: string - Name of the application receiving the event.
- timestamp: Date *(optional)* - Time at which this event was created.
- bridge: Bridge
- channel: Channel *(optional)*

ChannelHangupRequest

Base type: Event

A hangup was requested on the channel.


```
{
  "properties": {
    "soft": {
      "type": "boolean",
      "description": "Whether the hangup request was a soft hangup request."
    },
    "cause": {
      "type": "int",
      "description": "Integer representation of the cause of the hangup."
    },
    "channel": {
      "required": true,
      "type": "Channel",
      "description": "The channel on which the hangup was requested."
    }
  },
  "extends": "Event",
  "id": "ChannelHangupRequest",
  "description": "A hangup was requested on the channel."
}
```


- type: string - Indicates the type of this message.
- application: string - Name of the application receiving the event.
- timestamp: Date *(optional)* - Time at which this event was created.
- cause: int *(optional)* - Integer representation of the cause of the hangup.
- channel: Channel - The channel on which the hangup was requested.
- soft: boolean *(optional)* - Whether the hangup request was a soft hangup request.

ChannelLeftBridge

Base type: Event

Notification that a channel has left a bridge.

```
{
  "properties": {
    "bridge": {
      "required": true,
      "type": "Bridge"
    },
    "channel": {
      "required": true,
      "type": "Channel"
    }
  },
  "extends": "Event",
  "id": "ChannelLeftBridge",
  "description": "Notification that a channel has left a bridge."
}
```

- type: string - Indicates the type of this message.
- application: string - Name of the application receiving the event.
- timestamp: Date *(optional)* - Time at which this event was created.
- bridge: Bridge
- channel: Channel

ChannelStateChange

Base type: Event

Notification of a channel's state change.

```
{
  "properties": {
    "channel": {
      "required": true,
      "type": "Channel"
    }
  },
  "extends": "Event",
  "id": "ChannelStateChange",
  "description": "Notification of a channel's state change."
}
```

- type: string - Indicates the type of this message.
- application: string - Name of the application receiving the event.
- timestamp: Date *(optional)* - Time at which this event was created.
- channel: Channel

ChannelUserevent

Base type: Event

User-generated event with additional user-defined fields in the object.

```
{
  "properties": {
    "eventname": {
      "required": true,
      "type": "string",
      "description": "The name of the user event."
    },
    "channel": {
      "required": true,
      "type": "Channel",
      "description": "The channel that signaled the user event."
    }
  },
  "extends": "Event",
  "id": "ChannelUserevent",
  "description": "User-generated event with additional user-defined fields in the
object."
}
```

- type: string - Indicates the type of this message.
- application: string - Name of the application receiving the event.
- timestamp: Date *(optional)* - Time at which this event was created.
- channel: Channel - The channel that signaled the user event.
- eventname: string - The name of the user event.

ChannelVarset

Base type: Event

Channel variable changed.

```
{
  "properties": {
    "variable": {
      "required": true,
      "type": "string",
      "description": "The variable that changed."
    },
    "channel": {
      "required": false,
      "type": "Channel",
      "description": "The channel on which the variable was set.\n\nIf missing, the
variable is a global variable."
    },
    "value": {
      "required": true,
      "type": "string",
      "description": "The new value of the variable."
    }
  },
  "extends": "Event",
  "id": "ChannelVarset",
  "description": "Channel variable changed."
}
```

- type: string - Indicates the type of this message.
- application: string - Name of the application receiving the event.
- timestamp: Date *(optional)* - Time at which this event was created.
- channel: Channel *(optional)* - The channel on which the variable was set.

If missing, the variable is a global variable.

- value: string - The new value of the variable.
- variable: string - The variable that changed.

Event

Base type: Message
Subtypes: ApplicationReplaced BridgeCreated BridgeDestroyed BridgeMerged ChannelCallerId ChannelCreated ChannelDestroyed ChannelDialplan ChannelDtmfReceived ChannelEnteredBridge ChannelHangupRequest ChannelLeftBridge ChannelStateChange ChannelUserevent ChannelVarset PlaybackFinished PlaybackStarted StasisEnd StasisStart

Base type for asynchronous events from Asterisk.

```
{
  "properties": {
    "application": {
      "required": true,
      "type": "string",
      "description": "Name of the application receiving the event."
    },
    "timestamp": {
      "required": false,
      "type": "Date",
      "description": "Time at which this event was created."
    }
  },
  "extends": "Message",
  "id": "Event",
  "description": "Base type for asynchronous events from Asterisk."
}
```

- type: string - Indicates the type of this message.
- application: string - Name of the application receiving the event.
- timestamp: Date *(optional)* - Time at which this event was created.

Message

Subtypes: ApplicationReplaced BridgeCreated BridgeDestroyed BridgeMerged ChannelCallerId ChannelCreated ChannelDestroyed ChannelDialplan ChannelDtmfReceived ChannelEnteredBridge ChannelHangupRequest ChannelLeftBridge ChannelStateChange ChannelUserevent ChannelVarset Event MissingParams PlaybackFinished PlaybackStarted StasisEnd StasisStart

Base type for errors and events

```
{
  "discriminator": "type",
  "properties": {
    "type": {
      "required": true,
      "type": "string",
      "description": "Indicates the type of this message."
    }
  },
  "id": "Message",
  "description": "Base type for errors and events"
}
```

- type: string - Indicates the type of this message.

MissingParams

Base type: Message

Error event sent when required params are missing.

```
{
  "properties": {
    "params": {
      "required": true,
      "type": "List[string]",
      "description": "A list of the missing parameters"
    }
  },
  "extends": "Message",
  "id": "MissingParams",
  "description": "Error event sent when required params are missing."
}
```

- type: string - Indicates the type of this message.
- params: List[string] - A list of the missing parameters

PlaybackFinished

Base type: Event

Event showing the completion of a media playback operation.

```
{
  "properties": {
    "playback": {
      "required": true,
      "type": "Playback",
      "description": "Playback control object"
    }
  },
  "extends": "Event",
  "id": "PlaybackFinished",
  "description": "Event showing the completion of a media playback operation."
}
```

- type: string - Indicates the type of this message.
- application: string - Name of the application receiving the event.
- timestamp: Date *(optional)* - Time at which this event was created.
- playback: Playback - Playback control object

PlaybackStarted

Base type: Event

Event showing the start of a media playback operation.

```
{
  "properties": {
    "playback": {
      "required": true,
      "type": "Playback",
      "description": "Playback control object"
    }
  },
  "extends": "Event",
  "id": "PlaybackStarted",
  "description": "Event showing the start of a media playback operation."
}
```

- type: string - Indicates the type of this message.
- application: string - Name of the application receiving the event.
- timestamp: Date *(optional)* - Time at which this event was created.
- playback: Playback - Playback control object

StasisEnd

Base type: Event

Notification that a channel has left a Stasis appliction.

```
{
  "properties": {
    "channel": {
      "required": true,
      "type": "Channel"
    }
  },
  "extends": "Event",
  "id": "StasisEnd",
  "description": "Notification that a channel has left a Stasis appliction."
}
```

- type: string - Indicates the type of this message.
- application: string - Name of the application receiving the event.
- timestamp: Date *(optional)* - Time at which this event was created.
- channel: Channel

StasisStart

Base type: Event

Notification that a channel has entered a Stasis appliction.

```
{
  "properties": {
    "args": {
      "required": true,
      "type": "List[string]",
      "description": "Arguments to the application"
    },
    "channel": {
      "required": true,
      "type": "Channel"
    }
  },
  "extends": "Event",
  "id": "StasisStart",
  "description": "Notification that a channel has entered a Stasis appliction."
}
```

- type: string - Indicates the type of this message.
- application: string - Name of the application receiving the event.
- timestamp: Date *(optional)* - Time at which this event was created.
- args: List[string] - Arguments to the application
- channel: Channel

Asterisk 12 Asterisk REST API

Asterisk

Method	Path	Return Model	Summary
GET	/asterisk/info	AsteriskInfo	Gets Asterisk system information.
GET	/asterisk/variable	Variable	Get the value of a global variable.
POST	/asterisk/variable	void	Set the value of a global variable.

GET /asterisk/info

Gets Asterisk system information.

Query parameters

- only: string - Filter information returned

GET /asterisk/variable

Get the value of a global variable.

Query parameters

- variable: string - **(required)** The variable to get

Error Responses

- 400 - Missing variable parameter.

POST /asterisk/variable

Set the value of a global variable.

Query parameters

- variable: string - **(required)** The variable to set
- value: string - The value to set the variable to

Error Responses

- 400 - Missing variable parameter.

Asterisk 12 Bridges REST API

Bridges

Method	Path	Return Model	Summary
GET	/bridges	List[Bridge]	List active bridges.
POST	/bridges	Bridge	Create a new bridge.
GET	/bridges/{bridgeId}	Bridge	Get bridge details.
DELETE	/bridges/{bridgeId}	void	Shut down a bridge.
POST	/bridges/{bridgeId}/addChannel	void	Add a channel to a bridge.
POST	/bridges/{bridgeId}/removeChannel	void	Remove a channel from a bridge.
POST	/bridges/{bridgeId}/mohStart	void	Play music on hold to a bridge or change the MOH class that is playing.
POST	/bridges/{bridgeId}/mohStop	void	Stop playing music on hold to a bridge.
POST	/bridges/{bridgeId}/play	Playback	Start playback of media on a bridge.
POST	/bridges/{bridgeId}/record	LiveRecording	Start a recording.

GET /bridges

List active bridges.

POST /bridges

Create a new bridge. This bridge persists until it has been shut down, or Asterisk has been shut down.

Query parameters

- type: string - Type of bridge to create.

GET /bridges/{bridgeId}

Get bridge details.

Path parameters

- bridgeId: string - Bridge's id

Error Responses

- 404 - Bridge not found

DELETE /bridges/{bridgeId}

Shut down a bridge. If any channels are in this bridge, they will be removed and resume whatever they were doing beforehand.

Path parameters

- bridgeId: string - Bridge's id

Error Responses

- 404 - Bridge not found

POST /bridges/{bridgeId}/addChannel

Add a channel to a bridge.

Path parameters

- bridgeId: string - Bridge's id

Query parameters

- channel: string - **(required)** Ids of channels to add to bridge
- role: string - Channel's role in the bridge

Error Responses

- 400 - Channel not found
- 404 - Bridge not found
- 409 - Bridge not in Stasis application
- 422 - Channel not in Stasis application

POST /bridges/{bridgeId}/removeChannel

Remove a channel from a bridge.

Path parameters

- bridgeId: string - Bridge's id

Query parameters

- channel: string - **(required)** Ids of channels to remove from bridge

Error Responses

- 400 - Channel not found
- 404 - Bridge not found
- 409 - Bridge not in Stasis application
- 422 - Channel not in this bridge

POST /bridges/{bridgeId}/mohStart

Play music on hold to a bridge or change the MOH class that is playing.

Path parameters

- bridgeId: string - Bridge's id

Query parameters

- mohClass: string - Channel's id

Error Responses

- 404 - Bridge not found
- 409 - Bridge not in Stasis application

POST /bridges/{bridgeId}/mohStop

Stop playing music on hold to a bridge. This will only stop music on hold being played via bridges/{bridgeId}/mohStart.

Path parameters

- bridgeId: string - Bridge's id

Error Responses

- 404 - Bridge not found
- 409 - Bridge not in Stasis application

POST /bridges/{bridgeId}/play

Start playback of media on a bridge. The media URI may be any of a number of URI's. You may use http: and https: URI's, as well as sound: and recording: URI's. This operation creates a playback resource that can be used to control the playback of media (pause, rewind, fast forward, etc.)

Path parameters

- bridgeId: string - Bridge's id

Query parameters

- media: string - **(required)** Media's URI to play.
- lang: string - For sounds, selects language for sound.
- offsetms: int - Number of media to skip before playing.
- skipms: int = 3000 - Number of milliseconds to skip for forward/reverse operations.

Error Responses

- 404 - Bridge not found
- 409 - Bridge not in a Stasis application

POST /bridges/{bridgeId}/record

Start a recording. This records the mixed audio from all channels participating in this bridge.

Path parameters

- bridgeId: string - Bridge's id

Query parameters

- name: string - **(required)** Recording's filename
- format: string - **(required)** Format to encode audio in
- maxDurationSeconds: int - Maximum duration of the recording, in seconds. 0 for no limit.
- maxSilenceSeconds: int - Maximum duration of silence, in seconds. 0 for no limit.
- ifExists: string = fail - Action to take if a recording with the same name already exists.
- beep: boolean - Play beep when recording begins
- terminateOn: string = none - DTMF input to terminate recording.

Asterisk 12 Channels REST API

Channels

Method	Path	Return Model	Summary
GET	/channels	List[Channel]	List active channels.
POST	/channels	void	Create a new channel (originate).
GET	/channels/{channelId}	Channel	Channel details.
DELETE	/channels/{channelId}	void	Delete (i.e. hangup) a channel.
POST	/channels/{channelId}/dial	Dialed	Create a new channel (originate) and bridge to this channel.
POST	/channels/{channelId}/continue	void	Exit application; continue execution in the dialplan.
POST	/channels/{channelId}/answer	void	Answer a channel.

POST	/channels/{channelId}/mute	void	Mute a channel.
POST	/channels/{channelId}/unmute	void	Unmute a channel.
POST	/channels/{channelId}/hold	void	Hold a channel.
POST	/channels/{channelId}/unhold	void	Remove a channel from hold.
POST	/channels/{channelId}/mohstart	void	Play music on hold to a channel.
POST	/channels/{channelId}/mohstop	void	Stop playing music on hold to a channel.
POST	/channels/{channelId}/play	Playback	Start playback of media.
POST	/channels/{channelId}/record	LiveRecording	Start a recording.
GET	/channels/{channelId}/variable	Variable	Get the value of a channel variable or function.
POST	/channels/{channelId}/variable	void	Set the value of a channel variable or function.

GET /channels

List active channels.

POST /channels

Create a new channel (originate).

Query parameters

- endpoint: string - **(required)** Endpoint to call.
- extension: string - The extension to dial after the endpoint answers
- context: string - The context to dial after the endpoint answers. If omitted, uses 'default'
- priority: long - The priority to dial after the endpoint answers. If omitted, uses 1
- app: string - The application name to pass to the Stasis application.
- appArgs: string - The application arguments to pass to the Stasis application.
- callerId: string - CallerID to use when dialing the endpoint or extension.
- timeout: int = 30 - Timeout (in seconds) before giving up dialing, or -1 for no timeout.

Error Responses

- 400 - Invalid parameters for originating a channel.

GET /channels/{channelId}

Channel details.

Path parameters

- channelId: string - Channel's id

Error Responses

- 404 - Channel not found

DELETE /channels/{channelId}

Delete (i.e. hangup) a channel.

Path parameters

- channelId: string - Channel's id

Error Responses

- 404 - Channel not found

POST /channels/{channelId}/dial

Create a new channel (originate) and bridge to this channel.

Path parameters

- channelId: string - Channel's id

Query parameters

- endpoint: string - Endpoint to call. If not specified, dial is routed via dialplan
- extension: string - Extension to dial
- context: string - When routing via dialplan, the context use. If omitted, uses 'default'
- timeout: int = 30 - Timeout (in seconds) before giving up dialing, or -1 for no timeout.

Error Responses

- 404 - Channel not found
- 409 - Channel not in a Stasis application

POST /channels/{channelId}/continue

Exit application; continue execution in the dialplan.

Path parameters

- channelId: string - Channel's id

Query parameters

- context: string - The context to continue to.
- extension: string - The extension to continue to.

- priority: int - The priority to continue to.

Error Responses

- 404 - Channel not found
- 409 - Channel not in a Stasis application

POST /channels/{channelId}/answer

Answer a channel.

Path parameters

- channelId: string - Channel's id

Error Responses

- 404 - Channel not found
- 409 - Channel not in a Stasis application

POST /channels/{channelId}/mute

Mute a channel.

Path parameters

- channelId: string - Channel's id

Query parameters

- direction: string = both - Direction in which to mute audio

Error Responses

- 404 - Channel not found
- 409 - Channel not in a Stasis application

POST /channels/{channelId}/unmute

Unmute a channel.

Path parameters

- channelId: string - Channel's id

Query parameters

- direction: string = both - Direction in which to unmute audio

Error Responses

- 404 - Channel not found
- 409 - Channel not in a Stasis application

POST /channels/{channelId}/hold

Hold a channel.

Path parameters

- channelId: string - Channel's id

Error Responses

- 404 - Channel not found
- 409 - Channel not in a Stasis application

POST /channels/{channelId}/unhold

Remove a channel from hold.

Path parameters

- channelId: string - Channel's id

Error Responses

- 404 - Channel not found
- 409 - Channel not in a Stasis application

POST /channels/{channelId}/mohstart

Play music on hold to a channel. Using media operations such as playOnChannel on a channel playing MOH in this manner will suspend MOH without resuming automatically. If continuing music on hold is desired, the stasis application must reinitiate music on hold.

Path parameters

- channelId: string - Channel's id

Query parameters

- mohClass: string - Music on hold class to use

Error Responses

- 404 - Channel not found
- 409 - Channel not in a Stasis application

POST /channels/{channelId}/mohstop

Stop playing music on hold to a channel.

Path parameters

- channelId: string - Channel's id

Error Responses

- 404 - Channel not found
- 409 - Channel not in a Stasis application

POST /channels/{channelId}/play

Start playback of media. The media URI may be any of a number of URI's. You may use http: and https: URI's, as well as sound: and recording: URI's. This operation creates a playback resource that can be used to control the playback of media (pause, rewind, fast forward, etc.)

Path parameters

- channelId: string - Channel's id

Query parameters

- media: string - **(required)** Media's URI to play.
- lang: string - For sounds, selects language for sound.
- offsetms: int - Number of media to skip before playing.
- skipms: int = 3000 - Number of milliseconds to skip for forward/reverse operations.

Error Responses

- 404 - Channel not found
- 409 - Channel not in a Stasis application

POST /channels/{channelId}/record

Start a recording. Record audio from a channel. Note that this will not capture audio sent to the channel. The bridge itself has a record feature if that's what you want.

Path parameters

- channelId: string - Channel's id

Query parameters

- name: string - **(required)** Recording's filename
- format: string - **(required)** Format to encode audio in
- maxDurationSeconds: int - Maximum duration of the recording, in seconds. 0 for no limit
- maxSilenceSeconds: int - Maximum duration of silence, in seconds. 0 for no limit
- ifExists: string = fail - Action to take if a recording with the same name already exists.
- beep: boolean - Play beep when recording begins
- terminateOn: string = none - DTMF input to terminate recording

Error Responses

- 400 - Invalid parameters
- 404 - Channel not found
- 409 - Channel is not in a Stasis application; the channel is currently bridged with other channels; A recording with the same name is currently in progress.

GET /channels/{channelId}/variable

Get the value of a channel variable or function.

Path parameters

- channelId: string - Channel's id

Query parameters

- variable: string - **(required)** The channel variable or function to get

Error Responses

- 400 - Missing variable parameter.
- 404 - Channel not found
- 409 - Channel not in a Stasis application

POST /channels/{channelId}/variable

Set the value of a channel variable or function.

Path parameters

- channelId: string - Channel's id

Query parameters

- variable: string - **(required)** The channel variable or function to set
- value: string - The value to set the variable to

Error Responses

- 400 - Missing variable parameter.
- 404 - Channel not found
- 409 - Channel not in a Stasis application

Asterisk 12 Endpoints REST API

Endpoints

Method	Path	Return Model	Summary
GET	/endpoints	List[Endpoint]	List all endoints.
GET	/endpoints/{tech}	List[Endpoint]	List available endoints for a given endpoint technology.
GET	/endpoints/{tech}/{resource}	Endpoint	Details for an endpoint.

GET /endpoints

List all endoints.

GET /endpoints/{tech}

List available endoints for a given endpoint technology.

Path parameters

- tech: string - Technology of the endpoints (sip,iax2,...)

GET /endpoints/{tech}/{resource}

Details for an endpoint.

Path parameters

- tech: string - Technology of the endpoint
- resource: string - ID of the endpoint

Asterisk 12 Events REST API

Events

Method	Path	Return Model	Summary
GET	/events	Message	WebSocket connection for events.

GET /events

WebSocket connection for events.

Query parameters

- app: string - **(required)** Applications to subscribe to.

Asterisk 12 Recordings REST API

Recordings

Method	Path	Return Model	Summary
GET	/recordings/stored	List[StoredRecording]	List recordings that are complete.
GET	/recordings/stored/{recordingName}	StoredRecording	Get a stored recording's details.
DELETE	/recordings/stored/{recordingName}	void	Delete a stored recording.
GET	/recordings/live	List[LiveRecording]	List libe recordings.
GET	/recordings/live/{recordingName}	LiveRecording	List live recordings.
DELETE	/recordings/live/{recordingName}	void	Stop a live recording and discard it.
POST	/recordings/live/{recordingName}/stop	void	Stop a live recording and store it.
POST	/recordings/live/{recordingName}/pause	void	Pause a live recording.
POST	/recordings/live/{recordingName}/unpause	void	Unpause a live recording.
POST	/recordings/live/{recordingName}/mute	void	Mute a live recording.
POST	/recordings/live/{recordingName}/unmute	void	Unmute a live recording.

GET /recordings/stored

List recordings that are complete.

GET /recordings/stored/{recordingName}

Get a stored recording's details.

Path parameters

- recordingName: string - The name of the recording

DELETE /recordings/stored/{recordingName}

Delete a stored recording.

Path parameters

- recordingName: string - The name of the recording

GET /recordings/live

List libe recordings.

GET /recordings/live/{recordingName}

List live recordings.

Path parameters

- recordingName: string - The name of the recording

Error Responses

- 404 - Recording not found

DELETE /recordings/live/{recordingName}

Stop a live recording and discard it.

Path parameters

- recordingName: string - The name of the recording

Error Responses

- 404 - Recording not found

POST /recordings/live/{recordingName}/stop

Stop a live recording and store it.

Path parameters

- recordingName: string - The name of the recording

Error Responses

- 404 - Recording not found

POST /recordings/live/{recordingName}/pause

Pause a live recording. Pausing a recording suspends silence detection, which will be restarted when the recording is unpaused. Paused time is not included in the accounting for maxDurationSeconds.

Path parameters

- recordingName: string - The name of the recording

Error Responses

- 404 - Recording not found
- 409 - Recording not in session

POST /recordings/live/{recordingName}/unpause

Unpause a live recording.

Path parameters

- recordingName: string - The name of the recording

Error Responses

- 404 - Recording not found
- 409 - Recording not in session

POST /recordings/live/{recordingName}/mute

Mute a live recording. Muting a recording suspends silence detection, which will be restarted when the recording is unmuted.

Path parameters

- recordingName: string - The name of the recording

Error Responses

- 404 - Recording not found
- 409 - Recording not in session

POST /recordings/live/{recordingName}/unmute

Unmute a live recording.

Path parameters

- recordingName: string - The name of the recording

Error Responses

- 404 - Recording not found
- 409 - Recording not in session

Asterisk 12 Playback REST API

Playback

Method	Path	Return Model	Summary
GET	/playback/{playbackId}	Playback	Get a playback's details.
DELETE	/playback/{playbackId}	Playback	Stop a playback.
POST	/playback/{playbackId}/control	void	Get a playback's details.

GET /playback/{playbackId}

Get a playback's details.

Path parameters

- playbackId: string - Playback's id

DELETE /playback/{playbackId}

Stop a playback.

Path parameters

- playbackId: string - Playback's id

POST /playback/{playbackId}/control

Get a playback's details.

Path parameters

- playbackId: string - Playback's id

Query parameters

- operation: string - **(required)** Operation to perform on the playback.

Error Responses

- 400 - The provided operation parameter was invalid
- 404 - The playback cannot be found
- 409 - The operation cannot be performed in the playback's current state

Asterisk 12 Sounds REST API

Sounds

Method	Path	Return Model	Summary
GET	/sounds	List[Sound]	List all sounds.
GET	/sounds/{soundId}	Sound	Get a sound's details.

GET /sounds

List all sounds.

Query parameters

- lang: string - Lookup sound for a specific language.
- format: string - Lookup sound in a specific format.

GET /sounds/{soundId}

Get a sound's details.

Path parameters

- soundId: string - Sound's id

Asterisk 12 Dialplan Applications

Asterisk 12 Application_AddQueueMember

AddQueueMember()

Synopsis

Dynamically adds queue members.

Description

Dynamically adds interface to an existing queue. If the interface is already in the queue it will return an error.

This application sets the following channel variable upon completion:

- `AQMSTATUS` - The status of the attempt to add a queue member as a text string.
  - ADDED
  - MEMBERALREADY
  - NOSUCHQUEUE

Syntax

```
AddQueueMember(queuename,interface,penalty,options,membername,stateinterfa
ce)
```

Arguments

- `queuename`
- `interface`
- `penalty`
- `options`
- `membername`
- `stateinterface`

See Also

- Asterisk 12 Application_Queue
- Asterisk 12 Application_QueueLog

- Asterisk 12 Application_AddQueueMember
- Asterisk 12 Application_RemoveQueueMember
- Asterisk 12 Application_PauseQueueMember
- Asterisk 12 Application_UnpauseQueueMember
- Asterisk 12 Function_QUEUE_VARIABLES
- Asterisk 12 Function_QUEUE_MEMBER
- Asterisk 12 Function_QUEUE_MEMBER_COUNT
- Asterisk 12 Function_QUEUE_EXISTS
- Asterisk 12 Function_QUEUE_WAITING_COUNT
- Asterisk 12 Function_QUEUE_MEMBER_LIST
- Asterisk 12 Function_QUEUE_MEMBER_PENALTY

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_ADSIProg

ADSIProg()

Synopsis

Load Asterisk ADSI Scripts into phone

Description

This application programs an ADSI Phone with the given script

Syntax

```
ADSIProg([script])
```

Arguments

- `script` - adsi script to use. If not given uses the default script `asterisk.adsi`

See Also

- Asterisk 12 Application_GetCPEID
- `adsi.conf`

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_AELSub

AELSub()

Synopsis

Launch subroutine built with AEL

Description

Execute the named subroutine, defined in AEL, from another dialplan language, such as extensions.conf, Realtime extensions, or Lua.

The purpose of this application is to provide a sane entry point into AEL subroutines, the implementation of which may change from time to time.

Syntax

```
AELSub(routine[,args])
```

Arguments

- `routine` - Named subroutine to execute.
- `args`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_AgentLogin

AgentLogin()

Synopsis

Login an agent.

Description

Login an agent to the system. Any agent authentication is assumed to already be done by dialplan. While logged in, the agent can receive calls and will hear the sound file specified by the config option custom_beep when a new call comes in for the agent. Login failures will continue in the dialplan with `AGENT_STATUS` set.

Before logging in, you can setup on the real agent channel the CHANNEL(dtmf-features) an agent will have when talking to a caller and you can setup on the channel running this application the CONNECTEDLINE() information the agent will see while waiting for a caller.

AGENT_STATUS enumeration values:

- `INVALID` - The specified agent is invalid.
- `ALREADY_LOGGED_IN` - The agent is already logged in.

> **Note**
> The Agents:*AgentId* device state is available to monitor the status of the agent.

Syntax

```
AgentLogin(AgentId,options)
```

Arguments

- `AgentId`
- `options`

- s - silent login - do not announce the login ok segment after agent logged on.

See Also

- Asterisk 12 Application_Authenticate
- Asterisk 12 Application_Queue
- Asterisk 12 Application_AddQueueMember
- Asterisk 12 Application_RemoveQueueMember
- Asterisk 12 Application_PauseQueueMember
- Asterisk 12 Application_UnpauseQueueMember
- Asterisk 12 Function_AGENT
- Asterisk 12 Function_CHANNEL(dtmf-features)
- Asterisk 12 Function_CONNECTEDLINE()
- `agents.conf`
- `queues.conf`

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_AgentRequest

AgentRequest()

Synopsis

Request an agent to connect with the channel.

Description

Request an agent to connect with the channel. Failure to find and alert an agent will continue in the dialplan with `AGENT_STATUS` set.

AGENT_STATUS enumeration values:

- `INVALID` - The specified agent is invalid.
- `NOT_LOGGED_IN` - The agent is not available.
- `BUSY` - The agent is on another call.
- `ERROR` - Alerting the agent failed.

Syntax

```
AgentRequest(AgentId)
```

Arguments

- `AgentId`

See Also

- Asterisk 12 Application_AgentLogin

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_AGI

AGI()

Synopsis

Executes an AGI compliant application.

Description

Executes an Asterisk Gateway Interface compliant program on a channel. AGI allows Asterisk to launch external programs written in any language to control a telephony channel, play audio, read DTMF digits, etc. by communicating with the AGI protocol on **stdin** and **stdout**. As of `1.6.0`, this channel will not stop dialplan execution on hangup inside of this application. Dialplan execution will continue normally, even upon hangup until the AGI application signals a desire to stop (either by exiting or, in the case of a net script, by closing the connection). A locally executed AGI script will receive SIGHUP on hangup from the channel except when using DeadAGI. A fast AGI server will correspondingly receive a HANGUP inline with the command dialog. Both of theses signals may be disabled by setting the `AGISIGHUP` channel variable to `no` before executing the AGI application. Alternatively, if you would like the AGI application to exit immediately after a channel hangup is detected, set the `AGIEXITONHANGUP` variable to `yes`.

Use the CLI command `agi show commands` to list available agi commands.

This application sets the following channel variable upon completion:

- `AGISTATUS` - The status of the attempt to the run the AGI script text string, one of:
    - SUCCESS
    - FAILURE
    - NOTFOUND
    - HANGUP

Syntax

```
AGI(commandarg1arg2[...])
```

Arguments

- `command`
- `args`
    - `arg1`
    - `arg2`

See Also

- Asterisk 12 Application_EAGI
- Asterisk 12 Application_DeadAGI

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_AlarmReceiver

AlarmReceiver()

Synopsis

Provide support for receiving alarm reports from a burglar or fire alarm panel.

Description

This application should be called whenever there is an alarm panel calling in to dump its events. The application will handshake with the alarm panel, and receive events, validate them, handshake them, and store them until the panel hangs up. Once the panel hangs up, the application will run the system command specified by the eventcmd setting in `alarmreceiver.conf` and pipe the events to the standard input of the application. The configuration file also contains settings for DTMF timing, and for the loudness of the acknowledgement tones.

> **Note**
> Few Ademco DTMF signalling formats are detected automaticaly: Contact ID, Express 4+1, Express 4+2, High Speed and Super Fast.

The application is affected by the following variables:

- `ALARMRECEIVER_CALL_LIMIT` - Maximum call time, in milliseconds.
  If set, this variable causes application to exit after the specified time.
- `ALARMRECEIVER_RETRIES_LIMIT` - Maximum number of retries per call.
  If set, this variable causes application to exit after the specified number of messages.

Syntax

```
AlarmReceiver()
```

Arguments

See Also

- `alarmreceiver.conf`

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_AMD

AMD()

Synopsis

Attempt to detect answering machines.

Description

This application attempts to detect answering machines at the beginning of outbound calls. Simply call this application after the call has been answered (outbound only, of course).

When loaded, AMD reads amd.conf and uses the parameters specified as default values. Those default values get overwritten when the calling AMD with parameters.

This application sets the following channel variables:

- `AMDSTATUS` - This is the status of the answering machine detection
    - MACHINE
    - HUMAN
    - NOTSURE
    - HANGUP
- `AMDCAUSE` - Indicates the cause that led to the conclusion
    - TOOLONG - Total Time.
    - INITIALSILENCE - Silence Duration - Initial Silence.
    - HUMAN - Silence Duration - afterGreetingSilence.
    - LONGGREETING - Voice Duration - Greeting.
    - MAXWORDLENGTH - Word Count - maximum number of words.

Syntax

```
AMD([initialSilence[,greeting[,afterGreetingSilence[,totalAnalysis
Time[,miniumWordLength[,betweenWordSilence[,maximumNumberOfWords[,silenceT
hreshold[,maximumWordLength]]]]]]]]])
```

Arguments

- `initialSilence` - Is maximum initial silence duration before greeting.
  If this is exceeded set as MACHINE
- `greeting` - is the maximum length of a greeting.
  If this is exceeded set as MACHINE
- `afterGreetingSilence` - Is the silence after detecting a greeting.
  If this is exceeded set as HUMAN
- `totalAnalysis Time` - Is the maximum time allowed for the algorithm
  to decide HUMAN or MACHINE
- `miniumWordLength` - Is the minimum duration of Voice considered to be a word
- `betweenWordSilence` - Is the minimum duration of silence after a word to consider the audio that follows to be a new word
- `maximumNumberOfWords` - Is the maximum number of words in a greeting
  If this is exceeded set as MACHINE
- `silenceThreshold` - How long do we consider silence
- `maximumWordLength` - Is the maximum duration of a word to accept.
  If exceeded set as MACHINE

See Also

- Asterisk 12 Application_WaitForSilence
- Asterisk 12 Application_WaitForNoise

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_Answer

Answer()

Synopsis

Answer a channel if ringing.

Description

If the call has not been answered, this application will answer it. Otherwise, it has no effect on the call.

Syntax

```
Answer(delay,nocdr)
```

Arguments

- `delay` - Asterisk will wait this number of milliseconds before returning to the dialplan after answering the call.
- `nocdr` - Asterisk will send an answer signal to the calling phone, but will not set the disposition or answer time in the CDR for this call.

See Also

- Asterisk 12 Application_Hangup

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_Authenticate

Authenticate()

Synopsis

Authenticate a user

Description

This application asks the caller to enter a given password in order to continue dialplan execution.

If the password begins with the `/` character, it is interpreted as a file which contains a list of valid passwords, listed 1 password per line in the file.

When using a database key, the value associated with the key can be anything.

Users have three attempts to authenticate before the channel is hung up.

Syntax

```
Authenticate(password[,options[,maxdigits[,prompt]]])
```

Arguments

- `password` - Password the user should know
- `options`
    - `a` - Set the channels' account code to the password that is entered
    - `d` - Interpret the given path as database key, not a literal file.
    - `m` - Interpret the given path as a file which contains a list of account codes and password hashes delimited with `:`, listed one per line in the file. When one of the passwords is matched, the channel will have its account code set to the corresponding account code in the file.
    - `r` - Remove the database key upon successful entry (valid with `d` only)
- `maxdigits` - maximum acceptable number of digits. Stops reading after maxdigits have been entered (without requiring the user to press the `#` key). Defaults to `0` - no limit - wait for the user press the `#` key.
- `prompt` - Override the agent-pass prompt file.

See Also

- Asterisk 12 Application_VMAuthenticate
- Asterisk 12 Application_DISA

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_BackGround

BackGround()

Synopsis

Play an audio file while waiting for digits of an extension to go to.

Description

This application will play the given list of files **(do not put extension)** while waiting for an extension to be dialed by the calling channel. To continue waiting for digits after this application has finished playing files, the `WaitExten` application should be used.

If one of the requested sound files does not exist, call processing will be terminated.

This application sets the following channel variable upon completion:

- `BACKGROUNDSTATUS` - The status of the background attempt as a text string.
    - SUCCESS
    - FAILED

Syntax

```
BackGround(filename1&filename2[&...],options,langoverride,context)
```

Arguments

- `filenames`
    - `filename1`
    - `filename2`
- `options`
    - `s` - Causes the playback of the message to be skipped if the channel is not in the `up` state (i.e. it hasn't been answered yet). If this happens, the application will return immediately.
    - `n` - Don't answer the channel before playing the files.
    - `m` - Only break if a digit hit matches a one digit extension in the destination context.
- `langoverride` - Explicitly specifies which language to attempt to use for the requested sound files.
- `context` - This is the dialplan context that this application will use when exiting to a dialed extension.

See Also

- Asterisk 12 Application_ControlPlayback
- Asterisk 12 Application_WaitExten
- Asterisk 12 Application_BackgroundDetect
- Asterisk 12 Function_TIMEOUT

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_BackgroundDetect

BackgroundDetect()

Synopsis

Background a file with talk detect.

Description

Plays back *filename*, waiting for interruption from a given digit (the digit must start the beginning of a valid extension, or it will be ignored). During the playback of the file, audio is monitored in the receive direction, and if a period of non-silence which is greater than *min* ms yet less than *max* ms is followed by silence for at least *sil* ms, which occurs during the first *analysistime* ms, then the audio playback is aborted and processing jumps to the *talk* extension, if available.

Syntax

```
BackgroundDetect(filename,sil,min,max,analysistime)
```

Arguments

- `filename`
- `sil` - If not specified, defaults to `1000`.
- `min` - If not specified, defaults to `100`.
- `max` - If not specified, defaults to `infinity`.
- `analysistime` - If not specified, defaults to `infinity`.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_Bridge

Bridge()

Synopsis

Bridge two channels.

Description

Allows the ability to bridge two channels via the dialplan.

This application sets the following channel variable upon completion:

- `BRIDGERESULT` - The result of the bridge attempt as a text string.
    - SUCCESS
    - FAILURE
    - LOOP
    - NONEXISTENT
    - INCOMPATIBLE

Syntax

```
Bridge(channel,options)
```

Arguments

- `channel` - The current channel is bridged to the specified *channel*.
- `options`
    - `p` - Play a courtesy tone to *channel*.
    - `F` - When the bridger hangs up, transfer the **bridged** party to the specified destination and **start** execution at that location.
        - `context`
        - `exten`
        - `priority`
    - `F` - When the bridger hangs up, transfer the **bridged** party to the next priority ofthe current extension and **start** execution at that location.
    - `h` - Allow the called party to hang up by sending the `*` DTMF digit.
    - `H` - Allow the calling party to hang up by pressing the `*` DTMF digit.
    - `k` - Allow the called party to enable parking of the call by sending the DTMF sequence defined for call parking in `features.conf`.
    - `K` - Allow the calling party to enable parking of the call by sending the DTMF sequence defined for call parking in `features.conf`.
    - `L(xyz)` - Limit the call to *x* ms. Play a warning when *y* ms are left. Repeat the warning every *z* ms. The following special variables can be used with this option:
        - `LIMIT_PLAYAUDIO_CALLER` - Play sounds to the caller. yes|no (default yes)
        - `LIMIT_PLAYAUDIO_CALLEE` - Play sounds to the callee. yes|no
        - `LIMIT_TIMEOUT_FILE` - File to play when time is up.
        - `LIMIT_CONNECT_FILE` - File to play when call begins.
        - `LIMIT_WARNING_FILE` - File to play as warning if *y* is defined. The default is to say the time remaining.
    - `S` - Hang up the call after *x* seconds **after** the called party has answered the call.
    - `t` - Allow the called party to transfer the calling party by sending the DTMF sequence defined in `features.conf`.
    - `T` - Allow the calling party to transfer the called party by sending the DTMF sequence defined in `features.conf`.
    - `w` - Allow the called party to enable recording of the call by sending the DTMF sequence defined for one-touch recording in `features.conf`.
    - `W` - Allow the calling party to enable recording of the call by sending the DTMF sequence defined for one-touch recording in `features.conf`.
    - `x` - Cause the called party to be hung up after the bridge, instead of being restarted in the dialplan.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_BridgeWait

BridgeWait()

Synopsis

Put a call into the holding bridge.

Description

This application places the incoming channel into a holding bridge. The channel will then wait in the holding bridge until some event occurs which removes it from the holding bridge.

Note

This application will answer calls which haven't already been answered.

Syntax

```
BridgeWait(name[,role,options])
```

Arguments

- `name` - Name of the holding bridge to join. This is a handle for `BridgeWait` only and does not affect the actual bridges that are created. If not provided, the reserved name `default` will be used.
- `role` - Defines the channel's purpose for entering the holding bridge. Values are case sensitive.
  - `participant` - The channel will enter the holding bridge to be placed on hold until it is removed from the bridge for some reason. (default)
  - `announcer` - The channel will enter the holding bridge to make announcements to channels that are currently in the holding bridge. While an announcer is present, holding for the participants will be suspended.
- `options`
  - `m` - The specified MOH class will be used/suggested for music on hold operations. This option will only be useful for entertainment modes that use it (m and h).
    - `class`
  - `e` - Which entertainment mechanism should be used while on hold in the holding bridge. Only the first letter is read.
    - `m` - Play music on hold (default)
    - `r` - Ring without pause
    - `s` - Generate silent audio
    - `h` - Put the channel on hold
    - `n` - No entertainment
  - `S` - Automatically exit the bridge and return to the PBX after **duration** seconds.
    - `duration`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_Busy

Busy()

Synopsis

Indicate the Busy condition.

Description

This application will indicate the busy condition to the calling channel.

Syntax

```
Busy(timeout)
```

Arguments

- `timeout` - If specified, the calling channel will be hung up after the specified number of seconds. Otherwise, this application will wait until the calling channel hangs up.

See Also

- Asterisk 12 Application_Congestion
- Asterisk 12 Application_Progress
- Asterisk 12 Application_Playtones
- Asterisk 12 Application_Hangup

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_CallCompletionCancel

CallCompletionCancel()

Synopsis

Cancel call completion service

Description

Cancel a Call Completion Request.

This application sets the following channel variables:

- `CC_CANCEL_RESULT` - This is the returned status of the cancel.
    - SUCCESS
    - FAIL
- `CC_CANCEL_REASON` - This is the reason the cancel failed.
    - NO_CORE_INSTANCE
    - NOT_GENERIC
    - UNSPECIFIED

Syntax

```
CallCompletionCancel()
```

Arguments

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_CallCompletionRequest

CallCompletionRequest()

Synopsis

Request call completion service for previous call

Description

Request call completion service for a previously failed call attempt.

This application sets the following channel variables:

- `CC_REQUEST_RESULT` - This is the returned status of the request.

    - SUCCESS
    - FAIL
- `CC_REQUEST_REASON` - This is the reason the request failed.
    - NO_CORE_INSTANCE
    - NOT_GENERIC
    - TOO_MANY_REQUESTS
    - UNSPECIFIED

Syntax

```
CallCompletionRequest()
```

Arguments

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_CELGenUserEvent

CELGenUserEvent()

Synopsis

Generates a CEL User Defined Event.

Description

A CEL event will be immediately generated by this channel, with the supplied name for a type.

Syntax

```
CELGenUserEvent(event-name[extra])
```

Arguments

- `event-name`
    - `event-name`
    - `extra` - Extra text to be included with the event.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_ChangeMonitor

ChangeMonitor()

Synopsis

Change monitoring filename of a channel.

Description

Changes monitoring filename of a channel. Has no effect if the channel is not monitored.

Syntax

```
ChangeMonitor(filename_base)
```

Arguments

- `filename_base` - The new filename base to use for monitoring this channel.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_ChanIsAvail

ChanIsAvail()

Synopsis

Check channel availability

Description

This application will check to see if any of the specified channels are available.

This application sets the following channel variables:

- `AVAILCHAN` - The name of the available channel, if one exists
- `AVAILORIGCHAN` - The canonical channel name that was used to create the channel
- `AVAILSTATUS` - The device state for the device
- `AVAILCAUSECODE` - The cause code returned when requesting the channel

Syntax

```
ChanIsAvail(Technology2/Resource2[&...][,options])
```

Arguments

- `Technology/Resource` - ** `Technology2/Resource2` - Optional extra devices to check
  If you need more then one enter them as Technology2/Resource2&Technology3/Resourse3&.....
  Specification of the device(s) to check. These must be in the format of `Technology/Resource`, where *Technology* represents a particular channel driver, and *Resource* represents a resource available to that particular channel driver.
- `options`
  - `a` - Check for all available channels, not only the first one
  - `s` - Consider the channel unavailable if the channel is in use at all
  - `t` - Simply checks if specified channels exist in the channel list

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_ChannelRedirect

ChannelRedirect()

Synopsis

Redirects given channel to a dialplan target

Description

Sends the specified channel to the specified extension priority

This application sets the following channel variables upon completion

- `CHANNELREDIRECT_STATUS` - Are set to the result of the redirection
    - NOCHANNEL
    - SUCCESS

Syntax

```
ChannelRedirect(channel[,context[,extension,priority]])
```

Arguments

- `channel`
- `context`
- `extension`
- `priority`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_ChanSpy

ChanSpy()

Synopsis

Listen to a channel, and optionally whisper into it.

Description

This application is used to listen to the audio from an Asterisk channel. This includes the audio coming in and out of the channel being spied on. If the `chanprefix` parameter is specified, only channels beginning with this string will be spied upon.

While spying, the following actions may be performed:

- Dialing # cycles the volume level.
- Dialing `*` will stop spying and look for another channel to spy on.
- Dialing a series of digits followed by # builds a channel name to append to 'chanprefix'. For example, executing ChanSpy(Agent) and then dialing the digits '1234#' while spying will begin spying on the channel 'Agent/1234'. Note that this feature will be overridden if the 'd' option is used

Note

The *X* option supersedes the three features above in that if a valid single digit extension exists in the correct context ChanSpy will exit to it. This also disables choosing a channel based on `chanprefix` and a digit sequence.

Syntax

```
ChanSpy(chanprefix,options)
```

Arguments

- `chanprefix`
- `options`
    - `b` - Only spy on channels involved in a bridged call.
    - `B` - Instead of whispering on a single channel barge in on both channels involved in the call.
    - `c`
        - `digit` - Specify a DTMF digit that can be used to spy on the next available channel.
    - `d` - Override the typical numeric DTMF functionality and instead use DTMF to switch between spy modes.
        - `4` - spy mode
        - `5` - whisper mode
        - `6` - barge mode
    - `e` - Enable **enforced** mode, so the spying channel can only monitor extensions whose name is in the *ext* : delimited list.
        - `ext`
    - `E` - Exit when the spied-on channel hangs up.
    - `g`
        - `grp` - Only spy on channels in which one or more of the groups listed in *grp* matches one or more groups from the `SPYGROUP` variable set on the channel to be spied upon.
    - `n` - Say the name of the person being spied on if that person has recorded his/her name. If a context is specified, then that voicemail context will be searched when retrieving the name, otherwise the `default` context be used when searching for the name (i.e. if SIP/1000 is the channel being spied on and no mailbox is specified, then `1000` will be used when searching for the name).
        - `mailbox`
        - `context`
    - `o` - Only listen to audio coming from this channel.
    - `q` - Don't play a beep when beginning to spy on a channel, or speak the selected channel name.
    - `r` - Record the session to the monitor spool directory. An optional base for the filename may be specified. The default is `chanspy`.
        - `basename`
    - `s` - Skip the playback of the channel type (i.e. SIP, IAX, etc) when speaking the selected channel name.
    - `S` - Stop when no more channels are left to spy on.
    - `v` - Adjust the initial volume in the range from `-4` to `4`. A negative value refers to a quieter setting.
        - `value`
    - `w` - Enable `whisper` mode, so the spying channel can talk to the spied-on channel.
    - `W` - Enable `private whisper` mode, so the spying channel can talk to the spied-on channel but cannot listen to that channel.
    - `x`
        - `digit` - Specify a DTMF digit that can be used to exit the application.
    - `X` - Allow the user to exit ChanSpy to a valid single digit numeric extension in the current context or the context specified by the `SPY_EXIT_CONTEXT` channel variable. The name of the last channel that was spied on will be stored in the `SPY_CHANNEL` variable.

See Also

- Asterisk 12 Application_ExtenSpy
- Asterisk 12 ManagerEvent_ChanSpyStart
- Asterisk 12 ManagerEvent_ChanSpyStop

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_ClearHash

ClearHash()

Synopsis

Clear the keys from a specified hashname.

Description

Clears all keys out of the specified *hashname*.

Syntax

```
ClearHash(hashname)
```

Arguments

- `hashname`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_ConfBridge

ConfBridge()

Synopsis

Conference bridge application.

Description

Enters the user into a specified conference bridge. The user can exit the conference by hangup or DTMF menu option.

Syntax

```
ConfBridge(conference,bridge_profile,user_profile,menu)
```

Arguments

- `conference` - Name of the conference bridge. You are not limited to just numbers.
- `bridge_profile` - The bridge profile name from confbridge.conf. When left blank, a dynamically built bridge profile created by the CONFBRIDGE dialplan function is searched for on the channel and used. If no dynamic profile is present, the 'default_bridge' profile found in confbridge.conf is used.
  It is important to note that while user profiles may be unique for each participant, mixing bridge profiles on a single conference is _NOT_ recommended and will produce undefined results.
- `user_profile` - The user profile name from confbridge.conf. When left blank, a dynamically built user profile created by the

CONFBRIDGE dialplan function is searched for on the channel and used. If no dynamic profile is present, the 'default_user' profile found in confbridge.conf is used.

- `menu` - The name of the DTMF menu in confbridge.conf to be applied to this channel. No menu is applied by default if this option is left blank.

See Also

- Asterisk 12 Application_ConfBridge
- Asterisk 12 Function_CONFBRIDGE
- Asterisk 12 Function_CONFBRIDGE_INFO

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_Congestion

Congestion()

Synopsis

Indicate the Congestion condition.

Description

This application will indicate the congestion condition to the calling channel.

Syntax

```
Congestion(timeout)
```

Arguments

- `timeout` - If specified, the calling channel will be hung up after the specified number of seconds. Otherwise, this application will wait until the calling channel hangs up.

See Also

- Asterisk 12 Application_Busy
- Asterisk 12 Application_Progress
- Asterisk 12 Application_Playtones
- Asterisk 12 Application_Hangup

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_ContinueWhile

ContinueWhile()

Synopsis

Restart a While loop.

Description

Returns to the top of the while loop and re-evaluates the conditional.

Syntax

```
ContinueWhile()
```

Arguments

See Also

- Asterisk 12 Application_While
- Asterisk 12 Application_EndWhile
- Asterisk 12 Application_ExitWhile

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_ControlPlayback

ControlPlayback()

Synopsis

Play a file with fast forward and rewind.

Description

This application will play back the given *filename*.

It sets the following channel variables upon completion:

- `CPLAYBACKSTATUS` - Contains the status of the attempt as a text string
    - SUCCESS
    - USERSTOPPED
    - REMOTESTOPPED
    - ERROR
- `CPLAYBACKOFFSET` - Contains the offset in ms into the file where playback was at when it stopped. `-1` is end of file.
- `CPLAYBACKSTOPKEY` - If the playback is stopped by the user this variable contains the key that was pressed.

Syntax

```
ControlPlayback(filename,skipms,ff,rew,stop,pause,restart,options)
```

Arguments

- `filename`
- `skipms` - This is number of milliseconds to skip when rewinding or fast-forwarding.
- `ff` - Fast-forward when this DTMF digit is received. (defaults to `#`)
- `rew` - Rewind when this DTMF digit is received. (defaults to `*`)
- `stop` - Stop playback when this DTMF digit is received.
- `pause` - Pause playback when this DTMF digit is received.
- `restart` - Restart playback when this DTMF digit is received.
- `options`
    - `o`
        - `time` - Start at *time* ms from the beginning of the file.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_DAHDIAcceptR2Call

DAHDIAcceptR2Call()

Synopsis

Accept an R2 call if its not already accepted (you still need to answer it)

Description

This application will Accept the R2 call either with charge or no charge.

Syntax

```
DAHDIAcceptR2Call(charge)
```

Arguments

- `charge` - Yes or No.
  Whether you want to accept the call with charge or without charge.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_DAHDIBarge

DAHDIBarge()

Synopsis

Barge in (monitor) DAHDI channel.

Description

Barges in on a specified DAHDI *channel* or prompts if one is not specified. Returns `-1` when caller user hangs up and is independent of the state of the channel being monitored.

Syntax

```
DAHDIBarge(channel)
```

Arguments

- `channel` - Channel to barge.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_DAHDIRAS

DAHDIRAS()

Synopsis

Executes DAHDI ISDN RAS application.

Description

Executes a RAS server using pppd on the given channel. The channel must be a clear channel (i.e. PRI source) and a DAHDI channel to be able to use this function (No modem emulation is included).

Your pppd must be patched to be DAHDI aware.

Syntax

```
DAHDIRAS(args)
```

Arguments

- `args` - A list of parameters to pass to the pppd daemon, separated by `,` characters.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_DAHDIScan

DAHDIScan()

Synopsis

Scan DAHDI channels to monitor calls.

Description

Allows a call center manager to monitor DAHDI channels in a convenient way. Use # to select the next channel and use `*` to exit.

Syntax

```
DAHDIScan(group)
```

Arguments

- `group` - Limit scanning to a channel *group* by setting this option.

See Also

- Asterisk 12 ManagerEvent_ChanSpyStart
- Asterisk 12 ManagerEvent_ChanSpyStop

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_DAHDISendCallreroutingFacility

DAHDISendCallreroutingFacility()

Synopsis

Send an ISDN call rerouting/deflection facility message.

Description

This application will send an ISDN switch specific call rerouting/deflection facility message over the current channel. Supported switches depend upon the version of libpri in use.

Syntax

```
DAHDISendCallreroutingFacility(destination,original,reason)
```

Arguments

- `destination` - Destination number.
- `original` - Original called number.
- `reason` - Diversion reason, if not specified defaults to `unknown`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_DAHDISendKeypadFacility

DAHDISendKeypadFacility()

Synopsis

Send digits out of band over a PRI.

Description

This application will send the given string of digits in a Keypad Facility IE over the current channel.

Syntax

```
DAHDISendKeypadFacility(digits)
```

Arguments

- `digits`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_DateTime

DateTime()

Synopsis

Says a specified time in a custom format.

Description

Say the date and time in a specified format.

Syntax

```
DateTime(unixtime,timezone,format)
```

Arguments

- `unixtime` - time, in seconds since Jan 1, 1970. May be negative. Defaults to now.
- `timezone` - timezone, see `/usr/share/zoneinfo` for a list. Defaults to machine default.
- `format` - a format the time is to be said in. See `voicemail.conf`. Defaults to `ABdY "digits/at" IMp`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_DBdel

DBdel()

Synopsis

Delete a key from the asterisk database.

Description

This application will delete a *key* from the Asterisk database.

> **Note**
> This application has been DEPRECATED in favor of the DB_DELETE function.

Syntax

```
DBdel(family/key)
```

Arguments

- `family`
- `key`

See Also

- Asterisk 12 Function_DB_DELETE
- Asterisk 12 Application_DBdeltree
- Asterisk 12 Function_DB

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_DBdeltree

DBdeltree()

Synopsis

Delete a family or keytree from the asterisk database.

Description

This application will delete a *family* or *keytree* from the Asterisk database.

Syntax

```
DBdeltree(family/keytree)
```

Arguments

- `family`
- `keytree`

See Also

- Asterisk 12 Function_DB_DELETE
- Asterisk 12 Application_DBdel
- Asterisk 12 Function_DB

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_DeadAGI

DeadAGI()

Synopsis

Executes AGI on a hungup channel.

Description

Executes an Asterisk Gateway Interface compliant program on a channel. AGI allows Asterisk to launch external programs written in any language to control a telephony channel, play audio, read DTMF digits, etc. by communicating with the AGI protocol on **stdin** and **stdout**. As of `1.6.0`, this channel will not stop dialplan execution on hangup inside of this application. Dialplan execution will continue normally, even upon hangup until the AGI application signals a desire to stop (either by exiting or, in the case of a net script, by closing the connection). A locally executed AGI script will receive SIGHUP on hangup from the channel except when using

DeadAGI. A fast AGI server will correspondingly receive a HANGUP inline with the command dialog. Both of theses signals may be disabled by setting the `AGISIGHUP` channel variable to `no` before executing the AGI application. Alternatively, if you would like the AGI application to exit immediately after a channel hangup is detected, set the `AGIEXITONHANGUP` variable to `yes`.

Use the CLI command `agi show commands` to list available agi commands.

This application sets the following channel variable upon completion:

- `AGISTATUS` - The status of the attempt to the run the AGI script text string, one of:
    - SUCCESS
    - FAILURE
    - NOTFOUND
    - HANGUP

Syntax

```
DeadAGI(commandarg1arg2[...])
```

Arguments

- `command`
- `args`
    - `arg1`
    - `arg2`

See Also

- Asterisk 12 Application_AGI
- Asterisk 12 Application_EAGI

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_Dial

Dial()

Synopsis

Attempt to connect to another device or endpoint and bridge the call.

Description

This application will place calls to one or more specified channels. As soon as one of the requested channels answers, the originating channel will be answered, if it has not already been answered. These two channels will then be active in a bridged call. All other channels that were requested will then be hung up.

Unless there is a timeout specified, the Dial application will wait indefinitely until one of the called channels answers, the user hangs up, or if all of the called channels are busy or unavailable.

Dialplan executing will continue if no requested channels can be called, or if the timeout expires. This application will report normal termination if the originating channel hangs up, or if the call is bridged and either of the parties in the bridge ends the call.

If the `OUTBOUND_GROUP` variable is set, all peer channels created by this application will be put into that group (as in Set(GROUP()=...). If the `OUTBOUND_GROUP_ONCE` variable is set, all peer channels created by this application will be put into that group (as in Set(GROUP()=...). Unlike `OUTBOUND_GROUP`, however, the variable will be unset after use.

This application sets the following channel variables:

- `DIALEDTIME` - This is the time from dialing a channel until when it is disconnected.
- `ANSWEREDTIME` - This is the amount of time for actual call.
- `DIALSTATUS` - This is the status of the call
    - CHANUNAVAIL
    - CONGESTION
    - NOANSWER
    - BUSY
    - ANSWER
    - CANCEL
    - DONTCALL - For the Privacy and Screening Modes. Will be set if the called party chooses to send the calling party to the 'Go Away' script.
    - TORTURE - For the Privacy and Screening Modes. Will be set if the called party chooses to send the calling party to the 'torture' script.
    - INVALIDARGS

Syntax

```
Dial(Technology/Resource[&Technology2/Resource2[&...]][,timeout[,options,URL]])
```

Arguments

- `Technology/Resource`
    - `Technology/Resource` - Specification of the device(s) to dial. These must be in the format of `Technology/Resource`, where *Technology* represents a particular channel driver, and *Resource* represents a resource available to that particular channel driver.
    - `Technology2/Resource2` - Optional extra devices to dial in parallel
      If you need more then one enter them as Technology2/Resource2&Technology3/Resourse3&.....
- `timeout` - Specifies the number of seconds we attempt to dial the specified devices
  If not specified, this defaults to 136 years.
- `options`
    - A - Play an announcement to the called party, where *x* is the prompt to be played
        - `x` - The file to play to the called party
    - a - Immediately answer the calling channel when the called channel answers in all cases. Normally, the calling channel is answered when the called channel answers, but when options such as A() and M() are used, the calling channel is not answered until all actions on the called channel (such as playing an announcement) are completed. This option can be used to answer the calling channel before doing anything on the called channel. You will rarely need to use this option, the default behavior is adequate in most cases.
    - b - Before initiating an outgoing call, Gosub to the specified location using the newly created channel. The Gosub will be executed for each destination channel.
        - `context`
        - `exten`
        - `priority`

        - `arg1`
        - `argN`
- `B` - Before initiating the outgoing call(s), Gosub to the specified location using the current channel.
    - `context`
    - `exten`
    - `priority`
        - `arg1`
        - `argN`
- `C` - Reset the call detail record (CDR) for this call.
- `c` - If the Dial() application cancels this call, always set HANGUPCAUSE to 'answered elsewhere'
- `d` - Allow the calling user to dial a 1 digit extension while waiting for a call to be answered. Exit to that extension if it exists in the current context, or the context defined in the `EXITCONTEXT` variable, if it exists.
- `D` - Send the specified DTMF strings **after** the called party has answered, but before the call gets bridged. The *called* DTMF string is sent to the called party, and the *calling* DTMF string is sent to the calling party. Both arguments can be used alone. If *pro gress* is specified, its DTMF is sent to the called party immediately after receiving a PROGRESS message.
  See SendDTMF for valid digits.
    - `called`
    - `calling`
    - `progress`
- `e` - Execute the `h` extension for peer after the call ends
- `f` - If *x* is not provided, force the CallerID sent on a call-forward or deflection to the dialplan extension of this Dial() using a dialplan `hint`. For example, some PSTNs do not allow CallerID to be set to anything other than the numbers assigned to you. If *x* is provided, force the CallerID sent to *x*.
    - `x`
- `F` - When the caller hangs up, transfer the **called** party to the specified destination and **start** execution at that location.
    - `context`
    - `exten`
    - `priority`
- `F` - When the caller hangs up, transfer the **called** party to the next priority of the current extension and **start** execution at that location.
- `g` - Proceed with dialplan execution at the next priority in the current extension if the destination channel hangs up.
- `G` - If the call is answered, transfer the calling party to the specified *priority* and the called party to the specified *priority* plus one.
    - `context`
    - `exten`
    - `priority`
- `h` - Allow the called party to hang up by sending the DTMF sequence defined for disconnect in `features.conf`.
- `H` - Allow the calling party to hang up by sending the DTMF sequence defined for disconnect in `features.conf`.
- `i` - Asterisk will ignore any forwarding requests it may receive on this dial attempt.
- `I` - Asterisk will ignore any connected line update requests or any redirecting party update requests it may receive on this dial attempt.
- `k` - Allow the called party to enable parking of the call by sending the DTMF sequence defined for call parking in `features.conf`.
- `K` - Allow the calling party to enable parking of the call by sending the DTMF sequence defined for call parking in `features.conf`.
- `L` - Limit the call to *x* milliseconds. Play a warning when *y* milliseconds are left. Repeat the warning every *z* milliseconds until time expires.
  This option is affected by the following variables:
    - `LIMIT_PLAYAUDIO_CALLER` - If set, this variable causes Asterisk to play the prompts to the caller.
        - YES default: (true)
        - NO
    - `LIMIT_PLAYAUDIO_CALLEE` - If set, this variable causes Asterisk to play the prompts to the callee.
        - YES
        - NO default: (true)
    - `LIMIT_TIMEOUT_FILE` - If specified, *filename* specifies the sound prompt to play when the timeout is reached. If not set, the time remaining will be announced.
        - FILENAME
    - `LIMIT_CONNECT_FILE` - If specified, *filename* specifies the sound prompt to play when the call begins. If not set, the time remaining will be announced.
        - FILENAME

        - `LIMIT_WARNING_FILE` - If specified, *filename* specifies the sound prompt to play as a warning when time *x* is reached. If not set, the time remaining will be announced.
            - FILENAME
        - `x` - Maximum call time, in milliseconds
        - `y` - Warning time, in milliseconds
        - `z` - Repeat time, in milliseconds
- `m` - Provide hold music to the calling party until a requested channel answers. A specific music on hold *class* (as defined in `musiconhold.conf`) can be specified.
    - `class`
- `M` - Execute the specified *macro* for the **called** channel before connecting to the calling channel. Arguments can be specified to the Macro using `^` as a delimiter. The macro can set the variable `MACRO_RESULT` to specify the following actions after the macro is finished executing:
    - `MACRO_RESULT` - If set, this action will be taken after the macro finished executing.
        - ABORT - Hangup both legs of the call
        - CONGESTION - Behave as if line congestion was encountered
        - BUSY - Behave as if a busy signal was encountered
        - CONTINUE - Hangup the called party and allow the calling party to continue dialplan execution at the next priority
        - GOTO:[[<CONTEXT>^]<EXTEN>^]<PRIORITY> - Transfer the call to the specified destination.
    - `macro` - Name of the macro that should be executed.
    - `arg` - Macro arguments
- `n` - This option is a modifier for the call screening/privacy mode. (See the `p` and `P` options.) It specifies that no introductions are to be saved in the `priv-callerintros` directory.
    - `delete` - With *delete* either not specified or set to `0`, the recorded introduction will not be deleted if the caller hangs up while the remote party has not yet answered.
      With *delete* set to `1`, the introduction will always be deleted.
- `N` - This option is a modifier for the call screening/privacy mode. It specifies that if Caller*ID is present, do not screen the call.
- `o` - If *x* is not provided, specify that the CallerID that was present on the **calling** channel be stored as the CallerID on the **called** channel. This was the behavior of Asterisk 1.0 and earlier. If *x* is provided, specify the CallerID stored on the **called** channel. Note that o(${CALLERID(all)}) is similar to option o without the parameter.
    - `x`
- `O` - Enables **operator services** mode. This option only works when bridging a DAHDI channel to another DAHDI channel only. if specified on non-DAHDI interfaces, it will be ignored. When the destination answers (presumably an operator services station), the originator no longer has control of their line. They may hang up, but the switch will not release their line until the destination party (the operator) hangs up.
    - `mode` - With *mode* either not specified or set to `1`, the originator hanging up will cause the phone to ring back immediately.
      With *mode* set to `2`, when the operator flashes the trunk, it will ring their phone back.
- `p` - This option enables screening mode. This is basically Privacy mode without memory.
- `P` - Enable privacy mode. Use *x* as the family/key in the AstDB database if it is provided. The current extension is used if a database family/key is not specified.
    - `x`
- `r` - Default: Indicate ringing to the calling party, even if the called party isn't actually ringing. Pass no audio to the calling party until the called channel has answered.
    - `tone` - Indicate progress to calling party. Send audio 'tone' from indications.conf
- `S` - Hang up the call *x* seconds **after** the called party has answered the call.
    - `x`
- `s` - Force the outgoing callerid tag parameter to be set to the string *x*.
  Works with the f option.
    - `x`
- `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.
- `U` - Execute via Gosub the routine *x* for the **called** channel before connecting to the calling channel. Arguments can be specified to the Gosub using `^` as a delimiter. The Gosub routine can set the variable `GOSUB_RESULT` to specify the following actions after the Gosub returns.
    - `GOSUB_RESULT`
        - ABORT - Hangup both legs of the call.

        - CONGESTION - Behave as if line congestion was encountered.
        - BUSY - Behave as if a busy signal was encountered.
        - CONTINUE - Hangup the called party and allow the calling party to continue dialplan execution at the next priority.
        - GOTO:[[<CONTEXT>^]<EXTEN>^]<PRIORITY> - Transfer the call to the specified destination.
      - `x` - Name of the subroutine to execute via Gosub
      - `arg` - Arguments for the Gosub routine
  - `u` - Works with the f option.
      - `x` - Force the outgoing callerid presentation indicator parameter to be set to one of the values passed in *x*: `allowed_not_screened` `allowed_passed_screen` `allowed_failed_screen` `allowed` `prohib_not_screened` `prohib_passed_screen` `prohib_failed_screen` `prohib` `unavailable`
  - `w` - Allow the called party to enable recording of the call by sending the DTMF sequence defined for one-touch recording in `features.conf`.
  - `W` - Allow the calling party to enable recording of the call by sending the DTMF sequence defined for one-touch recording in `features.conf`.
  - `x` - Allow the called party to enable recording of the call by sending the DTMF sequence defined for one-touch automixmonitor in `features.conf`.
  - `X` - Allow the calling party to enable recording of the call by sending the DTMF sequence defined for one-touch automixmonitor in `features.conf`.
  - `z` - On a call forward, cancel any dial timeout which has been set for this call.
- `URL` - The optional URL will be sent to the called party if the channel driver supports it.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_Dictate

Dictate()

Synopsis

Virtual Dictation Machine.

Description

Start dictation machine using optional *base_dir* for files.

Syntax

```
Dictate(base_dir,filename)
```

Arguments

- `base_dir`
- `filename`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_Directory

Directory()

Synopsis

Provide directory of voicemail extensions.

Description

This application will present the calling channel with a directory of extensions from which they can search by name. The list of names and corresponding extensions is retrieved from the voicemail configuration file, `voicemail.conf`.

This application will immediately exit if one of the following DTMF digits are received and the extension to jump to exists:

0 - Jump to the 'o' extension, if it exists.

- - Jump to the 'a' extension, if it exists.

Syntax

```
Directory(vm-context[,dial-context[,options]])
```

Arguments

- `vm-context` - This is the context within voicemail.conf to use for the Directory. If not specified and `searchcontexts=no` in `voicemai l.conf`, then `default` will be assumed.
- `dial-context` - This is the dialplan context to use when looking for an extension that the user has selected, or when jumping to the `o` o r `a` extension. If not specified, the current context will be used.
- `options`
    - `e` - In addition to the name, also read the extension number to the caller before presenting dialing options.
    - `f` - Allow the caller to enter the first name of a user in the directory instead of using the last name. If specified, the optional number argument will be used for the number of characters the user should enter.
        - `n`
    - `l` - Allow the caller to enter the last name of a user in the directory. This is the default. If specified, the optional number argument will be used for the number of characters the user should enter.
        - `n`
    - `b` - Allow the caller to enter either the first or the last name of a user in the directory. If specified, the optional number argument will be used for the number of characters the user should enter.
        - `n`
    - `a` - Allow the caller to additionally enter an alias for a user in the directory. This option must be specified in addition to the `f`, `l`, or `b` option.
    - `m` - Instead of reading each name sequentially and asking for confirmation, create a menu of up to 8 names.
    - `n` - Read digits even if the channel is not answered.
    - `p` - Pause for n milliseconds after the digits are typed. This is helpful for people with cellphones, who are not holding the receiver to their ear while entering DTMF.
        - `n`

> **Note**
> Only one of the *f*, *l*, or *b* options may be specified. **If more than one is specified**, then Directory will act as if *b* was specified. The number of characters for the user to type defaults to 3.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_DISA

DISA()

Synopsis

Direct Inward System Access.

Description

The DISA, Direct Inward System Access, application allows someone from outside the telephone switch (PBX) to obtain an **internal** system dialtone and to place calls from it as if they were placing a call from within the switch. DISA plays a dialtone. The user enters their numeric passcode, followed by the pound sign `#`. If the passcode is correct, the user is then given system dialtone within *context* on which a call may be placed. If the user enters an invalid extension and extension `i` exists in the specified *context*, it will be used.

Be aware that using this may compromise the security of your PBX.

The arguments to this application (in `extensions.conf`) allow either specification of a single global *passcode* (that everyone uses), or individual passcodes contained in a file (*filename*).

The file that contains the passcodes (if used) allows a complete specification of all of the same arguments available on the command line, with the sole exception of the options. The file may contain blank lines, or comments starting with `#` or `;`.

Syntax

```
DISA(passcode|filename,context,cidmailbox[@context],options)
```

Arguments

- `passcode|filename` - If you need to present a DISA dialtone without entering a password, simply set *passcode* to `no-password`
  You may specified a *filename* instead of a *passcode*, this filename must contain individual passcodes
- `context` - Specifies the dialplan context in which the user-entered extension will be matched. If no context is specified, the DISA application defaults to the `disa` context. Presumably a normal system will have a special context set up for DISA use with some or a lot of restrictions.
- `cid` - Specifies a new (different) callerid to be used for this call.
- `mailbox` - Will cause a stutter-dialtone (indication **dialrecall**) to be used, if the specified mailbox contains any new messages.
  - `mailbox`
  - `context`
- `options`
  - `n` - The DISA application will not answer initially.
  - `p` - The extension entered will be considered complete when a `#` is entered.

See Also

- Asterisk 12 Application_Authenticate
- Asterisk 12 Application_VMAuthenticate

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_DumpChan

DumpChan()

Synopsis

Dump Info About The Calling Channel.

Description

Displays information on channel and listing of all channel variables. If *level* is specified, output is only displayed when the verbose level is currently set to that number or greater.

Syntax

```
DumpChan(level)
```

Arguments

- `level` - Minimun verbose level

See Also

- Asterisk 12 Application_NoOp
- Asterisk 12 Application_Verbose

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_EAGI

EAGI()

Synopsis

Executes an EAGI compliant application.

Description

Using 'EAGI' provides enhanced AGI, with incoming audio available out of band on file descriptor 3.

Executes an Asterisk Gateway Interface compliant program on a channel. AGI allows Asterisk to launch external programs written in any language to control a telephony channel, play audio, read DTMF digits, etc. by communicating with the AGI protocol on **stdin** and **stdout**. As of `1.6.0`, this channel will not stop dialplan execution on hangup inside of this application. Dialplan execution will continue normally, even upon hangup until the AGI application signals a desire to stop (either by exiting or, in the case of a net script, by closing the connection). A locally executed AGI script will receive SIGHUP on hangup from the channel except when using DeadAGI. A fast AGI server will correspondingly receive a HANGUP inline with the command dialog. Both of theses signals may be disabled by setting the `AGISIGHUP` channel variable to `no` before executing the AGI application. Alternatively, if you would like the AGI application to exit immediately after a channel hangup is detected, set the `AGIEXITONHANGUP` variable to `yes`.

Use the CLI command `agi show commands` to list available agi commands.

This application sets the following channel variable upon completion:

- `AGISTATUS` - The status of the attempt to the run the AGI script text string, one of:
    - SUCCESS
    - FAILURE
    - NOTFOUND
    - HANGUP

Syntax

```
EAGI(commandarg1arg2[...])
```

Arguments

- `command`
- `args`
    - `arg1`
    - `arg2`

See Also

- Asterisk 12 Application_AGI
- Asterisk 12 Application_DeadAGI

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_Echo

Echo()

Synopsis

Echo media, DTMF back to the calling party

Description

Echos back any media or DTMF frames read from the calling channel back to itself. This will not echo CONTROL, MODEM, or NULL frames. Note: If '#' detected application exits.

This application does not automatically answer and should be preceeded by an application such as Answer() or Progress().

Syntax

```
Echo()
```

Arguments

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_EndWhile

EndWhile()

Synopsis

End a while loop.

Description

Return to the previous called `While()`.

Syntax

```
EndWhile()
```

Arguments

See Also

- Asterisk 12 Application_While
- Asterisk 12 Application_ExitWhile
- Asterisk 12 Application_ContinueWhile

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_Exec

Exec()

Synopsis

Executes dialplan application.

Description

Allows an arbitrary application to be invoked even when not hard coded into the dialplan. If the underlying application terminates the dialplan, or if the application cannot be found, Exec will terminate the dialplan.

To invoke external applications, see the application System. If you would like to catch any error instead, see TryExec.

Syntax

```
Exec(arguments)
```

Arguments

- `appname` - Application name and arguments of the dialplan application to execute.
    - `arguments`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_ExecIf

ExecIf()

Synopsis

Executes dialplan application, conditionally.

Description

If *expr* is true, execute and return the result of *appiftrue(args)*.

If *expr* is true, but *appiftrue* is not found, then the application will return a non-zero value.

Syntax

```
ExecIf(expressionappiftrue[:appiffalse])
```

Arguments

- `expression`
- `execapp`
    - `appiftrue`
        - `args`
    - `appiffalse`
        - `args`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_ExecIfTime

ExecIfTime()

Synopsis

Conditional application execution based on the current time.

Description

This application will execute the specified dialplan application, with optional arguments, if the current time matches the given time specification.

Syntax

```
ExecIfTime(timesweekdaysmdaysmonths[timezone]appargs)
```

Arguments

- `day_condition`
    - `times`
    - `weekdays`
    - `mdays`
    - `months`
    - `timezone`
- `appname`
    - `appargs`

See Also

- Asterisk 12 Application_Exec
- Asterisk 12 Application_ExecIf
- Asterisk 12 Application_TryExec
- Asterisk 12 Application_GotoIfTime

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_ExitWhile

ExitWhile()

Synopsis

End a While loop.

Description

Exits a `While()` loop, whether or not the conditional has been satisfied.

Syntax

```
ExitWhile()
```

Arguments

See Also

- Asterisk 12 Application_While
- Asterisk 12 Application_EndWhile
- Asterisk 12 Application_ContinueWhile

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_ExtenSpy

ExtenSpy()

Synopsis

Listen to a channel, and optionally whisper into it.

Description

This application is used to listen to the audio from an Asterisk channel. This includes the audio coming in and out of the channel being spied on. Only channels created by outgoing calls for the specified extension will be selected for spying. If the optional context is not supplied, the current channel's context will be used.

While spying, the following actions may be performed:

- Dialing # cycles the volume level.
- Dialing * will stop spying and look for another channel to spy on.

> **Note**
> The *X* option supersedes the three features above in that if a valid single digit extension exists in the correct context ChanSpy will exit to it. This also disables choosing a channel based on `chanprefix` and a digit sequence.

Syntax

```
ExtenSpy(exten@context,options)
```

Arguments

- `exten`
  - `exten` - Specify extension.
  - `context` - Optionally specify a context, defaults to `default`.
- `options`
  - b - Only spy on channels involved in a bridged call.
  - B - Instead of whispering on a single channel barge in on both channels involved in the call.
  - c
    - `digit` - Specify a DTMF digit that can be used to spy on the next available channel.
  - d - Override the typical numeric DTMF functionality and instead use DTMF to switch between spy modes.
    - 4 - spy mode
    - 5 - whisper mode
    - 6 - barge mode
  - e - Enable **enforced** mode, so the spying channel can only monitor extensions whose name is in the *ext* : delimited list.
    - `ext`
  - E - Exit when the spied-on channel hangs up.
  - g
    - `grp` - Only spy on channels in which one or more of the groups listed in *grp* matches one or more groups from the `SPYGROUP` variable set on the channel to be spied upon.
  - n - Say the name of the person being spied on if that person has recorded his/her name. If a context is specified, then that voicemail context will be searched when retrieving the name, otherwise the `default` context be used when searching for the name (i.e. if SIP/1000 is the channel being spied on and no mailbox is specified, then `1000` will be used when searching for the

name).
    - `mailbox`
    - `context`
- `o` - Only listen to audio coming from this channel.
- `q` - Don't play a beep when beginning to spy on a channel, or speak the selected channel name.
- `r` - Record the session to the monitor spool directory. An optional base for the filename may be specified. The default is `chanspy`.
    - `basename`
- `s` - Skip the playback of the channel type (i.e. SIP, IAX, etc) when speaking the selected channel name.
- `S` - Stop when there are no more extensions left to spy on.
- `v` - Adjust the initial volume in the range from `-4` to `4`. A negative value refers to a quieter setting.
    - `value`
- `w` - Enable `whisper` mode, so the spying channel can talk to the spied-on channel.
- `W` - Enable `private whisper` mode, so the spying channel can talk to the spied-on channel but cannot listen to that channel.
- `x`
    - `digit` - Specify a DTMF digit that can be used to exit the application.
- `X` - Allow the user to exit ChanSpy to a valid single digit numeric extension in the current context or the context specified by the `SPY_EXIT_CONTEXT` channel variable. The name of the last channel that was spied on will be stored in the `SPY_CHANNEL` variable.

See Also

- Asterisk 12 Application_ChanSpy
- Asterisk 12 ManagerEvent_ChanSpyStart
- Asterisk 12 ManagerEvent_ChanSpyStop

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_ExternalIVR

ExternalIVR()

Synopsis

Interfaces with an external IVR application.

Description

Either forks a process to run given command or makes a socket to connect to given host and starts a generator on the channel. The generator's play list is controlled by the external application, which can add and clear entries via simple commands issued over its stdout. The external application will receive all DTMF events received on the channel, and notification if the channel is hung up. The received on the channel, and notification if the channel is hung up. The application will not be forcibly terminated when the channel is hung up. For more information see `doc/AST.pdf`.

Syntax

```
ExternalIVR(arg1arg2[...],options)
```

Arguments

- `command|ivr://host`
    - `arg1`

        - `arg2`
- `options`
    - `n` - Tells ExternalIVR() not to answer the channel.
    - `i` - Tells ExternalIVR() not to send a hangup and exit when the channel receives a hangup, instead it sends an `I` informative message meaning that the external application MUST hang up the call with an `H` command.
    - `d` - Tells ExternalIVR() to run on a channel that has been hung up and will not look for hangups. The external application must exit with an `E` command.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_Festival

Festival()

Synopsis

Say text to the user.

Description

Connect to Festival, send the argument, get back the waveform, play it to the user, allowing any given interrupt keys to immediately terminate and return the value, or `any` to allow any number back (useful in dialplan).

Syntax

```
Festival(text,intkeys)
```

Arguments

- `text`
- `intkeys`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_Flash

Flash()

Synopsis

Flashes a DAHDI Trunk.

Description

Performs a flash on a DAHDI trunk. This can be used to access features provided on an incoming analogue circuit such as conference and call waiting. Use with SendDTMF() to perform external transfers.

Syntax

```
Flash()
```

Arguments

See Also

- Asterisk 12 Application_SendDTMF

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_FollowMe

FollowMe()

Synopsis

Find-Me/Follow-Me application.

Description

This application performs Find-Me/Follow-Me functionality for the caller as defined in the profile matching the *followmeid* parameter in `followme.conf`. If the specified *followmeid* profile doesn't exist in `followme.conf`, execution will be returned to the dialplan and call execution will continue at the next priority.

Returns -1 on hangup.

Syntax

```
FollowMe(followmeid,options)
```

Arguments

- `followmeid`
- `options`
    - a - Record the caller's name so it can be announced to the callee on each step.
    - `B` - Before initiating the outgoing call(s), Gosub to the specified location using the current channel.
        - `context`
        - `exten`
        - `priority`
            - `arg1`
            - `argN`
    - b - Before initiating an outgoing call, Gosub to the specified location using the newly created channel. The Gosub will be executed for each destination channel.
        - `context`
        - `exten`
        - `priority`
            - `arg1`
            - `argN`
    - d - Disable the 'Please hold while we try to connect your call' announcement.
    - `I` - Asterisk will ignore any connected line update requests it may receive on this dial attempt.

- `l` - Disable local call optimization so that applications with audio hooks between the local bridge don't get dropped when the calls get joined directly.
- `N` - Don't answer the incoming call until we're ready to connect the caller or give up.
- `n` - Playback the unreachable status message if we've run out of steps or the callee has elected not to be reachable.
- `s` - Playback the incoming status message prior to starting the follow-me step(s)

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_ForkCDR

ForkCDR()

Synopsis

Forks the current Call Data Record for this channel.

Description

Causes the Call Data Record engine to fork a new CDR starting from the time the application is executed. The forked CDR will be linked to the end of the CDRs associated with the channel.

Syntax

```
ForkCDR(options)
```

Arguments

- `options`
    - `a` - If the channel is answered, set the answer time on the forked CDR to the current time. If this option is not used, the answer time on the forked CDR will be the answer time on the original CDR. If the channel is not answered, this option has no effect.
      Note that this option is implicitly assumed if the `r` option is used.
    - `e` - End (finalize) the original CDR.
    - `r` - Reset the start and answer times on the forked CDR. This will set the start and answer times (if the channel is answered) to be set to the current time.
      Note that this option implicitly assumes the a option.
    - `v` - Do not copy CDR variables and attributes from the original CDR to the forked CDR.

See Also

- Asterisk 12 Function_CDR
- Asterisk 12 Application_NoCDR
- Asterisk 12 Application_ResetCDR

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_GetCPEID

GetCPEID()

Synopsis

Get ADSI CPE ID.

Description

Obtains and displays ADSI CPE ID and other information in order to properly setup `dahdi.conf` for on-hook operations.

Syntax

```
GetCPEID()
```

Arguments

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_Gosub

Gosub()

Synopsis

Jump to label, saving return address.

Description

Jumps to the label specified, saving the return address.

Syntax

```
Gosub(context,extenarg1[...]argN)
```

Arguments

- `context`
- `exten`
- `priority`
    - `arg1`
    - `argN`

See Also

- Asterisk 12 Application_GosubIf
- Asterisk 12 Application_Macro
- Asterisk 12 Application_Goto
- Asterisk 12 Application_Return
- Asterisk 12 Application_StackPop

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_GosubIf

GosubIf()

Synopsis

Conditionally jump to label, saving return address.

Description

If the condition is true, then jump to labeliftrue. If false, jumps to labeliffalse, if specified. In either case, a jump saves the return point in the dialplan, to be returned to with a Return.

Syntax

```
GosubIf(conditionlabeliftrue:labeliffalse)
```

Arguments

- `condition`
- `destination`
    - `labeliftrue` - Continue at *labeliftrue* if the condition is true. Takes the form similar to Goto() of [[context,]extension,]priority.
        - `arg1`
        - `argN`
    - `labeliffalse` - Continue at *labeliffalse* if the condition is false. Takes the form similar to Goto() of [[context,]extension,]priority.
        - `arg1`
        - `argN`

See Also

- Asterisk 12 Application_Gosub
- Asterisk 12 Application_Return
- Asterisk 12 Application_MacroIf
- Asterisk 12 Function_IF
- Asterisk 12 Application_GotoIf
- Asterisk 12 Application_Goto

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_Goto

Goto()

Synopsis

Jump to a particular priority, extension, or context.

Description

This application will set the current context, extension, and priority in the channel structure. After it completes, the pbx engine will continue dialplan execution at the specified location. If no specific *extension*, or *extension* and *context*, are specified, then this application will just set the specified *priority* of the current extension.

At least a *priority* is required as an argument, or the goto will return a `-1`,and the channel and call will be terminated.

If the location that is put into the channel information is bogus, and asterisk cannot find that location in the dialplan, then the execution engine will try to find and execute the code in the `i` (in valid) extension in the current context. If that does not exist, it will try to execute the `h` extension. If neither the `h` nor `i` extensions have been defined, the channel is hung up, and the execution of instructions on the channel is terminated. What this means is that, for example, you specify a context that does not exist, then it will not be possible to find the `h` or `i` extensions, and the call will terminate!

Syntax

```
Goto(context,extensions,priority)
```

Arguments

- `context`
- `extensions`
- `priority`

See Also

- Asterisk 12 Application_GotoIf
- Asterisk 12 Application_GotoIfTime
- Asterisk 12 Application_Gosub
- Asterisk 12 Application_Macro

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_GotoIf

GotoIf()

Synopsis

Conditional goto.

Description

This application will set the current context, extension, and priority in the channel structure based on the evaluation of the given condition. After this application completes, the pbx engine will continue dialplan execution at the specified location in the dialplan. The labels are specified with the same syntax as used within the Goto application. If the label chosen by the condition is omitted, no jump is performed, and the execution passes to the next instruction. If the target location is bogus, and does not exist, the execution engine will try to find and execute the code in the `i` (invalid) extension in the current context. If that does not exist, it will try to execute the `h` ext ension. If neither the `h` nor `i` extensions have been defined, the channel is hung up, and the execution of instructions on the channel is terminated. Remember that this command can set the current context, and if the context specified does not exist, then it will not be able to find any 'h' or 'i' extensions there, and the channel and call will both be terminated!.

Syntax

```
GotoIf(conditionlabeliftrue:labeliffalse)
```

Arguments

- `condition`
- `destination`
    - `labeliftrue` - Continue at *labeliftrue* if the condition is true. Takes the form similar to Goto() of [[context,]extension,]priority.
    - `labeliffalse` - Continue at *labeliffalse* if the condition is false. Takes the form similar to Goto() of [[context,]extension,]priority.

See Also

- Asterisk 12 Application_Goto
- Asterisk 12 Application_GotoIfTime
- Asterisk 12 Application_GosubIf
- Asterisk 12 Application_MacroIf

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_GotoIfTime

GotoIfTime()

Synopsis

Conditional Goto based on the current time.

Description

This application will set the context, extension, and priority in the channel structure based on the evaluation of the given time specification. After this application completes, the pbx engine will continue dialplan execution at the specified location in the dialplan. If the current time is within the given time specification, the channel will continue at *labeliftrue*. Otherwise the channel will continue at *labeliffalse*. If the label chosen by the condition is omitted, no jump is performed, and execution passes to the next instruction. If the target jump location is bogus, the same actions would be taken as for `Goto`. Further information on the time specification can be found in examples illustrating how to do time-based context includes in the dialplan.

Syntax

```
GotoIfTime(timesweekdaysmdaysmonths[timezone]labeliftrue:labeliffalse)
```

Arguments

- `condition`
    - `times`
    - `weekdays`
    - `mdays`
    - `months`
    - `timezone`
- `destination`

- `labeliftrue` - Continue at *labeliftrue* if the condition is true. Takes the form similar to Goto() of [[context,]extension,]priority.
- `labeliffalse` - Continue at *labeliffalse* if the condition is false. Takes the form similar to Goto() of [[context,]extension,]priority.

See Also

- Asterisk 12 Application_GotoIf
- Asterisk 12 Application_Goto
- Asterisk 12 Function_IFTIME
- Asterisk 12 Function_TESTTIME

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_Hangup

Hangup()

Synopsis

Hang up the calling channel.

Description

This application will hang up the calling channel.

Syntax

```
Hangup(causecode)
```

Arguments

- `causecode` - If a *causecode* is given the channel's hangup cause will be set to the given value.

See Also

- Asterisk 12 Application_Answer
- Asterisk 12 Application_Busy
- Asterisk 12 Application_Congestion

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_HangupCauseClear

HangupCauseClear()

Synopsis

Clears hangup cause information from the channel that is available through HANGUPCAUSE.

Description

Clears all channel-specific hangup cause information from the channel. This is never done automatically (i.e. for new Dial()s).

Syntax

See Also

- Asterisk 12 Function_HANGUPCAUSE
- Asterisk 12 Function_HANGUPCAUSE_KEYS

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_IAX2Provision

IAX2Provision()

Synopsis

Provision a calling IAXy with a given template.

Description

Provisions the calling IAXy (assuming the calling entity is in fact an IAXy) with the given *template*. Returns `-1` on error or `0` on success.

Syntax

```
IAX2Provision(template)
```

Arguments

- `template` - If not specified, defaults to `default`.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_ICES

ICES()

Synopsis

Encode and stream using 'ices'.

Description

Streams to an icecast server using ices (available separately). A configuration file must be supplied for ices (see contrib/asterisk-ices.xml).

> **Note**
> ICES version 2 client and server required.

Syntax

```
ICES(config)
```

Arguments

- `config` - ICES configuration file.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_ImportVar

ImportVar()

Synopsis

Import a variable from a channel into a new variable.

Description

This application imports a *variable* from the specified *channel* (as opposed to the current one) and stores it as a variable (*newvar*) in the current channel (the channel that is calling this application). Variables created by this application have the same inheritance properties as those created with the `Set` application.

Syntax

```
ImportVar(newvarchannelnamevariable)
```

Arguments

- `newvar`
- `vardata`
  - `channelname`
  - `variable`

See Also

- Asterisk 12 Application_Set

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_Incomplete

Incomplete()

Synopsis

Returns AST_PBX_INCOMPLETE value.

Description

Signals the PBX routines that the previous matched extension is incomplete and that further input should be allowed before matching can be considered to be complete. Can be used within a pattern match when certain criteria warrants a longer match.

Syntax

```
Incomplete(n)
```

Arguments

- n - If specified, then Incomplete will not attempt to answer the channel first.

> **Note**
> Most channel types need to be in Answer state in order to receive DTMF.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_IVRDemo

IVRDemo()

Synopsis

IVR Demo Application.

Description

This is a skeleton application that shows you the basic structure to create your own asterisk applications and demonstrates the IVR demo.

Syntax

```
IVRDemo(filename)
```

Arguments

- `filename`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_JabberJoin_res_jabber

JabberJoin() - [res_jabber]

Synopsis

Join a chat room

Description

Allows Asterisk to join a chat room.

Syntax

```
JabberJoin(Jabber,RoomJID[,Nickname])
```

Arguments

- `Jabber` - Client or transport Asterisk uses to connect to Jabber.
- `RoomJID` - XMPP/Jabber JID (Name) of chat room.
- `Nickname` - The nickname Asterisk will use in the chat room.

> **Note**
> If a different nickname is supplied to an already joined room, the old nick will be changed to the new one.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_JabberJoin_res_xmpp

JabberJoin() - [res_xmpp]

Synopsis

Join a chat room

Description

Allows Asterisk to join a chat room.

Syntax

```
JabberJoin(Jabber,RoomJID[,Nickname])
```

Arguments

- `Jabber` - Client or transport Asterisk uses to connect to Jabber.
- `RoomJID` - XMPP/Jabber JID (Name) of chat room.
- `Nickname` - The nickname Asterisk will use in the chat room.

> **Note**
> If a different nickname is supplied to an already joined room, the old nick will be changed to the new one.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_JabberLeave_res_jabber

JabberLeave() - [res_jabber]

Synopsis

Leave a chat room

Description

Allows Asterisk to leave a chat room.

Syntax

```
JabberLeave(Jabber,RoomJID[,Nickname])
```

Arguments

- `Jabber` - Client or transport Asterisk uses to connect to Jabber.
- `RoomJID` - XMPP/Jabber JID (Name) of chat room.
- `Nickname` - The nickname Asterisk uses in the chat room.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_JabberLeave_res_xmpp

JabberLeave() - [res_xmpp]

Synopsis

Leave a chat room

Description

Allows Asterisk to leave a chat room.

Syntax

```
JabberLeave(Jabber,RoomJID[,Nickname])
```

Arguments

- `Jabber` - Client or transport Asterisk uses to connect to Jabber.
- `RoomJID` - XMPP/Jabber JID (Name) of chat room.
- `Nickname` - The nickname Asterisk uses in the chat room.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_JabberSend_res_jabber

JabberSend() - [res_jabber]

Synopsis

Sends an XMPP message to a buddy.

Description

Sends the content of *message* as text message from the given *account* to the buddy identified by *jid*

Example: JabberSend(asterisk,bob@domain.com,Hello world) sends "Hello world" to *bob@domain.com* as an XMPP message from the account *asterisk*, configured in jabber.conf.

Syntax

```
JabberSend(account,jid,message)
```

Arguments

- `account` - The local named account to listen on (specified in jabber.conf)
- `jid` - Jabber ID of the buddy to send the message to. It can be a bare JID (username@domain) or a full JID (username@domain/resource).
- `message` - The message to send.

See Also

- Asterisk 12 Function_JABBER_STATUS_res_jabber
- Asterisk 12 Function_JABBER_RECEIVE_res_jabber

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_JabberSend_res_xmpp

JabberSend() - [res_xmpp]

Synopsis

Sends an XMPP message to a buddy.

Description

Sends the content of *message* as text message from the given *account* to the buddy identified by *jid*

Example: JabberSend(asterisk,bob@domain.com,Hello world) sends "Hello world" to *bob@domain.com* as an XMPP message from the account *asterisk*, configured in xmpp.conf.

Syntax

```
JabberSend(account,jid,message)
```

Arguments

- `account` - The local named account to listen on (specified in xmpp.conf)
- `jid` - Jabber ID of the buddy to send the message to. It can be a bare JID (username@domain) or a full JID (username@domain/resource).
- `message` - The message to send.

See Also

- Asterisk 12 Function_JABBER_STATUS_res_xmpp
- Asterisk 12 Function_JABBER_RECEIVE_res_xmpp

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_JabberSendGroup_res_jabber

JabberSendGroup() - [res_jabber]

Synopsis

Send a Jabber Message to a specified chat room

Description

Allows user to send a message to a chat room via XMPP.

> **Note**
> To be able to send messages to a chat room, a user must have previously joined it. Use the *JabberJoin* function to do so.

Syntax

```
JabberSendGroup(Jabber,RoomJID,Message[,Nickname])
```

Arguments

- `Jabber` - Client or transport Asterisk uses to connect to Jabber.
- `RoomJID` - XMPP/Jabber JID (Name) of chat room.
- `Message` - Message to be sent to the chat room.
- `Nickname` - The nickname Asterisk uses in the chat room.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_JabberSendGroup_res_xmpp

JabberSendGroup() - [res_xmpp]

Synopsis

Send a Jabber Message to a specified chat room

Description

Allows user to send a message to a chat room via XMPP.

> **Note**
> To be able to send messages to a chat room, a user must have previously joined it. Use the *JabberJoin* function to do so.

Syntax

```
JabberSendGroup(Jabber,RoomJID,Message[,Nickname])
```

Arguments

- `Jabber` - Client or transport Asterisk uses to connect to Jabber.
- `RoomJID` - XMPP/Jabber JID (Name) of chat room.
- `Message` - Message to be sent to the chat room.
- `Nickname` - The nickname Asterisk uses in the chat room.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_JabberStatus_res_jabber

JabberStatus() - [res_jabber]

Synopsis

Retrieve the status of a jabber list member

Description

This application is deprecated. Please use the JABBER_STATUS() function instead.

Retrieves the numeric status associated with the specified buddy *JID*. The return value in the _Variable_will be one of the following.

- `1` - Online.
- `2` - Chatty.
- `3` - Away.
- `4` - Extended Away.
- `5` - Do Not Disturb.
- `6` - Offline.
- `7` - Not In Roster.

Syntax

```
JabberStatus(Jabber,JID,Variable)
```

Arguments

- `Jabber` - Client or transport Asterisk users to connect to Jabber.
- `JID` - XMPP/Jabber JID (Name) of recipient.
- `Variable` - Variable to store the status of requested user.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_JabberStatus_res_xmpp

JabberStatus() - [res_xmpp]

Synopsis

Retrieve the status of a jabber list member

Description

This application is deprecated. Please use the JABBER_STATUS() function instead.

Retrieves the numeric status associated with the specified buddy *JID*. The return value in the _Variable_will be one of the following.

- `1` - Online.
- `2` - Chatty.
- `3` - Away.
- `4` - Extended Away.
- `5` - Do Not Disturb.
- `6` - Offline.
- `7` - Not In Roster.

Syntax

```
JabberStatus(Jabber,JID,Variable)
```

Arguments

- `Jabber` - Client or transport Asterisk users to connect to Jabber.
- `JID` - XMPP/Jabber JID (Name) of recipient.
- `Variable` - Variable to store the status of requested user.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_JACK

JACK()

Synopsis

Jack Audio Connection Kit

Description

When executing this application, two jack ports will be created; one input and one output. Other applications can be hooked up to these ports to access audio coming from, or being send to the channel.

Syntax

```
JACK([options])
```

Arguments

- `options`
    - `s`
        - `name` - Connect to the specified jack server name
    - `i`
        - `name` - Connect the output port that gets created to the specified jack input port
    - `o`
        - `name` - Connect the input port that gets created to the specified jack output port
    - `c`
        - `name` - By default, Asterisk will use the channel name for the jack client name. Use this option to specify a custom client name.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_Log

Log()

Synopsis

Send arbitrary text to a selected log level.

Description

Sends an arbitrary text message to a selected log level.

Syntax

```
Log(level,message)
```

Arguments

- `level` - Level must be one of `ERROR`, `WARNING`, `NOTICE`, `DEBUG`, `VERBOSE` or `DTMF`.
- `message` - Output text message.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_Macro

Macro()

Synopsis

Macro Implementation.

Description

Executes a macro using the context macro- *name*, jumping to the `s` extension of that context and executing each step, then returning when the steps end.

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.

> **Warning**
> Because of the way Macro is implemented (it executes the priorities contained within it via sub-engine), and a fixed per-thread memory stack allowance, macros are limited to 7 levels of nesting (macro calling macro calling macro, etc.); It may be possible that stack-intensive applications in deeply nested macros could cause asterisk to crash earlier than this limit. It is advised that if you need to deeply nest macro calls, that you use the Gosub application (now allows arguments like a Macro) with explict Return() calls instead.

> **Warning**
> Use of the application `WaitExten` within a macro will not function as expected. Please use the `Read` application in order to read DTMF from a channel currently executing a macro.

Syntax

```
Macro(namearg1arg2[...])
```

Arguments

- `name` - The name of the macro
- `args`

- `arg1`
- `arg2`

See Also

- Asterisk 12 Application_MacroExit
- Asterisk 12 Application_Goto
- Asterisk 12 Application_Gosub

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_MacroExclusive

MacroExclusive()

Synopsis

Exclusive Macro Implementation.

Description

Executes macro defined in the context macro- *name*. Only one call at a time may run the macro. (we'll wait if another call is busy executing in the Macro)

Arguments and return values as in application Macro()

> **Warning**
> Use of the application `WaitExten` within a macro will not function as expected. Please use the `Read` application in order to read DTMF from a channel currently executing a macro.

Syntax

```
MacroExclusive(name,arg1,arg2[,...])
```

Arguments

- `name` - The name of the macro
- `arg1`
- `arg2`

See Also

- Asterisk 12 Application_Macro

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_MacroExit

MacroExit()

Synopsis

Exit from Macro.

Description

Causes the currently running macro to exit as if it had ended normally by running out of priorities to execute. If used outside a macro, will likely cause unexpected behavior.

Syntax

```
MacroExit()
```

Arguments

See Also

- Asterisk 12 Application_Macro

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_MacroIf

MacroIf()

Synopsis

Conditional Macro implementation.

Description

Executes macro defined in *macroiftrue* if *expr* is true (otherwise *macroiffalse* if provided)

Arguments and return values as in application Macro()

> **Warning**
> Use of the application `WaitExten` within a macro will not function as expected. Please use the `Read` application in order to read DTMF from a channel currently executing a macro.

Syntax

```
MacroIf(exprmacroiftrue:macroiffalse)
```

Arguments

- `expr`
- `destination`
    - `macroiftrue`
        - `macroiftrue`
        - `arg1`
    - `macroiffalse`

* macroiffalse
* arg1

See Also

* Asterisk 12 Application_GotoIf
* Asterisk 12 Application_GosubIf
* Asterisk 12 Function_IF

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_MailboxExists

MailboxExists()

Synopsis

Check to see if Voicemail mailbox exists.

Description

> **Note**
> DEPRECATED. Use VM_INFO(mailbox[@context],exists) instead.

Check to see if the specified *mailbox* exists. If no voicemail *context* is specified, the `default` context will be used.

This application will set the following channel variable upon completion:

* `VMBOXEXISTSSTATUS` - This will contain the status of the execution of the MailboxExists application. Possible values include:
    * SUCCESS
    * FAILED

Syntax

```
MailboxExists(mailbox@context,options)
```

Arguments

* `mailbox`
    * `mailbox`
    * `context`
* `options` - None options.

See Also

* Asterisk 12 Function_VM_INFO

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_MeetMe

MeetMe()

Synopsis

MeetMe conference bridge.

Description

Enters the user into a specified MeetMe conference. If the *confno* is omitted, the user will be prompted to enter one. User can exit the conference by hangup, or if the `p` option is specified, by pressing `#`.

> **Note**
> The DAHDI kernel modules and a functional DAHDI timing source (see dahdi_test) must be present for conferencing to operate properly. In addition, the chan_dahdi channel driver must be loaded for the `i` and `r` options to operate at all.

Syntax

```
MeetMe(confno,options,pin)
```

Arguments

- `confno` - The conference number
- `options`
    - `a` - Set admin mode.
    - `A` - Set marked mode.
    - `b` - Run AGI script specified in `MEETME_AGI_BACKGROUND` Default: `conf-background.agi`.
    - `c` - Announce user(s) count on joining a conference.
    - `C` - Continue in dialplan when kicked out of conference.
    - `d` - Dynamically add conference.
    - `D` - Dynamically add conference, prompting for a PIN.
    - `e` - Select an empty conference.
    - `E` - Select an empty pinless conference.
    - `F` - Pass DTMF through the conference.
    - `G` - Play an intro announcement in conference.
        - `x` - The file to playback
    - `i` - Announce user join/leave with review.
    - `I` - Announce user join/leave without review.
    - `k` - Close the conference if there's only one active participant left at exit.
    - `l` - Set listen only mode (Listen only, no talking).
    - `m` - Set initially muted.
    - `M` - Enable music on hold when the conference has a single caller. Optionally, specify a musiconhold class to use. If one is not provided, it will use the channel's currently set music class, or `default`.
        - `class`
    - `n` - Disable the denoiser. By default, if `func_speex` is loaded, Asterisk will apply a denoiser to channels in the MeetMe conference. However, channel drivers that present audio with a varying rate will experience degraded performance with a denoiser attached. This parameter allows a channel joining the conference to choose not to have a denoiser attached without having to unload `func_speex`.
    - `o` - Set talker optimization - treats talkers who aren't speaking as being muted, meaning (a) No encode is done on transmission and (b) Received audio that is not registered as talking is omitted causing no buildup in background noise.
    - `p` - Allow user to exit the conference by pressing `#` (default) or any of the defined keys. Dial plan execution will continue at the next priority following MeetMe. The key used is set to channel variable `MEETME_EXIT_KEY`.
        - `keys`
    - `P` - Always prompt for the pin even if it is specified.

- q - Quiet mode (don't play enter/leave sounds).
        - r - Record conference (records as MEETME_RECORDINGFILE using format MEETME_RECORDINGFORMAT. Default filename is meetme-conf-rec-${CONFNO}-${UNIQUEID} and the default format is wav.
        - s - Present menu (user or admin) when * is received (send to menu).
        - t - Set talk only mode. (Talk only, no listening).
        - T - Set talker detection (sent to manager interface and meetme list).
        - v - Announce when a user is joining or leaving the conference. Use the voicemail greeting as the announcement. If the i or I options are set, the application will fall back to them if no voicemail greeting can be found.
            - mailbox@context - The mailbox and voicemail context to play from. If no context provided, assumed context is default.
        - w - Wait until the marked user enters the conference.
            - secs
        - x - Leave the conference when the last marked user leaves.
        - X - Allow user to exit the conference by entering a valid single digit extension MEETME_EXIT_CONTEXT or the current context if that variable is not defined.
        - 1 - Do not play message when first person enters
        - S - Kick the user *x* seconds **after** he entered into the conference.
            - x
        - L - Limit the conference to *x* ms. Play a warning when *y* ms are left. Repeat the warning every *z* ms. The following special variables can be used with this option:
            - CONF_LIMIT_TIMEOUT_FILE - File to play when time is up.
            - CONF_LIMIT_WARNING_FILE - File to play as warning if *y* is defined. The default is to say the time remaining.
            - x
            - y
            - z
- pin

See Also

- Asterisk 12 Application_MeetMeCount
- Asterisk 12 Application_MeetMeAdmin
- Asterisk 12 Application_MeetMeChannelAdmin

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_MeetMeAdmin

MeetMeAdmin()

Synopsis

MeetMe conference administration.

Description

Run admin *command* for conference *confno*.

Will additionally set the variable MEETMEADMINSTATUS with one of the following values:

- MEETMEADMINSTATUS
    - NOPARSE - Invalid arguments.
    - NOTFOUND - User specified was not found.
    - FAILED - Another failure occurred.
    - OK - The operation was completed successfully.

Syntax

```
MeetMeAdmin(confno,command,user)
```

Arguments

- `confno`
- `command`
  - `e` - Eject last user that joined.
  - `E` - Extend conference end time, if scheduled.
  - `k` - Kick one user out of conference.
  - `K` - Kick all users out of conference.
  - `l` - Unlock conference.
  - `L` - Lock conference.
  - `m` - Unmute one user.
  - `M` - Mute one user.
  - `n` - Unmute all users in the conference.
  - `N` - Mute all non-admin users in the conference.
  - `r` - Reset one user's volume settings.
  - `R` - Reset all users volume settings.
  - `s` - Lower entire conference speaking volume.
  - `S` - Raise entire conference speaking volume.
  - `t` - Lower one user's talk volume.
  - `T` - Raise one user's talk volume.
  - `u` - Lower one user's listen volume.
  - `U` - Raise one user's listen volume.
  - `v` - Lower entire conference listening volume.
  - `V` - Raise entire conference listening volume.
- `user`

See Also

- Asterisk 12 Application_MeetMe

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_MeetMeChannelAdmin

MeetMeChannelAdmin()

Synopsis

MeetMe conference Administration (channel specific).

Description

Run admin *command* for a specific *channel* in any conference.

Syntax

```
MeetMeChannelAdmin(channel,command)
```

Arguments

- `channel`

- `command`
  - `k` - Kick the specified user out of the conference he is in.
  - `m` - Unmute the specified user.
  - `M` - Mute the specified user.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_MeetMeCount

MeetMeCount()

Synopsis

MeetMe participant count.

Description

Plays back the number of users in the specified MeetMe conference. If *var* is specified, playback will be skipped and the value will be returned in the variable. Upon application completion, MeetMeCount will hangup the channel, unless priority `n+1` exists, in which case priority progress will continue.

Syntax

```
MeetMeCount(confno,var)
```

Arguments

- `confno` - Conference number.
- `var`

See Also

- Asterisk 12 Application_MeetMe

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_MessageSend

MessageSend()

Synopsis

Send a text message.

Description

Send a text message. The body of the message that will be sent is what is currently set to `MESSAGE(body)`. The technology chosen for sending the message is determined based on a prefix to the `to` parameter.

This application sets the following channel variables:

- `MESSAGE_SEND_STATUS` - This is the message delivery status returned by this application.
    - INVALID_PROTOCOL - No handler for the technology part of the URI was found.
    - INVALID_URI - The protocol handler reported that the URI was not valid.
    - SUCCESS - Successfully passed on to the protocol handler, but delivery has not necessarily been guaranteed.
    - FAILURE - The protocol handler reported that it was unabled to deliver the message for some reason.

Syntax

```
MessageSend(to[,from])
```

Arguments

- `to` - A To URI for the message.

Technology: SIP

Specifying a prefix of `sip:` will send the message as a SIP MESSAGE request.

Technology: XMPP

Specifying a prefix of `xmpp:` will send the message as an XMPP chat message.

- `from` - A From URI for the message if needed for the message technology being used to send this message.

Technology: SIP

The `from` parameter can be a configured peer name or in the form of "display-name" <URI>.

Technology: XMPP

Specifying a prefix of `xmpp:` will specify the account defined in `xmpp.conf` to send the message from. Note that this field is required for XMPP messages.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_Milliwatt

Milliwatt()

Synopsis

Generate a Constant 1004Hz tone at 0dbm (mu-law).

Description

Previous versions of this application generated the tone at 1000Hz. If for some reason you would prefer that behavior, supply the o option to get the old behavior.

Syntax

```
Milliwatt(options)
```

Arguments

- `options`
  - o - Generate the tone at 1000Hz like previous version.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_MinivmAccMess

MinivmAccMess()

Synopsis

Record account specific messages.

Description

This application is part of the Mini-Voicemail system, configured in `minivm.conf`.

Use this application to record account specific audio/video messages for busy, unavailable and temporary messages.

Account specific directories will be created if they do not exist.

- `MVM_ACCMESS_STATUS` - This is the result of the attempt to record the specified greeting.
  FAILED is set if the file can't be created.
  - SUCCESS
  - FAILED

Syntax

```
MinivmAccMess(username@domain[,options])
```

Arguments

- `mailbox`
  - `username` - Voicemail username
  - `domain` - Voicemail domain
- `options`
  - `u` - Record the `unavailable` greeting.
  - `b` - Record the `busy` greeting.
  - `t` - Record the temporary greeting.
  - `n` - Account name.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_MinivmDelete

MinivmDelete()

Synopsis

Delete Mini-Voicemail voicemail messages.

Description

This application is part of the Mini-Voicemail system, configured in `minivm.conf`.

It deletes voicemail file set in MVM_FILENAME or given filename.

- `MVM_DELETE_STATUS` - This is the status of the delete operation.
  - SUCCESS
  - FAILED

Syntax

```
MinivmDelete(filename)
```

Arguments

- `filename` - File to delete

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_MinivmGreet

MinivmGreet()

Synopsis

Play Mini-Voicemail prompts.

Description

This application is part of the Mini-Voicemail system, configured in minivm.conf.

MinivmGreet() plays default prompts or user specific prompts for an account.

Busy and unavailable messages can be choosen, but will be overridden if a temporary message exists for the account.

- `MVM_GREET_STATUS` - This is the status of the greeting playback.
  - SUCCESS
  - USEREXIT
  - FAILED

Syntax

```
MinivmGreet(username@domain[,options])
```

Arguments

- `mailbox`
    - `username` - Voicemail username
    - `domain` - Voicemail domain
- `options`
    - `b` - Play the `busy` greeting to the calling party.
    - `s` - Skip the playback of instructions for leaving a message to the calling party.
    - `u` - Play the `unavailable` greeting.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_MinivmMWI

MinivmMWI()

Synopsis

Send Message Waiting Notification to subscriber(s) of mailbox.

Description

This application is part of the Mini-Voicemail system, configured in `minivm.conf`.

MinivmMWI is used to send message waiting indication to any devices whose channels have subscribed to the mailbox passed in the first parameter.

Syntax

```
MinivmMWI(username@domain,urgent,new,old)
```

Arguments

- `mailbox`
    - `username` - Voicemail username
    - `domain` - Voicemail domain
- `urgent` - Number of urgent messages in mailbox.
- `new` - Number of new messages in mailbox.
- `old` - Number of old messages in mailbox.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_MinivmNotify

MinivmNotify()

Synopsis

Notify voicemail owner about new messages.

Description

This application is part of the Mini-Voicemail system, configured in minivm.conf.

MiniVMnotify forwards messages about new voicemail to e-mail and pager. If there's no user account for that address, a temporary account will be used with default options (set in `minivm.conf`).

If the channel variable `MVM_COUNTER` is set, this will be used in the message file name and available in the template for the message.

If no template is given, the default email template will be used to send email and default pager template to send paging message (if the user account is configured with a paging address.

- `MVM_NOTIFY_STATUS` - This is the status of the notification attempt
    - SUCCESS
    - FAILED

Syntax

```
MinivmNotify(username@domain[,options])
```

Arguments

- `mailbox`
    - `username` - Voicemail username
    - `domain` - Voicemail domain
- `options`
    - `template` - E-mail template to use for voicemail notification

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_MinivmRecord

MinivmRecord()

Synopsis

Receive Mini-Voicemail and forward via e-mail.

Description

This application is part of the Mini-Voicemail system, configured in `minivm.conf`

MiniVM records audio file in configured format and forwards message to e-mail and pager.

If there's no user account for that address, a temporary account will be used with default options.

The recorded file name and path will be stored in `MVM_FILENAME` and the duration of the message will be stored in `MVM_DURATION`

> **Note**
> If the caller hangs up after the recording, the only way to send the message and clean up is to execute in the h extension. The application will exit if any of the following DTMF digits are received and the requested extension exist in the current context.

- `MVM_RECORD_STATUS` - This is the status of the record operation
  - SUCCESS
  - USEREXIT
  - FAILED

Syntax

```
MinivmRecord(username@domain[,options])
```

Arguments

- `mailbox`
  - `username` - Voicemail username
  - `domain` - Voicemail domain
- `options`
  - `0` - Jump to the o extension in the current dialplan context.
  - `*` - Jump to the a extension in the current dialplan context.
  - `g` - Use the specified amount of gain when recording the voicemail message. The units are whole-number decibels (dB).
    - `gain` - Amount of gain to use

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_MixMonitor

MixMonitor()

Synopsis

Record a call and mix the audio during the recording. Use of StopMixMonitor is required to guarantee the audio file is available for processing during dialplan execution.

Description

Records the audio on the current channel to the specified file.

This application does not automatically answer and should be preceeded by an application such as Answer or Progress().

> **Note**
> MixMonitor runs as an audiohook.

* `MIXMONITOR_FILENAME` - Will contain the filename used to record.

Syntax

```
MixMonitor(filename.extension,options,command)
```

Arguments

* `file`
    * `filename` - If *filename* is an absolute path, uses that path, otherwise creates the file in the configured monitoring directory from `asterisk.conf.`
    * `extension`
* `options`
    * `a` - Append to the file instead of overwriting it.
    * `b` - Only save audio to the file while the channel is bridged.
    * `v` - Adjust the **heard** volume by a factor of *x* (range `-4` to `4`)
        * `x`
    * `V` - Adjust the **spoken** volume by a factor of *x* (range `-4` to `4`)
        * `x`
    * `W` - Adjust both, **heard and spoken** volumes by a factor of *x* (range `-4` to `4`)
        * `x`
    * `r` - Use the specified file to record the **receive** audio feed. Like with the basic filename argument, if an absolute path isn't given, it will create the file in the configured monitoring directory.
        * `file`
    * `t` - Use the specified file to record the **transmit** audio feed. Like with the basic filename argument, if an absolute path isn't given, it will create the file in the configured monitoring directory.
        * `file`
    * `i` - Stores the MixMonitor's ID on this channel variable.
        * `chanvar`
    * `m` - Create a copy of the recording as a voicemail in the indicated **mailbox**(es) separated by commas eg. m(1111default,...). Folders can be optionally specified using the syntax: mailbox@context/folder
        * `mailbox`
* `command` - Will be executed when the recording is over.
  Any strings matching `^{X}` will be unescaped to `X`.
  All variables will be evaluated at the time MixMonitor is called.

See Also

* Asterisk 12 Application_Monitor
* Asterisk 12 Application_StopMixMonitor
* Asterisk 12 Application_PauseMonitor
* Asterisk 12 Application_UnpauseMonitor
* Asterisk 12 Function_AUDIOHOOK_INHERIT

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_Monitor

Monitor()

Synopsis

Monitor a channel.

Description

Used to start monitoring a channel. The channel's input and output voice packets are logged to files until the channel hangs up or monitoring is stopped by the StopMonitor application.

By default, files are stored to `/var/spool/asterisk/monitor/`. Returns `-1` if monitor files can't be opened or if the channel is already monitored, otherwise `0`.

Syntax

```
Monitor(file_format:urlbase,fname_base,options)
```

Arguments

- `file_format`
    - `file_format` - optional, if not set, defaults to `wav`
    - `urlbase`
- `fname_base` - if set, changes the filename used to the one specified.
- `options`
    - `m` - when the recording ends mix the two leg files into one and delete the two leg files. If the variable `MONITOR_EXEC` is set, the application referenced in it will be executed instead of soxmix/sox and the raw leg files will NOT be deleted automatically. soxmix/sox or `MONITOR_EXEC` is handed 3 arguments, the two leg files and a target mixed file name which is the same as the leg file names only without the in/out designator.
      If `MONITOR_EXEC_ARGS` is set, the contents will be passed on as additional arguments to `MONITOR_EXEC`. Both `MONITOR_EXEC` and the Mix flag can be set from the administrator interface.
    - `b` - Don't begin recording unless a call is bridged to another channel.
    - `i` - Skip recording of input stream (disables `m` option).
    - `o` - Skip recording of output stream (disables `m` option).

See Also

- Asterisk 12 Application_StopMonitor

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_Morsecode

Morsecode()

Synopsis

Plays morse code.

Description

Plays the Morse code equivalent of the passed string.

This application does not automatically answer and should be preceeded by an application such as Answer() or Progress().

This application uses the following variables:

- `MORSEDITLEN` - Use this value in (ms) for length of dit
- `MORSETONE` - The pitch of the tone in (Hz), default is 800

Syntax

```
Morsecode(string)
```

Arguments

- `string` - String to playback as morse code to channel

See Also

- Asterisk 12 Application_SayAlpha
- Asterisk 12 Application_SayPhonetic

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_MP3Player

MP3Player()

Synopsis

Play an MP3 file or M3U playlist file or stream.

Description

Executes mpg123 to play the given location, which typically would be a mp3 filename or m3u playlist filename or a URL. Please read http://en.wikipedia.org/wiki/M3U to see how M3U playlist file format is like, Example usage would be exten => 1234,1,MP3Player(/var/lib/asterisk/playlist.m3u) User can exit by pressing any key on the dialpad, or by hanging up.

This application does not automatically answer and should be preceeded by an application such as Answer() or Progress().

Syntax

```
MP3Player(Location)
```

Arguments

- `Location` - Location of the file to be played. (argument passed to mpg123)

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_MSet

MSet()

Synopsis

Set channel variable(s) or function value(s).

Description

This function can be used to set the value of channel variables or dialplan functions. When setting variables, if the variable name is prefixed with _, the variable will be inherited into channels created from the current channel If the variable name is prefixed with __, the variable will be inherited into channels created from the current channel and all children channels. MSet behaves in a similar fashion to the way Set worked in 1.2/1.4 and is thus prone to doing things that you may not expect. For example, it strips surrounding double-quotes from the right-hand side (value). If you need to put a separator character (comma or vert-bar), you will need to escape them by inserting a backslash before them. Avoid its use if possible.

Syntax

```
MSet(name1=value1name2=value2)
```

Arguments

- `set1`
  - `name1`
  - `value1`
- `set2`
  - `name2`
  - `value2`

See Also

- Asterisk 12 Application_Set

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_MusicOnHold

MusicOnHold()

Synopsis

Play Music On Hold indefinitely.

Description

Plays hold music specified by class. If omitted, the default music source for the channel will be used. Change the default class with Set(CHANNEL(musicclass)=...). If duration is given, hold music will be played specified number of seconds. If duration is ommited, music plays indefinitely. Returns `0` when done, `-1` on hangup.

This application does not automatically answer and should be preceeded by an application such as Answer() or Progress().

Syntax

```
MusicOnHold(class,duration)
```

Arguments

- `class`
- `duration`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_NBScat

NBScat()

Synopsis

Play an NBS local stream.

Description

Executes nbscat to listen to the local NBS stream. User can exit by pressing any key.

Syntax

```
NBScat()
```

Arguments

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_NoCDR

NoCDR()

Synopsis

Tell Asterisk to not maintain a CDR for this channel.

Description

This application will tell Asterisk not to maintain a CDR for the current channel. This does **NOT** mean that information is not tracked; rather, if the channel is hung up no CDRs will be created for that channel.

If a subsequent call to ResetCDR occurs, all non-finalized CDRs created for the channel will be enabled.

Note
This application is deprecated. Please use the CDR_PROP function to disable CDRs on a channel.

Syntax

```
NoCDR()
```

Arguments

See Also

- Asterisk 12 Application_ResetCDR
- Asterisk 12 Function_CDR_PROP

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_NoOp

NoOp()

Synopsis

Do Nothing (No Operation).

Description

This application does nothing. However, it is useful for debugging purposes.

This method can be used to see the evaluations of variables or functions without having any effect.

Syntax

```
NoOp(text)
```

Arguments

- `text` - Any text provided can be viewed at the Asterisk CLI.

See Also

- Asterisk 12 Application_Verbose
- Asterisk 12 Application_Log

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_ODBC_Commit

ODBC_Commit()

Synopsis

Commits a currently open database transaction.

Description

Commits the database transaction specified by *transaction ID* or the current active transaction, if not specified.

Syntax

```
ODBC_Commit([transaction ID])
```

Arguments

- `transaction ID`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_ODBC_Rollback

ODBC_Rollback()

Synopsis

Rollback a currently open database transaction.

Description

Rolls back the database transaction specified by *transaction ID* or the current active transaction, if not specified.

Syntax

```
ODBC_Rollback([transaction ID])
```

Arguments

- `transaction ID`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_ODBCFinish

ODBCFinish()

Synopsis

Clear the resultset of a sucessful multirow query.

Description

For queries which are marked as mode=multirow, this will clear any remaining rows of the specified resultset.

Syntax

```
ODBCFinish(result-id)
```

Arguments

- `result-id`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_Originate

Originate()

Synopsis

Originate a call.

Description

This application originates an outbound call and connects it to a specified extension or application. This application will block until the outgoing call fails or gets answered. At that point, this application will exit with the status variable set and dialplan processing will continue.

This application sets the following channel variable before exiting:

- `ORIGINATE_STATUS` - This indicates the result of the call origination.
  - FAILED
  - SUCCESS
  - BUSY
  - CONGESTION
  - HANGUP
  - RINGING
  - UNKNOWN - In practice, you should never see this value. Please report it to the issue tracker if you ever see it.

Syntax

```
Originate(tech_data,type,arg1[,arg2[,arg3[,timeout]]])
```

Arguments

- `tech_data` - Channel technology and data for creating the outbound channel. For example, SIP/1234.
- `type` - This should be `app` or `exten`, depending on whether the outbound channel should be connected to an application or extension.
- `arg1` - If the type is `app`, then this is the application name. If the type is `exten`, then this is the context that the channel will be sent to.

* `arg2` - If the type is `app`, then this is the data passed as arguments to the application. If the type is `exten`, then this is the extension that the channel will be sent to.
* `arg3` - If the type is `exten`, then this is the priority that the channel is sent to. If the type is `app`, then this parameter is ignored.
* `timeout` - Timeout in seconds. Default is 30 seconds.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_OSPAuth

OSPAuth()

Synopsis

OSP Authentication.

Description

Authenticate a call by OSP.

Input variables:

* `OSPINPEERIP` - The last hop IP address.
* `OSPINTOKEN` - The inbound OSP token.
Output variables:

* `OSPINHANDLE` - The inbound call OSP transaction handle.
* `OSPINTIMELIMIT` - The inbound call duration limit in seconds.
This application sets the following channel variable upon completion:

* `OSPAUTHSTATUS` - The status of OSPAuth attempt as a text string, one of
    * SUCCESS
    * FAILED
    * ERROR

Syntax

```
OSPAuth(provider,options)
```

Arguments

* `provider` - The name of the provider that authenticates the call.
* `options` - Reserverd.

See Also

* Asterisk 12 Application_OSPLookup
* Asterisk 12 Application_OSPNext
* Asterisk 12 Application_OSPFinish

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_OSPFinish

OSPFinish()

Synopsis

Report OSP entry.

Description

Report call state.

Input variables:

- `OSPINHANDLE` - The inbound call OSP transaction handle.
- `OSPOUTHANDLE` - The outbound call OSP transaction handle.
- `OSPAUTHSTATUS` - The OSPAuth status.
- `OSPLOOKUPSTATUS` - The OSPLookup status.
- `OSPNEXTSTATUS` - The OSPNext status.
- `OSPINAUDIOQOS` - The inbound call leg audio QoS string.
- `OSPOUTAUDIOQOS` - The outbound call leg audio QoS string.
  This application sets the following channel variable upon completion:

- `OSPFINISHSTATUS` - The status of the OSPFinish attempt as a text string, one of
  - SUCCESS
  - FAILED
  - ERROR

Syntax

```
OSPFinish(cause,options)
```

Arguments

- `cause` - Hangup cause.
- `options` - Reserved.

See Also

- Asterisk 12 Application_OSPAuth
- Asterisk 12 Application_OSPLookup
- Asterisk 12 Application_OSPNext

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_OSPLookup

OSPLookup()

Synopsis

Lookup destination by OSP.

Description

Looks up destination via OSP.

Input variables:

- `OSPINACTUALSRC` - The actual source device IP address in indirect mode.
- `OSPINPEERIP` - The last hop IP address.

- `OSPINTECH` - The inbound channel technology for the call.
- `OSPINHANDLE` - The inbound call OSP transaction handle.
- `OSPINTIMELIMIT` - The inbound call duration limit in seconds.
- `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.
- `OSPINTOHOST` - The inbound To header host part.
- `OSPINRPIDUSER` - The inbound Remote-Party-ID header user part.
- `OSPINPAIUSER` - The inbound P-Asserted-Identify header user part.
- `OSPINDIVUSER` - The inbound Diversion header user part.
- `OSPINDIVHOST` - The inbound Diversion header host part.
- `OSPINPCIUSER` - The inbound P-Charge-Info header user part.
- `OSPINCUSTOMINFON` - The inbound custom information, where `n` is the index beginning with `1` upto `8`.
  Output variables:

- `OSPOUTHANDLE` - The outbound call OSP transaction handle.
- `OSPOUTTECH` - The outbound channel technology for the call.
- `OSPDESTINATION` - The outbound destination IP address.
- `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.
- `OSPOUTTOKEN` - The outbound OSP token.
- `OSPDESTREMAILS` - The number of remained destinations.
- `OSPOUTTIMELIMIT` - The outbound call duration limit in seconds.
- `OSPOUTCALLIDTYPES` - The outbound Call-ID types.
- `OSPOUTCALLID` - The outbound Call-ID. Only for H.323.
- `OSPDIALSTR` - The outbound Dial command string.
  This application sets the following channel variable upon completion:

- `OSPLOOKUPSTATUS` - The status of OSPLookup attempt as a text string, one of
  - SUCCESS
  - FAILED
  - ERROR

Syntax

```
OSPLookup(exten,provider,options)
```

Arguments

- `exten` - The exten of the call.

- `provider` - The name of the provider that is used to route the call.
- `options`
  - `h` - generate H323 call id for the outbound call
  - `s` - generate SIP call id for the outbound call. Have not been implemented
  - `i` - generate IAX call id for the outbound call. Have not been implemented

See Also

- Asterisk 12 Application_OSPAuth
- Asterisk 12 Application_OSPNext
- Asterisk 12 Application_OSPFinish

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_OSPNext

OSPNext()

Synopsis

Lookup next destination by OSP.

Description

Looks up the next destination via OSP.

Input variables:

- `OSPINHANDLE` - The inbound call OSP transaction handle.
- `OSPOUTHANDLE` - The outbound call OSP transaction handle.
- `OSPINTIMELIMIT` - The inbound call duration limit in seconds.
- `OSPOUTCALLIDTYPES` - The outbound Call-ID types.
- `OSPDESTREMAILS` - The number of remained destinations.
  Output variables:

- `OSPOUTTECH` - The outbound channel technology.
- `OSPDESTINATION` - The destination IP address.
- `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.
- `OSPOUTTOKEN` - The outbound OSP token.
- `OSPDESTREMAILS` - The number of remained destinations.
- `OSPOUTTIMELIMIT` - The outbound call duration limit in seconds.
- `OSPOUTCALLID` - The outbound Call-ID. Only for H.323.
- `OSPDIALSTR` - The outbound Dial command string.
  This application sets the following channel variable upon completion:

- `OSPNEXTSTATUS` - The status of the OSPNext attempt as a text string, one of
  - SUCCESS
  - FAILED

- ERROR

Syntax

See Also

- Asterisk 12 Application_OSPAuth
- Asterisk 12 Application_OSPLookup
- Asterisk 12 Application_OSPFinish

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_Page

Page()

Synopsis

Page series of phones

Description

Places outbound calls to the given *technology/resource* and dumps them into a conference bridge as muted participants. The original caller is dumped into the conference as a speaker and the room is destroyed when the original caller leaves.

Syntax

```
Page(Technology/Resource&Technology2/Resource2[&...],options,timeout)
```

Arguments

- `Technology/Resource`
    - `Technology/Resource` - Specification of the device(s) to dial. These must be in the format of `Technology/Resource`, where *Technology* represents a particular channel driver, and *Resource* represents a resource available to that particular channel driver.
    - `Technology2/Resource2` - Optional extra devices to dial in parallel
      If you need more than one, enter them as Technology2/Resource2& Technology3/Resourse3&.....
- `options`
    - `d` - Full duplex audio
    - `i` - Ignore attempts to forward the call
    - `q` - Quiet, do not play beep to caller
    - `r` - Record the page into a file (`CONFBRIDGE(bridge,record_conference)`)
    - `s` - Only dial a channel if its device state says that it is `NOT_INUSE`
    - `A` - Play an announcement to all paged participants
        - `x` - The announcement to playback to all devices
    - `n` - Do not play announcement to caller (alters A behavior)
- `timeout` - Specify the length of time that the system will attempt to connect a call. After this duration, any page calls that have not been answered will be hung up by the system.

See Also

- Asterisk 12 Application_ConfBridge

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_Park

Park()

Synopsis

Park yourself.

Description

Used to park yourself (typically in combination with an attended transfer to know the parking space).

If you set the `PARKINGEXTEN` variable to a parking space extension in the parking lot, Park() will attempt to park the call on that extension. If the extension is already in use then execution will continue at the next priority.

Syntax

```
Park(parking_lot_name,options)
```

Arguments

- `parking_lot_name` - Specify in which parking lot to park a call.
  The parking lot used is selected in the following order:
  1) parking_lot_name option to this application
  2) `PARKINGLOT` variable
  3) `CHANNEL(parkinglot)` function (Possibly preset by the channel driver.)
  4) Default parking lot.
- `options` - A list of options for this parked call.
    - `r` - Send ringing instead of MOH to the parked call.
    - `R` - Randomize the selection of a parking space.
    - `s` - Silence announcement of the parking space number.
    - `c` - If the parking times out, go to this place in the dialplan instead of where the parking lot defines the call should go.
        - `context`
        - `extension`
        - `priority`
    - `t` - Use a timeout of `duration` seconds instead of the timeout specified by the parking lot.
        - `duration`

See Also

- Asterisk 12 Application_ParkedCall

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_ParkAndAnnounce

ParkAndAnnounce()

Synopsis

Park and Announce.

Description

Park a call into the parkinglot and announce the call to another channel.

The variable `PARKEDAT` will contain the parking extension into which the call was placed. Use with the Local channel to allow the dialplan to make use of this information.

Syntax

```
ParkAndAnnounce(parking_lot_name,optionsannounce:announce1[:...],dial)
```

Arguments

- `parking_lot_name` - Specify in which parking lot to park a call.
  The parking lot used is selected in the following order:
  1) parking_lot_name option to this application
  2) `PARKINGLOT` variable
  3) `CHANNEL(parkinglot)` function (Possibly preset by the channel driver.)
  4) Default parking lot.
- `options` - A list of options for this parked call.
  - `r` - Send ringing instead of MOH to the parked call.
  - `R` - Randomize the selection of a parking space.
  - `c` - If the parking times out, go to this place in the dialplan instead of where the parking lot defines the call should go.
    - `context`
    - `extension`
    - `priority`
  - `t` - Use a timeout of `duration` seconds instead of the timeout specified by the parking lot.
    - `duration`
- `announce_template`
  - `announce` - Colon-separated list of files to announce. The word `PARKED` will be replaced by a say_digits of the extension in which the call is parked.
  - `announce1`
- `dial` - The app_dial style resource to call to make the announcement. Console/dsp calls the console.

See Also

- Asterisk 12 Application_Park
- Asterisk 12 Application_ParkedCall

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_ParkedCall

ParkedCall()

Synopsis

Retrieve a parked call.

Description

Used to retrieve a parked call from a parking lot.

> **Note**

If a parking lot's parkext option is set, then Parking lots will automatically create and manage dialplan extensions in the parking lot context. If that is the case then you will not need to manage parking extensions yourself, just include the parking context of the parking lot.

Syntax

```
ParkedCall(parking_lot_name,parking_space)
```

Arguments

- `parking_lot_name` - Specify from which parking lot to retrieve a parked call.
  The parking lot used is selected in the following order:
  1) parking_lot_name option
  2) `PARKINGLOT` variable
  3) `CHANNEL(parkinglot)` function (Possibly preset by the channel driver.)
  4) Default parking lot.
- `parking_space` - Parking space to retrieve a parked call from. If not provided then the first available parked call in the parking lot will be retrieved.

See Also

- Asterisk 12 Application_Park

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_PauseMonitor

PauseMonitor()

Synopsis

Pause monitoring of a channel.

Description

Pauses monitoring of a channel until it is re-enabled by a call to UnpauseMonitor.

Syntax

```
PauseMonitor()
```

Arguments

See Also

- Asterisk 12 Application_UnpauseMonitor

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_PauseQueueMember

PauseQueueMember()

Synopsis

Pauses a queue member.

Description

Pauses (blocks calls for) a queue member. The given interface will be paused in the given queue. This prevents any calls from being sent from the queue to the interface until it is unpaused with UnpauseQueueMember or the manager interface. If no queuename is given, the interface is paused in every queue it is a member of. The application will fail if the interface is not found.

This application sets the following channel variable upon completion:

- `PQMSTATUS` - The status of the attempt to pause a queue member as a text string.
    - PAUSED
    - NOTFOUND
    Example: PauseQueueMember(,SIP/3000)

Syntax

```
PauseQueueMember(queuename,interface,options,reason)
```

Arguments

- `queuename`
- `interface`
- `options`
- `reason` - Is used to add extra information to the appropriate queue_log entries and manager events.

See Also

- Asterisk 12 Application_Queue
- Asterisk 12 Application_QueueLog
- Asterisk 12 Application_AddQueueMember
- Asterisk 12 Application_RemoveQueueMember
- Asterisk 12 Application_PauseQueueMember
- Asterisk 12 Application_UnpauseQueueMember
- Asterisk 12 Function_QUEUE_VARIABLES
- Asterisk 12 Function_QUEUE_MEMBER
- Asterisk 12 Function_QUEUE_MEMBER_COUNT
- Asterisk 12 Function_QUEUE_EXISTS
- Asterisk 12 Function_QUEUE_WAITING_COUNT
- Asterisk 12 Function_QUEUE_MEMBER_LIST
- Asterisk 12 Function_QUEUE_MEMBER_PENALTY

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_Pickup

Pickup()

Synopsis

Directed extension call pickup.

Description

This application can pickup a specified ringing channel. The channel to pickup can be specified in the following ways.

1) If no *extension* targets are specified, the application will pickup a channel matching the pickup group of the requesting channel.

2) If the *extension* is specified with a *context* of the special string `PICKUPMARK` (for example 10@PICKUPMARK), the application will pickup a channel which has defined the channel variable `PICKUPMARK` with the same value as *extension* (in this example, `10`).

3) If the *extension* is specified with or without a *context*, the channel with a matching *extension* a nd *context* will be picked up. If no *context* is specified, the current context will be used.

> **Note**
> The *extension* is typically set on matching channels by the dial application that created the channel. The *context* is set on matching channels by the channel driver for the device.

Syntax

```
Pickup(extension&extension2[&...])
```

Arguments

- `targets`
    - `extension` - Specification of the pickup target.
        - `extension`
        - `context`
    - `extension2` - Additional specifications of pickup targets.
        - `extension2`
        - `context2`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_PickupChan

PickupChan()

Synopsis

Pickup a ringing channel.

Description

This will pickup a specified *channel* if ringing.

Syntax

```
PickupChan(Technology/Resource[&Technology2/Resource2[&...]][,options])
```

Arguments

- `Technology/Resource`
  - `Technology/Resource`
  - `Technology2/Resource2`
- `options`
  - p - Channel name specified partial name. Used when find channel by callid.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_Playback

Playback()

Synopsis

Play a file.

Description

Plays back given filenames (do not put extension of wav/alaw etc). The playback command answer the channel if no options are specified. If the file is non-existant it will fail

This application sets the following channel variable upon completion:

- `PLAYBACKSTATUS` - The status of the playback attempt as a text string.
  - SUCCESS
  - FAILED

    See Also: Background (application) – for playing sound files that are interruptible

WaitExten (application) – wait for digits from caller, optionally play music on hold

Syntax

```
Playback(filename&filename2[&...],options)
```

Arguments

- `filenames`
  - `filename`
  - `filename2`
- `options` - Comma separated list of options
  - `skip` - Do not play if not answered
  - `noanswer` - Playback without answering, otherwise the channel will be answered before the sound is played.

See Also

- Asterisk 12 Application_Background

- Asterisk 12 Application_WaitExten
- Asterisk 12 Application_ControlPlayback
- Asterisk 12 AGICommand_stream file
- Asterisk 12 AGICommand_control stream file
- Asterisk 12 ManagerAction_ControlPlayback

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_PlayTones

PlayTones()

Synopsis

Play a tone list.

Description

Plays a tone list. Execution will continue with the next step in the dialplan immediately while the tones continue to play.

See the sample `indications.conf` for a description of the specification of a tonelist.

Syntax

```
PlayTones(arg)
```

Arguments

- `arg` - Arg is either the tone name defined in the `indications.conf` configuration file, or a directly specified list of frequencies and durations.

See Also

- Asterisk 12 Application_StopPlayTones

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_PrivacyManager

PrivacyManager()

Synopsis

Require phone number to be entered, if no CallerID sent

Description

If no Caller*ID is sent, PrivacyManager answers the channel and asks the caller to enter their phone number. The caller is given *maxretries* attempts to do so. The application does **nothing** if Caller*ID was received on the channel.

The application sets the following channel variable upon completion:

- `PRIVACYMGRSTATUS` - The status of the privacy manager's attempt to collect a phone number from the user.
    - SUCCESS
    - FAILED

Syntax

```
PrivacyManager(maxretries,minlength,options,context)
```

Arguments

- `maxretries` - Total tries caller is allowed to input a callerid. Defaults to `3`.
- `minlength` - Minimum allowable digits in the input callerid number. Defaults to `10`.
- `options` - Position reserved for options.
- `context` - Context to check the given callerid against patterns.

See Also

- Asterisk 12 Application_Zapateller

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_Proceeding

Proceeding()

Synopsis

Indicate proceeding.

Description

This application will request that a proceeding message be provided to the calling channel.

Syntax

```
Proceeding()
```

Arguments

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_Progress

Progress()

Synopsis

Indicate progress.

Description

This application will request that in-band progress information be provided to the calling channel.

Syntax

```
Progress()
```

Arguments

See Also

- Asterisk 12 Application_Busy
- Asterisk 12 Application_Congestion
- Asterisk 12 Application_Ringing
- Asterisk 12 Application_Playtones

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_Queue

Queue()

Synopsis

Queue a call for a call queue.

Description

In addition to transferring the call, a call may be parked and then picked up by another user.

This application will return to the dialplan if the queue does not exist, or any of the join options cause the caller to not enter the queue.

This application does not automatically answer and should be preceeded by an application such as Answer(), Progress(), or Ringing().

This application sets the following channel variable upon completion:

- `QUEUESTATUS` - The status of the call as a text string.
    - TIMEOUT
    - FULL
    - JOINEMPTY
    - LEAVEEMPTY
    - JOINUNAVAIL
    - LEAVEUNAVAIL
    - CONTINUE

Syntax

```
Queue(queuename,options,URL,announceoverride,timeout,AGI,macro,gosub,rule,
position)
```

Arguments

- `queuename`
- `options`
    - `C` - Mark all calls as "answered elsewhere" when cancelled.
    - `c` - Continue in the dialplan if the callee hangs up.
    - `d` - data-quality (modem) call (minimum delay).
    - `F` - When the caller hangs up, transfer the **called member** to the specified destination and **start** execution at that location.
        - `context`
        - `exten`
        - `priority`
    - `F` - When the caller hangs up, transfer the **called member** to the next priority of the current extension and **start** execution at that location.
    - `h` - Allow **callee** to hang up by pressing `*`.
    - `H` - Allow **caller** to hang up by pressing `*`.
    - `n` - No retries on the timeout; will exit this application and go to the next step.
    - `i` - Ignore call forward requests from queue members and do nothing when they are requested.
    - `I` - Asterisk will ignore any connected line update requests or any redirecting party update requests it may receive on this dial attempt.
    - `r` - Ring instead of playing MOH. Periodic Announcements are still made, if applicable.
    - `R` - Ring instead of playing MOH when a member channel is actually ringing.
    - `t` - Allow the **called** user to transfer the calling user.
    - `T` - Allow the **calling** user to transfer the call.
    - `w` - Allow the **called** user to write the conversation to disk via Monitor.
    - `W` - Allow the **calling** user to write the conversation to disk via Monitor.
    - `k` - Allow the **called** party to enable parking of the call by sending the DTMF sequence defined for call parking in `features.conf`.
    - `K` - Allow the **calling** party to enable parking of the call by sending the DTMF sequence defined for call parking in `features.conf`.
    - `x` - Allow the **called** user to write the conversation to disk via MixMonitor.
    - `X` - Allow the **calling** user to write the conversation to disk via MixMonitor.
- `URL` - URL will be sent to the called party if the channel supports it.
- `announceoverride`
- `timeout` - Will cause the queue to fail out after a specified number of seconds, checked between each `queues.conf` *timeout* and *retry* cycle.
- `AGI` - Will setup an AGI script to be executed on the calling party's channel once they are connected to a queue member.
- `macro` - Will run a macro on the calling party's channel once they are connected to a queue member.
- `gosub` - Will run a gosub on the calling party's channel once they are connected to a queue member.
- `rule` - Will cause the queue's defaultrule to be overridden by the rule specified.
- `position` - Attempt to enter the caller into the queue at the numerical position specified. `1` would attempt to enter the caller at the head of the queue, and `3` would attempt to place the caller third in the queue.

See Also

- Asterisk 12 Application_Queue
- Asterisk 12 Application_QueueLog
- Asterisk 12 Application_AddQueueMember
- Asterisk 12 Application_RemoveQueueMember
- Asterisk 12 Application_PauseQueueMember
- Asterisk 12 Application_UnpauseQueueMember
- Asterisk 12 Function_QUEUE_VARIABLES
- Asterisk 12 Function_QUEUE_MEMBER
- Asterisk 12 Function_QUEUE_MEMBER_COUNT
- Asterisk 12 Function_QUEUE_EXISTS
- Asterisk 12 Function_QUEUE_WAITING_COUNT
- Asterisk 12 Function_QUEUE_MEMBER_LIST
- Asterisk 12 Function_QUEUE_MEMBER_PENALTY

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_QueueLog

QueueLog()

Synopsis

Writes to the queue_log file.

Description

Allows you to write your own events into the queue log.

Example: QueueLog(101,${UNIQUEID},${AGENT},WENTONBREAK,600)

Syntax

```
QueueLog(queuename,uniqueid,agent,event,additionalinfo)
```

Arguments

- `queuename`
- `uniqueid`
- `agent`
- `event`
- `additionalinfo`

See Also

- Asterisk 12 Application_Queue
- Asterisk 12 Application_QueueLog
- Asterisk 12 Application_AddQueueMember
- Asterisk 12 Application_RemoveQueueMember
- Asterisk 12 Application_PauseQueueMember
- Asterisk 12 Application_UnpauseQueueMember
- Asterisk 12 Function_QUEUE_VARIABLES
- Asterisk 12 Function_QUEUE_MEMBER
- Asterisk 12 Function_QUEUE_MEMBER_COUNT
- Asterisk 12 Function_QUEUE_EXISTS
- Asterisk 12 Function_QUEUE_WAITING_COUNT
- Asterisk 12 Function_QUEUE_MEMBER_LIST
- Asterisk 12 Function_QUEUE_MEMBER_PENALTY

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_RaiseException

RaiseException()

Synopsis

Handle an exceptional condition.

Description

This application will jump to the e extension in the current context, setting the dialplan function EXCEPTION(). If the e extension does not exist, the call will hangup.

Syntax

```
RaiseException(reason)
```

Arguments

- `reason`

See Also

- Asterisk 12 Function_Exception

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_Read

Read()

Synopsis

Read a variable.

Description

Reads a #-terminated string of digits a certain number of times from the user in to the given *varia ble*.

This application sets the following channel variable upon completion:

- `READSTATUS` - This is the status of the read operation.
    - OK
    - ERROR
    - HANGUP
    - INTERRUPTED
    - SKIPPED
    - TIMEOUT

Syntax

```
Read(variablefilename&filename2[&...],maxdigits,options,attempts,timeout)
```

Arguments

- `variable` - The input digits will be stored in the given *variable* name.
- `filenames`
    - `filename` - file(s) to play before reading digits or tone with option i
    - `filename2`
- `maxdigits` - Maximum acceptable number of digits. Stops reading after *maxdigits* have been entered (without requiring the user to press the `#` key).
  Defaults to `0` - no limit - wait for the user press the `#` key. Any value below `0` means the same. Max accepted value is `255`.
- `options`
    - `s` - to return immediately if the line is not up.
    - `i` - to play filename as an indication tone from your `indications.conf`.
    - `n` - to read digits even if the line is not up.

- `attempts` - If greater than `1`, that many *attempts* will be made in the event no data is entered.
- `timeout` - The number of seconds to wait for a digit response. If greater than `0`, that value will override the default timeout. Can be floating point.

See Also

- Asterisk 12 Application_SendDTMF

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_ReadExten

ReadExten()

Synopsis

Read an extension into a variable.

Description

Reads a # terminated string of digits from the user into the given variable.

Will set READEXTENSTATUS on exit with one of the following statuses:

- `READEXTENSTATUS`
  - OK - A valid extension exists in ${variable}.
  - TIMEOUT - No extension was entered in the specified time. Also sets ${variable} to "t".
  - INVALID - An invalid extension, ${INVALID_EXTEN}, was entered. Also sets ${variable} to "i".
  - SKIP - Line was not up and the option 's' was specified.
  - ERROR - Invalid arguments were passed.

Syntax

```
ReadExten(variable,filename,context,option,timeout)
```

Arguments

- `variable`
- `filename` - File to play before reading digits or tone with option `i`
- `context` - Context in which to match extensions.
- `option`
  - `s` - Return immediately if the channel is not answered.
  - `i` - Play *filename* as an indication tone from your `indications.conf` or a directly specified list of frequencies and durations.
  - `n` - Read digits even if the channel is not answered.
- `timeout` - An integer number of seconds to wait for a digit response. If greater than `0`, that value will override the default timeout.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_ReadFile

ReadFile()

Synopsis

Read the contents of a text file into a channel variable.

Description

Read the contents of a text file into channel variable *varname*

Warning
ReadFile has been deprecated in favor of Set(varname=${FILE(file,0,length)})

Syntax

```
ReadFile(varnamefile[length])
```

Arguments

- `varname` - Result stored here.
- `fileparams`
  - `file` - The name of the file to read.
  - `length` - Maximum number of characters to capture.
    If not specified defaults to max.

See Also

- Asterisk 12 Application_System
- Asterisk 12 Application_Read

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_ReceiveFAX_app_fax

ReceiveFAX() - [app_fax]

Synopsis

Receive a Fax

Description

Receives a FAX from the channel into the given filename overwriting the file if it already exists.

File created will be in TIFF format.

This application sets the following channel variables:

- `LOCALSTATIONID` - To identify itself to the remote end
- `LOCALHEADERINFO` - To generate a header line on each page
- `FAXSTATUS`
  - SUCCESS
  - FAILED
- `FAXERROR` - Cause of failure
- `REMOTESTATIONID` - The CSID of the remote side
- `FAXPAGES` - Number of pages sent
- `FAXBITRATE` - Transmission rate
- `FAXRESOLUTION` - Resolution of sent fax

Syntax

```
ReceiveFAX(filename[,c])
```

Arguments

- `filename` - Filename of TIFF file save incoming fax
- `c` - Makes the application behave as the calling machine
  (Default behavior is as answering machine)

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_ReceiveFAX_res_fax

ReceiveFAX() - [res_fax]

Synopsis

Receive a FAX and save as a TIFF/F file.

Description

This application is provided by res_fax, which is a FAX technology agnostic module that utilizes FAX technology resource modules to complete a FAX transmission.

Session arguments can be set by the FAXOPT function and to check results of the ReceiveFax() application.

Syntax

```
ReceiveFAX(filename,options)
```

Arguments

- `filename`
- `options`
  - `d` - Enable FAX debugging.
  - `f` - Allow audio fallback FAX transfer on T.38 capable channels.
  - `F` - Force usage of audio mode on T.38 capable channels.
  - `s` - Send progress Manager events (overrides statusevents setting in res_fax.conf).

See Also

- Asterisk 12 Function_FAXOPT

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_Record

Record()

Synopsis

Record to a file.

Description

If filename contains %d, these characters will be replaced with a number incremented by one each time the file is recorded. Use `core show file formats` to see the available formats on your system User can press # to terminate the recording and continue to the next priority. If the user hangs up during a recording, all data will be lost and the application will terminate.

- `RECORDED_FILE` - Will be set to the final filename of the recording.
- `RECORD_STATUS` - This is the final status of the command
    - DTMF - A terminating DTMF was received ('#' or '*', depending upon option 't')
    - SILENCE - The maximum silence occurred in the recording.
    - SKIP - The line was not yet answered and the 's' option was specified.
    - TIMEOUT - The maximum length was reached.
    - HANGUP - The channel was hung up.
    - ERROR - An unrecoverable error occurred, which resulted in a WARNING to the logs.

Syntax

```
Record(filename.format,silence,maxduration,options)
```

Arguments

- `filename`
    - `filename`
    - `format` - Is the format of the file type to be recorded (wav, gsm, etc).
- `silence` - Is the number of seconds of silence to allow before returning.
- `maxduration` - Is the maximum recording duration in seconds. If missing or 0 there is no maximum.
- `options`
    - a - Append to existing recording rather than replacing.
    - n - Do not answer, but record anyway if line not yet answered.
    - q - quiet (do not play a beep tone).
    - s - skip recording if the line is not yet answered.
    - `t` - use alternate '*' terminator key (DTMF) instead of default '#'
    - x - Ignore all terminator keys (DTMF) and keep recording until hangup.
    - k - Keep recorded file upon hangup.
    - y - Terminate recording if **any** DTMF digit is received.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_RemoveQueueMember

RemoveQueueMember()

Synopsis

Dynamically removes queue members.

Description

If the interface is **NOT** in the queue it will return an error.

This application sets the following channel variable upon completion:

- `RQMSTATUS`
    - REMOVED
    - NOTINQUEUE
    - NOSUCHQUEUE
    - NOTDYNAMIC

      Example: RemoveQueueMember(techsupport,SIP/3000)

Syntax

```
RemoveQueueMember(queuename,interface)
```

Arguments

- `queuename`
- `interface`

See Also

- Asterisk 12 Application_Queue
- Asterisk 12 Application_QueueLog
- Asterisk 12 Application_AddQueueMember
- Asterisk 12 Application_RemoveQueueMember
- Asterisk 12 Application_PauseQueueMember
- Asterisk 12 Application_UnpauseQueueMember
- Asterisk 12 Function_QUEUE_VARIABLES
- Asterisk 12 Function_QUEUE_MEMBER
- Asterisk 12 Function_QUEUE_MEMBER_COUNT
- Asterisk 12 Function_QUEUE_EXISTS
- Asterisk 12 Function_QUEUE_WAITING_COUNT
- Asterisk 12 Function_QUEUE_MEMBER_LIST
- Asterisk 12 Function_QUEUE_MEMBER_PENALTY

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_ResetCDR

ResetCDR()

Synopsis

Resets the Call Data Record.

Description

This application causes the Call Data Record to be reset. Depending on the flags passed in, this can have several effects. With no options, a reset does the following:

1. The `start` time is set to the current time.

2. If the channel is answered, the `answer` time is set to the current time.

3. All variables are wiped from the CDR. Note that this step can be prevented with the v option.

On the other hand, if the e option is specified, the effects of the NoCDR application will be lifted. CDRs will be re-enabled for this channel.

> **Note**
> The e option is deprecated. Please use the CDR_PROP function instead.

Syntax

```
ResetCDR(options)
```

Arguments

- `options`
    - v - Save the CDR variables during the reset.
    - e - Enable the CDRs for this channel only (negate effects of NoCDR).

See Also

- Asterisk 12 Application_ForkCDR
- Asterisk 12 Application_NoCDR
- Asterisk 12 Function_CDR_PROP

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_RetryDial

RetryDial()

Synopsis

Place a call, retrying on failure allowing an optional exit extension.

Description

This application will attempt to place a call using the normal Dial application. If no channel can be reached, the *announce* file will be played. Then, it will wait *sleep* number of seconds before retrying the call. After *retries* number of attempts, the calling channel will continue at the next priority in the dialplan. If the *retries* setting is set to 0, this application will retry endlessly. While waiting to retry a call, a 1 digit extension may be dialed. If that extension exists in either the context defined in `EXITCONTEXT` or the current one, The call will jump to that extension immediately. The *dialargs* are specified in the same format that arguments are provided to the Dial application.

Syntax

```
RetryDial(announce,sleep,retries,dialargs)
```

Arguments

- `announce` - Filename of sound that will be played when no channel can be reached
- `sleep` - Number of seconds to wait after a dial attempt failed before a new attempt is made
- `retries` - Number of retries
  When this is reached flow will continue at the next priority in the dialplan
- `dialargs` - Same format as arguments provided to the Dial application

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_Return

Return()

Synopsis

Return from gosub routine.

Description

Jumps to the last label on the stack, removing it. The return *value*, if any, is saved in the channel variable `GOSUB_RETVAL`.

Syntax

```
Return(value)
```

Arguments

- `value` - Return value.

See Also

- Asterisk 12 Application_Gosub
- Asterisk 12 Application_StackPop

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_Ringing

Ringing()

Synopsis

Indicate ringing tone.

Description

This application will request that the channel indicate a ringing tone to the user.

Syntax

```
Ringing()
```

Arguments

See Also

- Asterisk 12 Application_Busy
- Asterisk 12 Application_Congestion
- Asterisk 12 Application_Progress
- Asterisk 12 Application_Playtones

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_SayAlpha

SayAlpha()

Synopsis

Say Alpha.

Description

This application will play the sounds that correspond to the letters of the given *string*.

Syntax

```
SayAlpha(string)
```

Arguments

- `string`

See Also

- Asterisk 12 Application_SayDigits
- Asterisk 12 Application_SayNumber
- Asterisk 12 Application_SayPhonetic
- Asterisk 12 Function_CHANNEL

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_SayAlphaCase

SayAlphaCase()

Synopsis

Say Alpha.

Description

This application will play the sounds that correspond to the letters of the given *string*. Optionally, a *casetype* may be specified. This will be used for case-insensitive or case-sensitive pronunciations.

Syntax

```
SayAlphaCase(casetype,string)
```

Arguments

- `casetype`
    - a - Case sensitive (all) pronunciation. (Ex: SayAlphaCase(a,aBc); - lowercase a uppercase b lowercase c).
    - `l` - Case sensitive (lower) pronunciation. (Ex: SayAlphaCase(l,aBc); - lowercase a b lowercase c).
    - n - Case insensitive pronunciation. Equivalent to SayAlpha. (Ex: SayAlphaCase(n,aBc) - a b c).
    - u - Case sensitive (upper) pronunciation. (Ex: SayAlphaCase(u,aBc); - a uppercase b c).
- `string`

See Also

- Asterisk 12 Application_SayDigits
- Asterisk 12 Application_SayNumber
- Asterisk 12 Application_SayPhonetic
- Asterisk 12 Application_SayAlpha
- Asterisk 12 Function_CHANNEL

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_SayCountedAdj

SayCountedAdj()

Synopsis

Say a adjective in declined form in order to count things

Description

Selects and plays the proper form of an adjective according to the gender and of the noun which it modifies and the number of objects named by the noun-verb combination which have been counted. Used when saying things such as "5 new messages". The various singular and plural forms of the adjective are selected by adding suffixes to *filename*.

If the channel language is English, then no suffix will ever be added (since, in English, adjectives are not declined). If the channel language is Russian or some other slavic language, then the suffix will the specified *gender* for nominative, and "x" for genative plural. (The genative singular is not used when counting things.) For example, SayCountedAdj(1,new,f) will play sound file "newa" (containing the word "novaya"), but SayCountedAdj(5,new,f) will play sound file "newx" (containing the word "novikh").

This application does not automatically answer and should be preceeded by an application such as Answer(), Progress(), or Proceeding().

Syntax

```
SayCountedAdj(number,filename,gender)
```

Arguments

- `number` - The number of things
- `filename` - File name stem for the adjective
- `gender` - The gender of the noun modified, one of 'm', 'f', 'n', or 'c'

See Also

- Asterisk 12 Application_SayCountedNoun
- Asterisk 12 Application_SayNumber

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_SayCountedNoun

SayCountedNoun()

Synopsis

Say a noun in declined form in order to count things

Description

Selects and plays the proper singular or plural form of a noun when saying things such as "five calls". English has simple rules for deciding when to say "call" and when to say "calls", but other languages have complicated rules which would be extremely difficult to implement in the Asterisk dialplan language.

The correct sound file is selected by examining the *number* and adding the appropriate suffix to *fi lename*. If the channel language is English, then the suffix will be either empty or "s". If the channel language is Russian or some other Slavic language, then the suffix will be empty for nominative, "x1" for genative singular, and "x2" for genative plural.

Note that combining *filename* with a suffix will not necessarily produce a correctly spelled plural form. For example, SayCountedNoun(2,man) will play the sound file "mans" rather than "men". This behavior is intentional. Since the file name is never seen by the end user, there is no need to implement complicated spelling rules. We simply record the word "men" in the sound file named "mans".

This application does not automatically answer and should be preceeded by an application such as Answer() or Progress.

Syntax

```
SayCountedNoun(number,filename)
```

Arguments

- `number` - The number of things
- `filename` - File name stem for the noun that is the the name of the things

See Also

- Asterisk 12 Application_SayCountedAdj
- Asterisk 12 Application_SayNumber

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_SayCountPL

SayCountPL()

Synopsis

Say Polish counting words.

Description

Polish grammar has some funny rules for counting words. for example 1 zloty, 2 zlote, 5 zlotych. This application will take the words for 1, 2-4 and 5 and decide based on grammar rules which one to use with the number you pass to it.

Example: SayCountPL(zloty,zlote,zlotych,122) will give: zlote

Syntax

```
SayCountPL(word1,word2,word5,number)
```

Arguments

- `word1`
- `word2`
- `word5`
- `number`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_SayDigits

SayDigits()

Synopsis

Say Digits.

Description

This application will play the sounds that correspond to the digits of the given number. This will use the language that is currently set for the channel.

Syntax

```
SayDigits(digits)
```

Arguments

- `digits`

See Also

- Asterisk 12 Application_SayAlpha
- Asterisk 12 Application_SayNumber
- Asterisk 12 Application_SayPhonetic
- Asterisk 12 Function_CHANNEL

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_SayNumber

SayNumber()

Synopsis

Say Number.

Description

This application will play the sounds that correspond to the given *digits*. Optionally, a *gender* may be specified. This will use the language that is currently set for the channel. See the CHANNEL() function for more information on setting the language for the channel.

Syntax

```
SayNumber(digits,gender)
```

Arguments

- `digits`
- `gender`

See Also

- Asterisk 12 Application_SayAlpha
- Asterisk 12 Application_SayDigits
- Asterisk 12 Application_SayPhonetic
- Asterisk 12 Function_CHANNEL

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_SayPhonetic

SayPhonetic()

Synopsis

Say Phonetic.

Description

This application will play the sounds from the phonetic alphabet that correspond to the letters in the given *string*.

Syntax

```
SayPhonetic(string)
```

Arguments

- `string`

See Also

- Asterisk 12 Application_SayAlpha
- Asterisk 12 Application_SayDigits
- Asterisk 12 Application_SayNumber

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_SayUnixTime

SayUnixTime()

Synopsis

Says a specified time in a custom format.

Description

Uses some of the sound files stored in `/var/lib/asterisk/sounds` to construct a phrase saying the specified date and/or time in the specified format.

Syntax

```
SayUnixTime([unixtime[,timezone[,format[,options]]]])
```

Arguments

- `unixtime` - time, in seconds since Jan 1, 1970. May be negative. Defaults to now.
- `timezone` - timezone, see `/usr/share/zoneinfo` for a list. Defaults to machine default.
- `format` - a format the time is to be said in. See `voicemail.conf`. Defaults to `ABdY "digits/at" IMp`
- `options`

- `j` - Allow the calling user to dial digits to jump to that extension.

See Also

- Asterisk 12 Function_STRFTIME
- Asterisk 12 Function_STRPTIME
- Asterisk 12 Function_IFTIME

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_SendDTMF

SendDTMF()

Synopsis

Sends arbitrary DTMF digits

Description

It will send all digits or terminate if it encounters an error.

Syntax

```
SendDTMF(digits[,timeout_ms[,duration_ms[,channel]]])
```

Arguments

- `digits` - List of digits 0-9,*#,a-d,A-D to send also w for a half second pause, W for a one second pause, and f or F for a flash-hook if the channel supports flash-hook.
- `timeout_ms` - Amount of time to wait in ms between tones. (defaults to .25s)
- `duration_ms` - Duration of each digit
- `channel` - Channel where digits will be played

See Also

- Asterisk 12 Application_Read

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_SendFAX_app_fax

SendFAX() - [app_fax]

Synopsis

Send a Fax

Description

Send a given TIFF file to the channel as a FAX.

This application sets the following channel variables:

- `LOCALSTATIONID` - To identify itself to the remote end
- `LOCALHEADERINFO` - To generate a header line on each page

* `FAXSTATUS`
    * SUCCESS
    * FAILED
* `FAXERROR` - Cause of failure
* `REMOTESTATIONID` - The CSID of the remote side
* `FAXPAGES` - Number of pages sent
* `FAXBITRATE` - Transmission rate
* `FAXRESOLUTION` - Resolution of sent fax

Syntax

```
SendFAX(filename[,a])
```

Arguments

* `filename` - Filename of TIFF file to fax
* a - Makes the application behave as the answering machine
  (Default behavior is as calling machine)

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_SendFAX_res_fax

SendFAX() - [res_fax]

Synopsis

Sends a specified TIFF/F file as a FAX.

Description

This application is provided by res_fax, which is a FAX technology agnostic module that utilizes FAX technology resource modules to complete a FAX transmission.

Session arguments can be set by the FAXOPT function and to check results of the SendFax() application.

Syntax

```
SendFAX(filename2[&...],options)
```

Arguments

* `filename`
    * `filename2` - TIFF file to send as a FAX.
* `options`
    * d - Enable FAX debugging.
    * `f` - Allow audio fallback FAX transfer on T.38 capable channels.
    * F - Force usage of audio mode on T.38 capable channels.
    * s - Send progress Manager events (overrides statusevents setting in res_fax.conf).
    * z - Initiate a T.38 reinvite on the channel if the remote end does not.

See Also

- Asterisk 12 Function_FAXOPT

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_SendImage

SendImage()

Synopsis

Sends an image file.

Description

Send an image file on a channel supporting it.

Result of transmission will be stored in `SENDIMAGESTATUS`

- `SENDIMAGESTATUS`
    - SUCCESS - Transmission succeeded.
    - FAILURE - Transmission failed.
    - UNSUPPORTED - Image transmission not supported by channel.

Syntax

```
SendImage(filename)
```

Arguments

- `filename` - Path of the filename (image) to send.

See Also

- Asterisk 12 Application_SendText
- Asterisk 12 Application_SendURL

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_SendText

SendText()

Synopsis

Send a Text Message.

Description

Sends *text* to current channel (callee).

Result of transmission will be stored in the `SENDTEXTSTATUS`

- `SENDTEXTSTATUS`
    - SUCCESS - Transmission succeeded.

- FAILURE - Transmission failed.
- UNSUPPORTED - Text transmission not supported by channel.

> **Note**
> At this moment, text is supposed to be 7 bit ASCII in most channels.

Syntax

```
SendText(text)
```

Arguments

- `text`

See Also

- Asterisk 12 Application_SendImage
- Asterisk 12 Application_SendURL

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_SendURL

SendURL()

Synopsis

Send a URL.

Description

Requests client go to *URL* (IAX2) or sends the URL to the client (other channels).

Result is returned in the `SENDURLSTATUS` channel variable:

- `SENDURLSTATUS`
    - SUCCESS - URL successfully sent to client.
    - FAILURE - Failed to send URL.
    - NOLOAD - Client failed to load URL (wait enabled).
    - UNSUPPORTED - Channel does not support URL transport.
      SendURL continues normally if the URL was sent correctly or if the channel does not support HTML transport. Otherwise, the channel is hung up.

Syntax

```
SendURL(URL,option)
```

Arguments

- `URL`
- `option`
    - `w` - Execution will wait for an acknowledgement that the URL has been loaded before continuing.

See Also

- Asterisk 12 Application_SendImage
- Asterisk 12 Application_SendText

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_Set

Set()

Synopsis

Set channel variable or function value.

Description

This function can be used to set the value of channel variables or dialplan functions. When setting variables, if the variable name is prefixed with _, the variable will be inherited into channels created from the current channel. If the variable name is prefixed with __, the variable will be inherited into channels created from the current channel and all children channels.

> **Note**
> If (and only if), in `/etc/asterisk/asterisk.conf`, you have a `[compat]` category, and you have `app_set = 1.4` under that, then the behavior of this app changes, and strips surrounding quotes from the right hand side as it did previously in 1.4. The advantages of not stripping out quoting, and not caring about the separator characters (comma and vertical bar) were sufficient to make these changes in 1.6. Confusion about how many backslashes would be needed to properly protect separators and quotes in various database access strings has been greatly reduced by these changes.

Syntax

```
Set(name=value)
```

Arguments

- `name`
- `value`

See Also

- Asterisk 12 Application_MSet
- Asterisk 12 Function_GLOBAL
- Asterisk 12 Function_SET
- Asterisk 12 Function_ENV

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_SetAMAFlags

SetAMAFlags()

Synopsis

Set the AMA Flags.

Description

This application will set the channel's AMA Flags for billing purposes.

> **Warning**
> This application is deprecated. Please use the CHANNEL function instead.

Syntax

```
SetAMAFlags(flag)
```

Arguments

- `flag`

See Also

- Asterisk 12 Function_CDR
- Asterisk 12 Function_CHANNEL

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_SetCallerPres

SetCallerPres()

Synopsis

Set CallerID Presentation.

Description

Set Caller*ID presentation on a call.

Syntax

```
SetCallerPres(presentation)
```

Arguments

- `presentation`
  - `allowed_not_screened` - Presentation Allowed, Not Screened.
  - `allowed_passed_screen` - Presentation Allowed, Passed Screen.
  - `allowed_failed_screen` - Presentation Allowed, Failed Screen.
  - `allowed` - Presentation Allowed, Network Number.
  - `prohib_not_screened` - Presentation Prohibited, Not Screened.
  - `prohib_passed_screen` - Presentation Prohibited, Passed Screen.

- `prohib_failed_screen` - Presentation Prohibited, Failed Screen.
- `prohib` - Presentation Prohibited, Network Number.
- `unavailable` - Number Unavailable.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_SetMusicOnHold

SetMusicOnHold()

Synopsis

Set default Music On Hold class.

Description

!!! DEPRECATED. USe Set(CHANNEL(musicclass)=...) instead !!!

Sets the default class for music on hold for a given channel. When music on hold is activated, this class will be used to select which music is played.

!!! DEPRECATED. USe Set(CHANNEL(musicclass)=...) instead !!!

Syntax

```
SetMusicOnHold(class)
```

Arguments

- `class`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_SIPAddHeader

SIPAddHeader()

Synopsis

Add a SIP header to the outbound call.

Description

Adds a header to a SIP call placed with DIAL.

Remember to use the X-header if you are adding non-standard SIP headers, like `X-Asterisk-Accountcode:`. Use this with care. Adding the wrong headers may jeopardize the SIP dialog.

Always returns `0`.

Syntax

```
SIPAddHeader(Header:Content)
```

Arguments

- `Header`
- `Content`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_SIPDtmfMode

SIPDtmfMode()

Synopsis

Change the dtmfmode for a SIP call.

Description

Changes the dtmfmode for a SIP call.

Syntax

```
SIPDtmfMode(mode)
```

Arguments

- `mode`
    - `inband`
    - `info`
    - `rfc2833`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_SIPRemoveHeader

SIPRemoveHeader()

Synopsis

Remove SIP headers previously added with SIPAddHeader

Description

SIPRemoveHeader() allows you to remove headers which were previously added with SIPAddHeader(). If no parameter is supplied, all previously added headers will be removed. If a parameter is supplied, only the matching headers will be removed.

For example you have added these 2 headers:

SIPAddHeader(P-Asserted-Identity: sip:foo@bar);

SIPAddHeader(P-Preferred-Identity: sip:bar@foo);

// remove all headers

SIPRemoveHeader();

// remove all P- headers

SIPRemoveHeader(P-);

// remove only the PAI header (note the : at the end)

SIPRemoveHeader(P-Asserted-Identity🙂;

Always returns `0`.

Syntax

```
SIPRemoveHeader([Header])
```

Arguments

- `Header`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_SIPSendCustomINFO

SIPSendCustomINFO()

Synopsis

Send a custom INFO frame on specified channels.

Description

SIPSendCustomINFO() allows you to send a custom INFO message on all active SIP channels or on channels with the specified User Agent. This application is only available if TEST_FRAMEWORK is defined.

Syntax

```
SIPSendCustomINFO(Data[,UserAgent])
```

Arguments

- `Data`
- `UserAgent`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_SkelGuessNumber

SkelGuessNumber()

Synopsis

An example number guessing game

Description

This simple number guessing application is a template to build other applications from. It shows you the basic structure to create your own Asterisk applications.

Syntax

```
SkelGuessNumber(level,options)
```

Arguments

- `level`
- `options`
  - c - The computer should cheat
  - n - How many games to play before hanging up

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_SLAStation

SLAStation()

Synopsis

Shared Line Appearance Station.

Description

This application should be executed by an SLA station. The argument depends on how the call was initiated. If the phone was just taken off hook, then the argument *station* should be just the station name. If the call was initiated by pressing a line key, then the station name should be preceded by an underscore and the trunk name associated with that line button.

For example: `station1_line1`

On exit, this application will set the variable `SLASTATION_STATUS` to one of the following values:

- `SLASTATION_STATUS`
    - FAILURE
    - CONGESTION
    - SUCCESS

Syntax

```
SLAStation(station)
```

Arguments

- `station` - Station name

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_SLATrunk

SLATrunk()

Synopsis

Shared Line Appearance Trunk.

Description

This application should be executed by an SLA trunk on an inbound call. The channel calling this application should correspond to the SLA trunk with the name *trunk* that is being passed as an argument.

On exit, this application will set the variable `SLATRUNK_STATUS` to one of the following values:

- `SLATRUNK_STATUS`
    - FAILURE
    - SUCCESS
    - UNANSWERED
    - RINGTIMEOUT

Syntax

```
SLATrunk(trunk,options)
```

Arguments

- `trunk` - Trunk name
- `options`
    - `M` - Play back the specified MOH *class* instead of ringing
        - `class`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_SMS

SMS()

Synopsis

Communicates with SMS service centres and SMS capable analogue phones.

Description

SMS handles exchange of SMS data with a call to/from SMS capable phone or SMS PSTN service center. Can send and/or receive SMS messages. Works to ETSI ES 201 912; compatible with BT SMS PSTN service in UK and Telecom Italia in Italy.

Typical usage is to use to handle calls from the SMS service centre CLI, or to set up a call using `outgoing` or manager interface to connect service centre to SMS().

"Messages are processed as per text file message queues. smsq (a separate software) is a command to generate message queues and send messages.

> **Note**
> The protocol has tight delay bounds. Please use short frames and disable/keep short the jitter buffer on the ATA to make sure that respones (ACK etc.) are received in time.

Syntax

```
SMS(name,options,addr,body)
```

Arguments

- `name` - The name of the queue used in `/var/spool/asterisk/sms`
- `options`
  - `a` - Answer, i.e. send initial FSK packet.
  - `s` - Act as service centre talking to a phone.
  - `t` - Use protocol 2 (default used is protocol 1).
  - `p` - Set the initial delay to N ms (default is `300`). addr and body are a deprecated format to send messages out.
  - `r` - Set the Status Report Request (SRR) bit.
  - `o` - The body should be coded as octets not 7-bit symbols.
- `addr`
- `body`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_SoftHangup

SoftHangup()

Synopsis

Hangs up the requested channel.

Description

Hangs up the requested channel. If there are no channels to hangup, the application will report it.

Syntax

```
SoftHangup(Technology/Resource,options)
```

Arguments

- `Technology/Resource`
- `options`
  - a - Hang up all channels on a specified device instead of a single resource

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_SpeechActivateGrammar

SpeechActivateGrammar()

Synopsis

Activate a grammar.

Description

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.

Hangs up the channel on failure. If this is not desired, use TryExec.

Syntax

```
SpeechActivateGrammar(grammar_name)
```

Arguments

- `grammar_name`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_SpeechBackground

SpeechBackground()

Synopsis

Play a sound file and wait for speech to be recognized.

Description

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. Once results are available the application returns and results (score and text) are available using dialplan functions.

The first text and score are ${SPEECH_TEXT(0)} AND ${SPEECH_SCORE(0)} while the second are ${SPEECH_TEXT(1)} and ${SPEECH_SCORE(1)}.

The first argument is the sound file and the second is the timeout integer in seconds.

Hangs up the channel on failure. If this is not desired, use TryExec.

Syntax

```
SpeechBackground(sound_file,timeout,options)
```

Arguments

- `sound_file`
- `timeout` - Timeout integer in seconds. Note the timeout will only start once the sound file has stopped playing.
- `options`
    - n - Don't answer the channel if it has not already been answered.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_SpeechCreate

SpeechCreate()

Synopsis

Create a Speech Structure.

Description

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.

Sets the ERROR channel variable to 1 if the engine cannot be used.

Syntax

```
SpeechCreate(engine_name)
```

Arguments

- `engine_name`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_SpeechDeactivateGrammar

SpeechDeactivateGrammar()

Synopsis

Deactivate a grammar.

Description

This deactivates the specified grammar so that it is no longer recognized.

Hangs up the channel on failure. If this is not desired, use TryExec.

Syntax

```
SpeechDeactivateGrammar(grammar_name)
```

Arguments

- `grammar_name` - The grammar name to deactivate

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_SpeechDestroy

SpeechDestroy()

Synopsis

End speech recognition.

Description

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.

Hangs up the channel on failure. If this is not desired, use TryExec.

Syntax

```
SpeechDestroy()
```

Arguments

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_SpeechLoadGrammar

SpeechLoadGrammar()

Synopsis

Load a grammar.

Description

Load a grammar only on the channel, not globally.

Hangs up the channel on failure. If this is not desired, use TryExec.

Syntax

```
SpeechLoadGrammar(grammar_name,path)
```

Arguments

- `grammar_name`
- `path`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_SpeechProcessingSound

SpeechProcessingSound()

Synopsis

Change background processing sound.

Description

This changes the processing sound that SpeechBackground plays back when the speech recognition engine is processing and working to get results.

Hangs up the channel on failure. If this is not desired, use TryExec.

Syntax

```
SpeechProcessingSound(sound_file)
```

Arguments

- `sound_file`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_SpeechStart

SpeechStart()

Synopsis

Start recognizing voice in the audio stream.

Description

Tell the speech recognition engine that it should start trying to get results from audio being fed to it.

Hangs up the channel on failure. If this is not desired, use TryExec.

Syntax

```
SpeechStart()
```

Arguments

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_SpeechUnloadGrammar

SpeechUnloadGrammar()

Synopsis

Unload a grammar.

Description

Unload a grammar.

Hangs up the channel on failure. If this is not desired, use TryExec.

Syntax

```
SpeechUnloadGrammar(grammar_name)
```

Arguments

- `grammar_name`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_StackPop

StackPop()

Synopsis

Remove one address from gosub stack.

Description

Removes last label on the stack, discarding it.

Syntax

```
StackPop()
```

Arguments

See Also

- Asterisk 12 Application_Return
- Asterisk 12 Application_Gosub

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_StartMusicOnHold

StartMusicOnHold()

Synopsis

Play Music On Hold.

Description

Starts playing music on hold, uses default music class for channel. Starts playing music specified by class. If omitted, the default music source for the channel will be used. Always returns `0`.

Syntax

```
StartMusicOnHold(class)
```

Arguments

- `class`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_Stasis

Stasis()

Synopsis

Invoke an external Stasis application.

Description

Invoke a Stasis application.

Syntax

```
Stasis(app_name,args)
```

Arguments

- `app_name` - Name of the application to invoke.
- `args` - Optional comma-delimited arguments for the application invocation.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_StopMixMonitor

StopMixMonitor()

Synopsis

Stop recording a call through MixMonitor, and free the recording's file handle.

Description

Stops the audio recording that was started with a call to `MixMonitor()` on the current channel.

Syntax

```
StopMixMonitor([MixMonitorID])
```

Arguments

- `MixMonitorID` - If a valid ID is provided, then this command will stop only that specific MixMonitor.

See Also

- Asterisk 12 Application_MixMonitor

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_StopMonitor

StopMonitor()

Synopsis

Stop monitoring a channel.

Description

Stops monitoring a channel. Has no effect if the channel is not monitored.

Syntax

```
StopMonitor()
```

Arguments

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_StopMusicOnHold

StopMusicOnHold()

Synopsis

Stop playing Music On Hold.

Description

Stops playing music on hold.

Syntax

```
StopMusicOnHold()
```

Arguments

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_StopPlayTones

StopPlayTones()

Synopsis

Stop playing a tone list.

Description

Stop playing a tone list, initiated by PlayTones().

Syntax

```
StopPlayTones()
```

Arguments

See Also

- Asterisk 12 Application_PlayTones

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_System

System()

Synopsis

Execute a system command.

Description

Executes a command by using system(). If the command fails, the console should report a fallthrough.

Result of execution is returned in the `SYSTEMSTATUS` channel variable:

- SYSTEMSTATUS
    - FAILURE - Could not execute the specified command.
    - SUCCESS - Specified command successfully executed.

Syntax

```
System(command)
```

Arguments

- `command` - Command to execute

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_TestClient

TestClient()

Synopsis

Execute Interface Test Client.

Description

Executes test client with given *testid*. Results stored in `/var/log/asterisk/testreports/<testid>-client.txt`

Syntax

```
TestClient(testid)
```

Arguments

- `testid` - An ID to identify this test.

See Also

- Asterisk 12 Application_TestServer

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_TestServer

TestServer()

Synopsis

Execute Interface Test Server.

Description

Perform test server function and write call report. Results stored in `/var/log/asterisk/testreports/<testid>-server.txt`

Syntax

```
TestServer()
```

Arguments

See Also

- Asterisk 12 Application_TestClient

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_Transfer

Transfer()

Synopsis

Transfer caller to remote extension.

Description

Requests the remote caller be transferred to a given destination. If TECH (SIP, IAX2, LOCAL etc) is used, only an incoming call with the same channel technology will be transfered. Note that for SIP, if you transfer before call is setup, a 302 redirect SIP message will be returned to the caller.

The result of the application will be reported in the `TRANSFERSTATUS` channel variable:

- `TRANSFERSTATUS`
  - SUCCESS - Transfer succeeded.
  - FAILURE - Transfer failed.
  - UNSUPPORTED - Transfer unsupported by channel driver.

Syntax

```
Transfer(Tech/destination)
```

Arguments

- `dest`
  - `Tech/`
  - `destination`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_TryExec

TryExec()

Synopsis

Executes dialplan application, always returning.

Description

Allows an arbitrary application to be invoked even when not hard coded into the dialplan. To invoke external applications see the application System. Always returns to the dialplan. The channel variable TRYSTATUS will be set to one of:

- `TRYSTATUS`
    - SUCCESS - If the application returned zero.
    - FAILED - If the application returned non-zero.
    - NOAPP - If the application was not found or was not specified.

Syntax

```
TryExec(arguments)
```

Arguments

- `appname`
    - `arguments`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_TrySystem

TrySystem()

Synopsis

Try executing a system command.

Description

Executes a command by using system().

Result of execution is returned in the `SYSTEMSTATUS` channel variable:

- `SYSTEMSTATUS`
    - FAILURE - Could not execute the specified command.
    - SUCCESS - Specified command successfully executed.
    - APPERROR - Specified command successfully executed, but returned error code.

Syntax

```
TrySystem(command)
```

Arguments

- `command` - Command to execute

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_UnpauseMonitor

UnpauseMonitor()

Synopsis

Unpause monitoring of a channel.

Description

Unpauses monitoring of a channel on which monitoring had previously been paused with PauseMonitor.

Syntax

```
UnpauseMonitor()
```

Arguments

See Also

- Asterisk 12 Application_PauseMonitor

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_UnpauseQueueMember

UnpauseQueueMember()

Synopsis

Unpauses a queue member.

Description

Unpauses (resumes calls to) a queue member. This is the counterpart to `PauseQueueMember()` and operates exactly the same way, except it unpauses instead of pausing the given interface.

This application sets the following channel variable upon completion:

- `UPQMSTATUS` - The status of the attempt to unpause a queue member as a text string.
    - UNPAUSED
    - NOTFOUND
      Example: UnpauseQueueMember(,SIP/3000)

Syntax

```
UnpauseQueueMember(queuename,interface,options,reason)
```

Arguments

- `queuename`
- `interface`
- `options`

* `reason` - Is used to add extra information to the appropriate queue_log entries and manager events.

See Also

* Asterisk 12 Application_Queue
* Asterisk 12 Application_QueueLog
* Asterisk 12 Application_AddQueueMember
* Asterisk 12 Application_RemoveQueueMember
* Asterisk 12 Application_PauseQueueMember
* Asterisk 12 Application_UnpauseQueueMember
* Asterisk 12 Function_QUEUE_VARIABLES
* Asterisk 12 Function_QUEUE_MEMBER
* Asterisk 12 Function_QUEUE_MEMBER_COUNT
* Asterisk 12 Function_QUEUE_EXISTS
* Asterisk 12 Function_QUEUE_WAITING_COUNT
* Asterisk 12 Function_QUEUE_MEMBER_LIST
* Asterisk 12 Function_QUEUE_MEMBER_PENALTY

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_UserEvent

UserEvent()

Synopsis

Send an arbitrary user-defined event to parties interested in a channel (AMI users and relevant res_stasis applications).

Description

Sends an arbitrary event to interested parties, with an optional *body* representing additional arguments. The *body* may be specified as a `,` delimited list of key:value pairs.

For AMI, each additional argument will be placed on a new line in the event and the format of the event will be:

Event: UserEvent

UserEvent: <specified event name>

[body]

If no *body* is specified, only Event and UserEvent headers will be present.

For res_stasis applications, the event will be provided as a JSON blob with additional arguments appearing as keys in the object and the *eventname* under the *eventname* key.

Syntax

```
UserEvent(eventname,body)
```

Arguments

- `eventname`
- `body`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_Verbose

Verbose()

Synopsis

Send arbitrary text to verbose output.

Description

Sends an arbitrary text message to verbose output.

Syntax

```
Verbose(level,message)
```

Arguments

- `level` - Must be an integer value. If not specified, defaults to 0.
- `message` - Output text message.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_VMAuthenticate

VMAuthenticate()

Synopsis

Authenticate with Voicemail passwords.

Description

This application behaves the same way as the Authenticate application, but the passwords are taken from `voicemail.conf`. If the *mailbox* is specified, only that mailbox's password will be considered valid. If the *mailbox* is not specified, the channel variable `AUTH_MAILBOX` will be set with the authenticated mailbox.

The VMAuthenticate application will exit if the following DTMF digit is entered as Mailbox or Password, and the extension exists:

- `*` - Jump to the a extension in the current dialplan context.

Syntax

```
VMAuthenticate(mailbox@context,options)
```

Arguments

- `mailbox`
    - `mailbox`
    - `context`
- `options`
    - `s` - Skip playing the initial prompts.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_VMSayName

VMSayName()

Synopsis

Play the name of a voicemail user

Description

This application will say the recorded name of the voicemail user specified as the argument to this application. If no context is provided, `default` is assumed.

Syntax

```
VMSayName(mailbox@context)
```

Arguments

- `mailbox`
    - `mailbox`
    - `context`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_VoiceMail

VoiceMail()

Synopsis

Leave a Voicemail message.

Description

This application allows the calling party to leave a message for the specified list of mailboxes.

When multiple mailboxes are specified, the greeting will be taken from the first mailbox specified. Dialplan execution will stop if the specified mailbox does not exist.

The Voicemail application will exit if any of the following DTMF digits are received:

- `0` - Jump to the o extension in the current dialplan context.
- `*` - Jump to the a extension in the current dialplan context.
  This application will set the following channel variable upon completion:

- `VMSTATUS` - This indicates the status of the execution of the VoiceMail application.
  - SUCCESS
  - USEREXIT
  - FAILED

Syntax

```
VoiceMail(mailbox1&mailbox2[&...],options)
```

Arguments

- `mailboxs`
  - `mailbox1`
    - `mailbox`
    - `context`
  - `mailbox2`
    - `mailbox`
    - `context`
- `options`
  - `b` - Play the `busy` greeting to the calling party.
  - `d` - Accept digits for a new extension in context *c*, if played during the greeting. Context defaults to the current context.
    - `c`
  - `g` - Use the specified amount of gain when recording the voicemail message. The units are whole-number decibels (dB). Only works on supported technologies, which is DAHDI only.
    - `#`
  - `s` - Skip the playback of instructions for leaving a message to the calling party.
  - `u` - Play the `unavailable` greeting.
  - `U` - Mark message as `URGENT`.
  - `P` - Mark message as `PRIORITY`.

See Also

- Asterisk 12 Application_VoiceMailMain

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_VoiceMailMain

VoiceMailMain()

Synopsis

Check Voicemail messages.

Description

This application allows the calling party to check voicemail messages. A specific *mailbox*, and optional corresponding *context*, may be specified. If a *mailbox* is not provided, the calling party will be prompted to enter one. If a *context* is not specified, the `default` context will be used.

The VoiceMailMain application will exit if the following DTMF digit is entered as Mailbox or Password, and the extension exists:

- `*` - Jump to the a extension in the current dialplan context.

Syntax

```
VoiceMailMain(mailbox@context,options)
```

Arguments

- `mailbox`
    - `mailbox`
    - `context`
- `options`
    - p - Consider the *mailbox* parameter as a prefix to the mailbox that is entered by the caller.
    - g - Use the specified amount of gain when recording a voicemail message. The units are whole-number decibels (dB).
        - `#`
    - s - Skip checking the passcode for the mailbox.
    - a - Skip folder prompt and go directly to *folder* specified. Defaults to `INBOX` (or `0`).
        - `folder`
        - `0` - INBOX
        - `1` - Old
        - `2` - Work
        - `3` - Family
        - `4` - Friends
        - `5` - Cust1
        - `6` - Cust2
        - `7` - Cust3
        - `8` - Cust4
        - `9` - Cust5

See Also

- Asterisk 12 Application_VoiceMail

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_VoiceMailPlayMsg

VoiceMailPlayMsg()

Synopsis

Play a single voice mail msg from a mailbox by msg id.

Description

This application sets the following channel variable upon completion:

- `VOICEMAIL_PLAYBACKSTATUS` - The status of the playback attempt as a text string.
    - SUCCESS

- FAILED

Syntax

```
VoiceMailPlayMsg(mailbox@context,msg_id)
```

Arguments

- `mailbox`
    - `mailbox`
    - `context`
- `msg_id` - The msg id of the msg to play back.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_Wait

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.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_WaitExten

WaitExten()

Synopsis

Waits for an extension to be entered.

Description

This application waits for the user to enter a new extension for a specified number of *seconds*.

Warning
Use of the application `WaitExten` within a macro will not function as expected. Please use the `Read` application in order to read DTMF from a channel currently executing a macro.

Syntax

```
WaitExten(seconds,options)
```

Arguments

- `seconds` - Can be passed with fractions of a second. For example, `1.5` will ask the application to wait for 1.5 seconds.
- `options`
  - `m` - Provide music on hold to the caller while waiting for an extension.
    - `x` - Specify the class for music on hold. **CHANNEL(musicclass) will be used instead if set**

See Also

- Asterisk 12 Application_Background
- Asterisk 12 Function_TIMEOUT

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_WaitForNoise

WaitForNoise()

Synopsis

Waits for a specified amount of noise.

Description

Waits for up to *noiserequired* milliseconds of noise, *iterations* times. An optional *timeout* specified the number of seconds to return after, even if we do not receive the specified amount of noise. Use *timeout* with caution, as it may defeat the purpose of this application, which is to wait indefinitely until noise is detected on the line.

Syntax

```
WaitForNoise(noiserequired,iterations,timeout)
```

Arguments

- `noiserequired`
- `iterations` - If not specified, defaults to `1`.
- `timeout` - Is specified only to avoid an infinite loop in cases where silence is never achieved.

See Also

- Asterisk 12 Application_WaitForSilence

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_WaitForRing

WaitForRing()

Synopsis

Wait for Ring Application.

Description

Returns `0` after waiting at least *timeout* seconds, and only after the next ring has completed. Returns `0` on success or `-1` on hangup.

Syntax

```
WaitForRing(timeout)
```

Arguments

- `timeout`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_WaitForSilence

WaitForSilence()

Synopsis

Waits for a specified amount of silence.

Description

Waits for up to *silencerequired* milliseconds of silence, *iterations* times. An optional *timeout* speci fied the number of seconds to return after, even if we do not receive the specified amount of silence. Use *timeout* with caution, as it may defeat the purpose of this application, which is to wait indefinitely until silence is detected on the line. This is particularly useful for reverse-911-type call broadcast applications where you need to wait for an answering machine to complete its spiel before playing a message.

Typically you will want to include two or more calls to WaitForSilence when dealing with an answering machine; first waiting for the spiel to finish, then waiting for the beep, etc.

Examples:

WaitForSilence(500,2) will wait for 1/2 second of silence, twice

WaitForSilence(1000) will wait for 1 second of silence, once

WaitForSilence(300,3,10) will wait for 300ms silence, 3 times, and returns after 10 sec, even if silence is not detected

Sets the channel variable `WAITSTATUS` to one of these values:

- `WAITSTATUS`
    - SILENCE - if exited with silence detected.
    - TIMEOUT - if exited without silence detected after timeout.

Syntax

```
WaitForSilence(silencerequired,iterations,timeout)
```

Arguments

- `silencerequired`
- `iterations` - If not specified, defaults to `1`.
- `timeout` - Is specified only to avoid an infinite loop in cases where silence is never achieved.

See Also

- Asterisk 12 Application_WaitForNoise

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_WaitMusicOnHold

WaitMusicOnHold()

Synopsis

Wait, playing Music On Hold.

Description

!!! DEPRECATED. Use MusicOnHold instead !!!

Plays hold music specified number of seconds. Returns `0` when done, or `-1` on hangup. If no hold music is available, the delay will still occur with no sound.

!!! DEPRECATED. Use MusicOnHold instead !!!

Syntax

```
WaitMusicOnHold(delay)
```

Arguments

- `delay`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_WaitUntil

WaitUntil()

Synopsis

Wait (sleep) until the current time is the given epoch.

Description

Waits until the given *epoch*.

Sets `WAITUNTILSTATUS` to one of the following values:

- `WAITUNTILSTATUS`
  - OK - Wait succeeded.
  - FAILURE - Invalid argument.
  - HANGUP - Channel hungup before time elapsed.
  - PAST - Time specified had already past.

Syntax

```
WaitUntil(epoch)
```

Arguments

- `epoch`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_While

While()

Synopsis

Start a while loop.

Description

Start a While Loop. Execution will return to this point when `EndWhile()` is called until expr is no longer true.

Syntax

```
While(expr)
```

Arguments

- `expr`

See Also

- Asterisk 12 Application_EndWhile
- Asterisk 12 Application_ExitWhile
- Asterisk 12 Application_ContinueWhile

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Application_Zapateller

Zapateller()

Synopsis

Block telemarketers with SIT.

Description

Generates special information tone to block telemarketers from calling you.

This application will set the following channel variable upon completion:

- `ZAPATELLERSTATUS` - This will contain the last action accomplished by the Zapateller application. Possible values include:
    - NOTHING
    - ANSWERED
    - ZAPPED

Syntax

```
Zapateller(options)
```

Arguments

- `options` - Comma delimited list of options.
    - `answer` - Causes the line to be answered before playing the tone.
    - `nocallerid` - Causes Zapateller to only play the tone if there is no callerid information available.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Dialplan Functions

Asterisk 12 Function_AES_DECRYPT

AES_DECRYPT()

Synopsis

Decrypt a string encoded in base64 with AES given a 16 character key.

Description

Returns the plain text string.

Syntax

```
AES_DECRYPT(key,string)
```

Arguments

- `key` - AES Key
- `string` - Input string.

See Also

- Asterisk 12 Function_AES_ENCRYPT
- Asterisk 12 Function_BASE64_ENCODE
- Asterisk 12 Function_BASE64_DECODE

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_AES_ENCRYPT

AES_ENCRYPT()

Synopsis

Encrypt a string with AES given a 16 character key.

Description

Returns an AES encrypted string encoded in base64.

Syntax

```
AES_ENCRYPT(key,string)
```

Arguments

- `key` - AES Key
- `string` - Input string

See Also

- Asterisk 12 Function_AES_DECRYPT
- Asterisk 12 Function_BASE64_ENCODE
- Asterisk 12 Function_BASE64_DECODE

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_AGC

AGC()

Synopsis

Apply automatic gain control to audio on a channel.

Description

The AGC function will apply automatic gain control to the audio on the channel that it is executed on. Using `rx` for audio received and `tx` for audio transmitted to the channel. When using this function you set a target audio level. It is primarily intended for use with analog lines, but could be useful for other channels as well. The target volume is set with a number between `1-32768`. The larger the number the louder (more gain) the channel will receive.

Examples:

exten => 1,1,Set(AGC(rx)=8000)

exten => 1,2,Set(AGC(tx)=off)

Syntax

```
AGC(channeldirection)
```

Arguments

- `channeldirection` - This can be either `rx` or `tx`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_AGENT

AGENT()

Synopsis

Gets information about an Agent

Description

Syntax

```
AGENT(AgentId:item)
```

Arguments

- `AgentId`
- `item` - The valid items to retrieve are:
    - `status` - (default) The status of the agent (LOGGEDIN | LOGGEDOUT)
    - `password` - Deprecated. The dialplan handles any agent authentication.
    - `name` - The name of the agent
    - `mohclass` - MusicOnHold class
    - `channel` - The name of the active channel for the Agent (AgentLogin)
    - `fullchannel` - The untruncated name of the active channel for the Agent (AgentLogin)

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_AMI_CLIENT

AMI_CLIENT()

Synopsis

Checks attributes of manager accounts

Description

Currently, the only supported parameter is "sessions" which will return the current number of active sessions for this AMI account.

Syntax

```
AMI_CLIENT(loginname,field)
```

Arguments

- `loginname` - Login name, specified in manager.conf
- `field` - The manager account attribute to return
    - `sessions` - The number of sessions for this AMI account

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_ARRAY

ARRAY()

Synopsis

Allows setting multiple variables at once.

Description

The comma-delimited list passed as a value to which the function is set will be interpreted as a set of values to which the comma-delimited list of variable names in the argument should be set.

Example: Set(ARRAY(var1,var2)=1,2) will set var1 to 1 and var2 to 2

Syntax

```
ARRAY(var1[,var2[,...][,varN]])
```

Arguments

- `var1`
- `var2`

- `varN`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_AST_CONFIG

AST_CONFIG()

Synopsis

Retrieve a variable from a configuration file.

Description

This function reads a variable from an Asterisk configuration file.

Syntax

```
AST_CONFIG(config_file,category,variable_name)
```

Arguments

- `config_file`
- `category`
- `variable_name`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_AUDIOHOOK_INHERIT

AUDIOHOOK_INHERIT()

Synopsis

Set whether an audiohook may be inherited to another channel

Description

By enabling audiohook inheritance on the channel, you are giving permission for an audiohook to be inherited by a descendent channel. Inheritance may be be disabled at any point as well.

Example scenario:

exten => 2000,1,MixMonitor(blah.wav)

exten => 2000,n,Set(AUDIOHOOK_INHERIT(MixMonitor)=yes)

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

exten => 4000,1,Dial(SIP/4000)

exten => 5000,1,MixMonitor(blah2.wav)

exten => 5000,n,Dial(SIP/5000)

In this basic dialplan scenario, let's consider the following sample calls

Call 1: Caller dials 2000. The person who answers then executes an attended transfer to 4000.

Result: Since extension 2000 set MixMonitor to be inheritable, after the transfer to 4000 has completed, the call will continue to be recorded to blah.wav

Call 2: Caller dials 5000. The person who answers then executes an attended transfer to 4000.

Result: Since extension 5000 did not set MixMonitor to be inheritable, the recording will stop once the call has been transferred to 4000.

Syntax

```
AUDIOHOOK_INHERIT(source)
```

Arguments

- `source` - The built-in sources in Asterisk are
    - `MixMonitor`
    - `Chanspy`
    - `Volume`
    - `Speex`
    - `pitch_shift`
    - `JACK_HOOK`
    - `Mute`

    Note that the names are not case-sensitive

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_BASE64_DECODE

BASE64_DECODE()

Synopsis

Decode a base64 string.

Description

Returns the plain text string.

Syntax

```
BASE64_DECODE(string)
```

Arguments

- `string` - Input string.

See Also

- Asterisk 12 Function_BASE64_ENCODE
- Asterisk 12 Function_AES_DECRYPT
- Asterisk 12 Function_AES_ENCRYPT

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_BASE64_ENCODE

BASE64_ENCODE()

Synopsis

Encode a string in base64.

Description

Returns the base64 string.

Syntax

```
BASE64_ENCODE(string)
```

Arguments

- `string` - Input string

See Also

- Asterisk 12 Function_BASE64_DECODE
- Asterisk 12 Function_AES_DECRYPT
- Asterisk 12 Function_AES_ENCRYPT

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_BLACKLIST

BLACKLIST()

Synopsis

Check if the callerid is on the blacklist.

Description

Uses astdb to check if the Caller*ID is in family `blacklist`. Returns `1` or `0`.

Syntax

```
BLACKLIST()
```

Arguments

See Also

- Asterisk 12 Function_DB

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_CALENDAR_BUSY

CALENDAR_BUSY()

Synopsis

Determine if the calendar is marked busy at this time.

Description

Check the specified calendar's current busy status.

Syntax

```
CALENDAR_BUSY(calendar)
```

Arguments

- `calendar`

See Also

- Asterisk 12 Function_CALENDAR_EVENT
- Asterisk 12 Function_CALENDAR_QUERY
- Asterisk 12 Function_CALENDAR_QUERY_RESULT
- Asterisk 12 Function_CALENDAR_WRITE

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_CALENDAR_EVENT

CALENDAR_EVENT()

Synopsis

Get calendar event notification data from a notification call.

Description

Whenever a calendar event notification call is made, the event data may be accessed with this function.

Syntax

```
CALENDAR_EVENT(field)
```

Arguments

- `field`
    - `summary` - The VEVENT SUMMARY property or Exchange event 'subject'
    - `description` - The text description of the event
    - `organizer` - The organizer of the event
    - `location` - The location of the eventt
    - `categories` - The categories of the event
    - `priority` - The priority of the event
    - `calendar` - The name of the calendar associated with the event
    - `uid` - The unique identifier for this event
    - `start` - The start time of the event
    - `end` - The end time of the event
    - `busystate` - The busy state of the event 0=FREE, 1=TENTATIVE, 2=BUSY

See Also

- Asterisk 12 Function_CALENDAR_BUSY
- Asterisk 12 Function_CALENDAR_QUERY
- Asterisk 12 Function_CALENDAR_QUERY_RESULT
- Asterisk 12 Function_CALENDAR_WRITE

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_CALENDAR_QUERY

CALENDAR_QUERY()

Synopsis

Query a calendar server and store the data on a channel

Description

Get a list of events in the currently accessible timeframe of the *calendar* The function returns the id for accessing the result with CALENDAR_QUERY_RESULT()

Syntax

```
CALENDAR_QUERY(calendar[,start[,end]])
```

Arguments

- `calendar` - The calendar that should be queried
- `start` - The start time of the query (in seconds since epoch)
- `end` - The end time of the query (in seconds since epoch)

See Also

- Asterisk 12 Function_CALENDAR_BUSY

- Asterisk 12 Function_CALENDAR_EVENT
- Asterisk 12 Function_CALENDAR_QUERY_RESULT
- Asterisk 12 Function_CALENDAR_WRITE

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_CALENDAR_QUERY_RESULT

CALENDAR_QUERY_RESULT()

Synopsis

Retrieve data from a previously run CALENDAR_QUERY() call

Description

After running CALENDAR_QUERY and getting a result *id*, calling `CALENDAR_QUERY` with that *id* and a *field* will return the data for that field. If multiple events matched the query, and *entry* is provided, information from that event will be returned.

Syntax

```
CALENDAR_QUERY_RESULT(id,field[,entry])
```

Arguments

- `id` - The query ID returned by `CALENDAR_QUERY`
- `field`
  - `getnum` - number of events occurring during time range
  - `summary` - A summary of the event
  - `description` - The full event description
  - `organizer` - The event organizer
  - `location` - The event location
  - `categories` - The categories of the event
  - `priority` - The priority of the event
  - `calendar` - The name of the calendar associted with the event
  - `uid` - The unique identifier for the event
  - `start` - The start time of the event (in seconds since epoch)
  - `end` - The end time of the event (in seconds since epoch)
  - `busystate` - The busy status of the event 0=FREE, 1=TENTATIVE, 2=BUSY
- `entry` - Return data from a specific event returned by the query

See Also

- Asterisk 12 Function_CALENDAR_BUSY
- Asterisk 12 Function_CALENDAR_EVENT
- Asterisk 12 Function_CALENDAR_QUERY
- Asterisk 12 Function_CALENDAR_WRITE

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_CALENDAR_WRITE

CALENDAR_WRITE()

Synopsis

Write an event to a calendar

Description

Example: CALENDAR_WRITE(calendar,field1,field2,field3)=val1,val2,val3

The field and value arguments can easily be set/passed using the HASHKEYS() and HASH() functions

- `CALENDAR_SUCCESS` - The status of the write operation to the calendar
  - 1 - The event was successfully written to the calendar.
  - 0 - The event was not written to the calendar due to network issues, permissions, etc.

Syntax

```
CALENDAR_WRITE(calendar,field[,...])
```

Arguments

- `calendar` - The calendar to write to
- `field`
  - `summary` - A summary of the event
  - `description` - The full event description
  - `organizer` - The event organizer
  - `location` - The event location
  - `categories` - The categories of the event
  - `priority` - The priority of the event
  - `uid` - The unique identifier for the event
  - `start` - The start time of the event (in seconds since epoch)
  - `end` - The end time of the event (in seconds since epoch)
  - `busystate` - The busy status of the event 0=FREE, 1=TENTATIVE, 2=BUSY

See Also

- Asterisk 12 Function_CALENDAR_BUSY
- Asterisk 12 Function_CALENDAR_EVENT
- Asterisk 12 Function_CALENDAR_QUERY
- Asterisk 12 Function_CALENDAR_QUERY_RESULT

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_CALLCOMPLETION

CALLCOMPLETION()

Synopsis

Get or set a call completion configuration parameter for a channel.

Description

The CALLCOMPLETION function can be used to get or set a call completion configuration parameter for a channel. Note that setting a configuration parameter will only change the parameter for the duration of the call. For more information see `doc/AST.pdf`. For more information on call completion parameters, see `configs/ccss.conf.sample`.

Syntax

```
CALLCOMPLETION(option)
```

Arguments

- `option` - The allowable options are:
    - `cc_agent_policy`
    - `cc_monitor_policy`
    - `cc_offer_timer`
    - `ccnr_available_timer`
    - `ccbs_available_timer`
    - `cc_recall_timer`
    - `cc_max_agents`
    - `cc_max_monitors`
    - `cc_callback_macro`
    - `cc_agent_dialstring`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_CALLERID

CALLERID()

Synopsis

Gets or sets Caller*ID data on the channel.

Description

Gets or sets Caller*ID data on the channel. Uses channel callerid by default or optional callerid, if specified.

The allowable values for the *name-charset* field are the following:

- `unknown` - Unknown
- `iso8859-1` - ISO8859-1
- `withdrawn` - Withdrawn
- `iso8859-2` - ISO8859-2
- `iso8859-3` - ISO8859-3
- `iso8859-4` - ISO8859-4
- `iso8859-5` - ISO8859-5
- `iso8859-7` - ISO8859-7
- bmp - ISO10646 Bmp String
- `utf8` - ISO10646 UTF-8 String

Syntax

```
CALLERID(datatype,CID)
```

Arguments

- `datatype` - The allowable datatypes are:
    - `all`
    - `name`
    - `name-valid`
    - `name-charset`
    - `name-pres`
    - `num`
    - `num-valid`
    - `num-plan`
    - `num-pres`
    - `subaddr`
    - `subaddr-valid`
    - `subaddr-type`
    - `subaddr-odd`
    - `tag`
    - `priv-all`
    - `priv-name`
    - `priv-name-valid`
    - `priv-name-charset`
    - `priv-name-pres`
    - `priv-num`
    - `priv-num-valid`
    - `priv-num-plan`
    - `priv-num-pres`
    - `priv-subaddr`
    - `priv-subaddr-valid`
    - `priv-subaddr-type`
    - `priv-subaddr-odd`
    - `priv-tag`
    - `ANI-all`
    - `ANI-name`
    - `ANI-name-valid`
    - `ANI-name-charset`
    - `ANI-name-pres`
    - `ANI-num`
    - `ANI-num-valid`
    - `ANI-num-plan`
    - `ANI-num-pres`
    - `ANI-tag`
    - `RDNIS`
    - `DNID`
    - `dnid-num-plan`
    - `dnid-subaddr`
    - `dnid-subaddr-valid`
    - `dnid-subaddr-type`
    - `dnid-subaddr-odd`
- `CID` - Optional Caller*ID to parse instead of using the Caller*ID from the channel. This parameter is only optional when reading the Caller*ID.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_CALLERPRES

CALLERPRES()

Synopsis

Gets or sets Caller*ID presentation on the channel.

Description

Gets or sets Caller*ID presentation on the channel. This function is deprecated in favor of CALLERID(num-pres) and CALLERID(name-pres). The following values are valid:

- `allowed_not_screened` - Presentation Allowed, Not Screened.
- `allowed_passed_screen` - Presentation Allowed, Passed Screen.
- `allowed_failed_screen` - Presentation Allowed, Failed Screen.
- `allowed` - Presentation Allowed, Network Number.
- `prohib_not_screened` - Presentation Prohibited, Not Screened.
- `prohib_passed_screen` - Presentation Prohibited, Passed Screen.
- `prohib_failed_screen` - Presentation Prohibited, Failed Screen.
- `prohib` - Presentation Prohibited, Network Number.
- `unavailable` - Number Unavailable.

Syntax

```
CALLERPRES()
```

Arguments

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_CDR

CDR()

Synopsis

Gets or sets a CDR variable.

Description

All of the CDR field names are read-only, except for `accountcode`, `userfield`, and `amaflags`. You may, however, supply a name not on the above list, and create your own variable, whose value can be changed with this function, and this variable will be stored on the CDR.

> **Note**
> CDRs can only be modified before the bridge between two channels is torn down. For example, CDRs may not be modified after the `Dial` application has returned.

Example: exten => 1,1,Set(CDR(userfield)=test)

Syntax

```
CDR(name[,options])
```

Arguments

- `name` - CDR field name:
    - `clid` - Caller ID.
    - `lastdata` - Last application arguments.
    - `disposition` - The final state of the CDR.
        - `0` - NO ANSWER
        - `1` - NO ANSWER (NULL record)
        - `2` - FAILED
        - `4` - BUSY
        - `8` - ANSWERED
        - `16` - CONGESTION
    - `src` - Source.
    - `start` - Time the call started.
    - `amaflags` - R/W the Automatic Message Accounting (AMA) flags on the channel. When read from a channel, the integer value will always be returned. When written to a channel, both the string format or integer value is accepted.
        - `1` - OMIT
        - `2` - BILLING
        - `3` - DOCUMENTATION

        Warning
        Accessing this setting is deprecated in CDR. Please use the CHANNEL function instead.
    - `dst` - Destination.
    - `answer` - Time the call was answered.
    - `accountcode` - The channel's account code.

    Warning
    Accessing this setting is deprecated in CDR. Please use the CHANNEL function instead.
    - `dcontext` - Destination context.
    - `end` - Time the call ended.
    - `uniqueid` - The channel's unique id.
    - `dstchannel` - Destination channel.
    - `duration` - Duration of the call.
    - `userfield` - The channel's user specified field.
    - `lastapp` - Last application.
    - `billsec` - Duration of the call once it was answered.
    - `channel` - Channel name.
    - `sequence` - CDR sequence number.
- `options`
    - `f` - Returns billsec or duration fields as floating point values.
    - `u` - Retrieves the raw, unprocessed value.
    For example, 'start', 'answer', and 'end' will be retrieved as epoch values, when the `u` option is passed, but formatted as YYYY-MM-DD HH:MM:SS otherwise. Similarly, disposition and amaflags will return their raw integral values.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_CDR_PROP

CDR_PROP()

Synopsis

Set a property on a channel's CDR.

Description

This function sets a property on a channel's CDR. Properties alter the behavior of how the CDR operates for that channel.

Syntax

```
CDR_PROP(name)
```

Arguments

- `name` - The property to set on the CDR.
    - `party_a` - Set this channel as the preferred Party A when channels are associated together.
      Write-Only
    - `disable` - Disable CDRs for this channel.
      Write-Only

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_CHANNEL

CHANNEL()

Synopsis

Gets/sets various pieces of information about the channel.

Description

Gets/sets various pieces of information about the channel, additional *item* 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.

Syntax

```
CHANNEL(item)
```

Arguments

- `item` - Standard items (provided by all channel technologies) are:
    - `amaflags` - R/W the Automatic Message Accounting (AMA) flags on the channel. When read from a channel, the integer value will always be returned. When written to a channel, both the string format or integer value is accepted.
        - `1` - OMIT
        - `2` - BILLING
        - `3` - DOCUMENTATION
    - `accountcode` - R/W the channel's account code.
    - `audioreadformat` - R/O format currently being read.
    - `audionativeformat` - R/O format used natively for audio.
    - `audiowriteformat` - R/O format currently being written.
    - `dtmf_features` - R/W The channel's DTMF bridge features. May include one or more of 'T' 'K' 'H' 'W' and 'X' in a similar manner to options in the `Dial` application. When setting it, the features string must be all upper case.
    - `callgroup` - R/W numeric call pickup groups that this channel is a member.
    - `pickupgroup` - R/W numeric call pickup groups this channel can pickup.
    - `namedcallgroup` - R/W named call pickup groups that this channel is a member.
    - `namedpickupgroup` - R/W named call pickup groups this channel can pickup.
    - `channeltype` - R/O technology used for channel.
    - `checkhangup` - R/O Whether the channel is hanging up (1/0)
    - `after_bridge_goto` - R/W the parseable goto string indicating where the channel is expected to return to in the PBX after exiting the next bridge it joins on the condition that it doesn't hang up. The parseable goto string uses the same syntax as the `Goto` application.
    - `hangup_handler_pop` - W/O Replace the most recently added hangup handler with a new hangup handler on the channel if supplied. The assigned string is passed to the Gosub application when the channel is hung up. Any optionally omitted context and exten are supplied by the channel pushing the handler before it is pushed.
    - `hangup_handler_push` - W/O Push a hangup handler onto the channel hangup handler stack. The assigned string is passed to the Gosub application when the channel is hung up. Any optionally omitted context and exten are supplied by the channel pushing the handler before it is pushed.
    - `hangup_handler_wipe` - W/O Wipe the entire hangup handler stack and replace with a new hangup handler on the channel if supplied. The assigned string is passed to the Gosub application when the channel is hung up. Any optionally omitted context and exten are supplied by the channel pushing the handler before it is pushed.
    - `language` - R/W language for sounds played.
    - `musicclass` - R/W class (from musiconhold.conf) for hold music.
    - `name` - The name of the channel
    - `parkinglot` - R/W parkinglot for parking.
    - `rxgain` - R/W set rxgain level on channel drivers that support it.
    - `secure_bridge_signaling` - Whether or not channels bridged to this channel require secure signaling
    - `secure_bridge_media` - Whether or not channels bridged to this channel require secure media
    - `state` - R/O state for channel
    - `tonezone` - R/W zone for indications played
    - `transfercapability` - R/W ISDN Transfer Capability, one of:
        - `SPEECH`
        - `DIGITAL`
        - `RESTRICTED_DIGITAL`
        - `3K1AUDIO`
        - `DIGITAL_W_TONES`
        - `VIDEO`
    - `txgain` - R/W set txgain level on channel drivers that support it.
    - `videonativeformat` - R/O format used natively for video
    - `trace` - R/W whether or not context tracing is enabled, only available **if CHANNEL_TRACE is defined**.
      chan_sip provides the following additional options:
    - `peerip` - R/O Get the IP address of the peer.
    - `recvip` - R/O Get the source IP address of the peer.
    - `from` - R/O Get the URI from the From: header.
    - `uri` - R/O Get the URI from the Contact: header.
    - `useragent` - R/O Get the useragent.
    - `peername` - R/O Get the name of the peer.

* `t38passthrough` - R/O `1` if T38 is offered or enabled in this channel, otherwise `0`
* `rtpqos` - R/O Get QOS information about the RTP stream
  This option takes two additional arguments:
  Argument 1:
  `audio` Get data about the audio stream
  `video` Get data about the video stream
  `text` Get data about the text stream
  Argument 2:
  `local_ssrc` Local SSRC (stream ID)
  `local_lostpackets` Local lost packets
  `local_jitter` Local calculated jitter
  `local_maxjitter` Local calculated jitter (maximum)
  `local_minjitter` Local calculated jitter (minimum)
  {{local_normdevjitter}}Local calculated jitter (normal deviation)
  `local_stdevjitter` Local calculated jitter (standard deviation)
  `local_count` Number of received packets
  `remote_ssrc` Remote SSRC (stream ID)
  {{remote_lostpackets}}Remote lost packets
  `remote_jitter` Remote reported jitter
  `remote_maxjitter` Remote calculated jitter (maximum)
  `remote_minjitter` Remote calculated jitter (minimum)
  {{remote_normdevjitter}}Remote calculated jitter (normal deviation)
  {{remote_stdevjitter}}Remote calculated jitter (standard deviation)
  `remote_count` Number of transmitted packets
  `rtt` Round trip time
  `maxrtt` Round trip time (maximum)
  `minrtt` Round trip time (minimum)
  `normdevrtt` Round trip time (normal deviation)
  `stdevrtt` Round trip time (standard deviation)
  `all` All statistics (in a form suited to logging, but not for parsing)
* `rtpdest` - R/O Get remote RTP destination information.
  This option takes one additional argument:
  Argument 1:
  `audio` Get audio destination
  `video` Get video destination
  `text` Get text destination
  Defaults to `audio` if unspecified.
* `rtpsource` - R/O Get source RTP destination information.
  This option takes one additional argument:
  Argument 1:
  `audio` Get audio destination
  `video` Get video destination
  `text` Get text destination
  Defaults to `audio` if unspecified.
  chan_iax2 provides the following additional options:
* `osptoken` - R/O Get the peer's osptoken.
* `peerip` - R/O Get the peer's ip address.
* `peername` - R/O Get the peer's username.
* `secure_signaling` - R/O Get the if the IAX channel is secured.
* `secure_media` - R/O Get the if the IAX channel is secured.
  chan_dahdi provides the following additional options:
* `dahdi_channel` - R/O DAHDI channel related to this channel.
* `dahdi_span` - R/O DAHDI span related to this channel.
* `dahdi_type` - R/O DAHDI channel type, one of:
  * `analog`
  * `mfc/r2`
  * `pri`
  * `pseudo`

        - `ss7`
- `keypad_digits` - R/O PRI Keypad digits that came in with the SETUP message.
- `reversecharge` - R/O PRI Reverse Charging Indication, one of:
    - `-1` - None
    - {{ 1}} - Reverse Charging Requested
- `no_media_path` - R/O PRI Nonzero if the channel has no B channel. The channel is either on hold or a call waiting call.
- `buffers` - W/O Change the channel's buffer policy (for the current call only)
  This option takes two arguments:
  Number of buffers,
  Buffer policy being one of:
  `full`
  `immediate`
  `half`
- `echocan_mode` - W/O Change the configuration of the active echo canceller on the channel (if any), for the current call only.
  Possible values are:
  {{on}}Normal mode (the echo canceller is actually reinitalized)
  {{off}}Disabled
  {{fax}}FAX/data mode (NLP disabled if possible, otherwise completely disabled)
  {{voice}}Voice mode (returns from FAX mode, reverting the changes that were made)
  chan_ooh323 provides the following additional options:
- `faxdetect` - R/W Fax Detect
  Returns 0 or 1
  Write yes or no
- `t38support` - R/W t38support
  Returns 0 or 1
  Write yes or no
- `h323id_url` - R/0 Returns caller URL
- `caller_h323id` - R/0 Returns caller h323id
- `caller_dialeddigits` - R/0 Returns caller dialed digits
- `caller_email` - R/0 Returns caller email
- `callee_email` - R/0 Returns callee email
- `callee_dialeddigits` - R/0 Returns callee dialed digits
- `caller_url` - R/0 Returns caller URL

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_CHANNELS

CHANNELS()

Synopsis

Gets the list of channels, optionally filtering by a regular expression.

Description

Gets the list of channels, optionally filtering by a *regular_expression*. 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.

Syntax

```
CHANNELS(regular_expression)
```

Arguments

- `regular_expression`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_CHECKSIPDOMAIN

CHECKSIPDOMAIN()

Synopsis

Checks if domain is a local domain.

Description

This function checks if the *domain* in the argument is configured as a local SIP domain that this Asterisk server is configured to handle. Returns the domain name if it is locally handled, otherwise an empty string. Check the `domain=` configuration in `sip.conf`.

Syntax

```
CHECKSIPDOMAIN(domain)
```

Arguments

- `domain`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_CONFBRIDGE

CONFBRIDGE()

Synopsis

Set a custom dynamic bridge and user profile on a channel for the ConfBridge application using the same options defined in confbridge.conf.

Description

---- Example 1 ----

In this example the custom set user profile on this channel will automatically be used by the

ConfBridge app.

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 ----

This example shows how to use a predefined user or bridge profile in confbridge.conf as a template for a dynamic profile. Here we make a admin/marked user out of the default_user profile that is already defined in confbridge.conf.

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)

Syntax

```
CONFBRIDGE(type,option)
```

Arguments

- `type` - Type refers to which type of profile the option belongs too. Type can be `bridge` or `user`.
- `option` - Option refers to `confbridge.conf` option that is being set dynamically on this channel.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_CONFBRIDGE_INFO

CONFBRIDGE_INFO()

Synopsis

Get information about a ConfBridge conference.

Description

This function returns a non-negative integer for valid conference identifiers (0 or 1 for `locked`) and "" for invalid conference identifiers.

Syntax

```
CONFBRIDGE_INFO(type,conf)
```

Arguments

- `type` - Type can be `parties`, `admins`, `marked`, or `locked`.
- `conf` - Conf refers to the name of the conference being referenced.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_CONNECTEDLINE

CONNECTEDLINE()

Synopsis

Gets or sets Connected Line data on the channel.

Description

Gets or sets Connected Line data on the channel.

The allowable values for the *name-charset* field are the following:

- `unknown` - Unknown
- `iso8859-1` - ISO8859-1
- `withdrawn` - Withdrawn
- `iso8859-2` - ISO8859-2
- `iso8859-3` - ISO8859-3
- `iso8859-4` - ISO8859-4
- `iso8859-5` - ISO8859-5
- `iso8859-7` - ISO8859-7
- bmp - ISO10646 Bmp String
- `utf8` - ISO10646 UTF-8 String

Syntax

```
CONNECTEDLINE(datatype,i)
```

Arguments

- `datatype` - The allowable datatypes are:
    - `all`
    - `name`
    - `name-valid`
    - `name-charset`
    - `name-pres`
    - `num`
    - `num-valid`
    - `num-plan`
    - `num-pres`
    - `subaddr`
    - `subaddr-valid`

    - `subaddr-type`
    - `subaddr-odd`
    - `tag`
    - `priv-all`
    - `priv-name`
    - `priv-name-valid`
    - `priv-name-charset`
    - `priv-name-pres`
    - `priv-num`
    - `priv-num-valid`
    - `priv-num-plan`
    - `priv-num-pres`
    - `priv-subaddr`
    - `priv-subaddr-valid`
    - `priv-subaddr-type`
    - `priv-subaddr-odd`
    - `priv-tag`
- `i` - If set, this will prevent the channel from sending out protocol messages because of the value being set

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_CSV_QUOTE

CSV_QUOTE()

Synopsis

Quotes a given string for use in a CSV file, escaping embedded quotes as necessary

Description

Example: ${CSV_QUOTE("a,b" 123)} will return """a,b"" 123"

Syntax

```
CSV_QUOTE(string)
```

Arguments

- `string`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_CURL

CURL()

Synopsis

Retrieve content from a remote web or ftp server

Description

Syntax

```
CURL(url,post-data)
```

Arguments

- `url`
- `post-data` - If specified, an `HTTP POST` will be performed with the content of *post-data*, instead of an `HTTP GET` (default).

See Also

- Asterisk 12 Function_CURLOPT

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_CURLOPT

CURLOPT()

Synopsis

Sets various options for future invocations of CURL.

Description

Options may be set globally or per channel. Per-channel settings will override global settings.

Syntax

```
CURLOPT(key)
```

Arguments

- `key`
    - `cookie` - A cookie to send with the request. Multiple cookies are supported.
    - `conntimeout` - Number of seconds to wait for a connection to succeed
    - `dnstimeout` - Number of seconds to wait for DNS to be resolved
    - `ftptext` - For FTP URIs, force a text transfer (boolean)
    - `ftptimeout` - For FTP URIs, number of seconds to wait for a server response
    - `header` - Include header information in the result (boolean)
    - `httptimeout` - For HTTP(S) URIs, number of seconds to wait for a server response
    - `maxredirs` - Maximum number of redirects to follow
    - `proxy` - Hostname or IP address to use as a proxy server
    - `proxytype` - Type of `proxy`
        - `http`
        - `socks4`
        - `socks5`
    - `proxyport` - Port number of the `proxy`
    - `proxyuserpwd` - A *username*`:`*password* combination to use for authenticating requests through a `proxy`
    - `referer` - Referer URL to use for the request
    - `useragent` - UserAgent string to use for the request
    - `userpwd` - A *username*`:`*password* to use for authentication when the server response to an initial request indicates a 401 status code.
    - `ssl_verifypeer` - Whether to verify the server certificate against a list of known root certificate authorities (boolean).

- `hashcompat` - Assuming the responses will be in `key1=value1&key2=value2` format, reformat the response such that it can be used by the `HASH` function.
    - `yes`
    - `no`
    - `legacy` - Also translate + to the space character, in violation of current RFC standards.

See Also

- Asterisk 12 Function_CURL
- Asterisk 12 Function_HASH

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_CUT

CUT()

Synopsis

Slices and dices strings, based upon a named delimiter.

Description

Cut out information from a string (*varname*), based upon a named delimiter.

Syntax

```
CUT(varname,char-delim,range-spec)
```

Arguments

- `varname` - Variable you want cut
- `char-delim` - Delimiter, defaults to `-`
- `range-spec` - Number of the field you want (1-based offset), may also be specified as a range (with `-`) or group of ranges and fields (with `&`)

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_DB

DB()

Synopsis

Read from or write to the Asterisk database.

Description

This function will read from or write a value to the Asterisk 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. If you wish to find out if an entry exists, use the DB_EXISTS function.

Syntax

```
DB(family/key)
```

Arguments

- `family`
- `key`

See Also

- Asterisk 12 Application_DBdel
- Asterisk 12 Function_DB_DELETE
- Asterisk 12 Application_DBdeltree
- Asterisk 12 Function_DB_EXISTS

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_DB_DELETE

DB_DELETE()

Synopsis

Return a value from the database and delete it.

Description

This 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.

Syntax

```
DB_DELETE(family/key)
```

Arguments

- `family`
- `key`

See Also

- Asterisk 12 Application_DBdel
- Asterisk 12 Function_DB
- Asterisk 12 Application_DBdeltree

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_DB_EXISTS

DB_EXISTS()

Synopsis

Check to see if a key exists in the Asterisk database.

Description

This function will check to see if a key exists in the Asterisk database. If it exists, the function will return `1`. If not, it will return `0`. Checking for existence of a database key will also set the variable DB_RESULT to the key's value if it exists.

Syntax

```
DB_EXISTS(family/key)
```

Arguments

- `family`
- `key`

See Also

- Asterisk 12 Function_DB

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_DB_KEYS

DB_KEYS()

Synopsis

Obtain a list of keys within the Asterisk database.

Description

This function 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.

Syntax

```
DB_KEYS(prefix)
```

Arguments

- `prefix`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_DEC

DEC()

Synopsis

Decrements the value of a variable, while returning the updated value to the dialplan

Description

Decrements the value of a variable, while returning the updated value to the dialplan

Example: DEC(MyVAR) - Decrements MyVar

Note: DEC(${MyVAR}) - Is wrong, as DEC expects the variable name, not its value

Syntax

```
DEC(variable)
```

Arguments

- `variable` - The variable name to be manipulated, without the braces.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_DENOISE

DENOISE()

Synopsis

Apply noise reduction to audio on a channel.

Description

The DENOISE function will apply noise reduction to audio on the channel that it is executed on. It is very useful for noisy analog lines, especially when adjusting gains or using AGC. Use `rx` for audio received from the channel and `tx` to apply the filter to the audio being sent to the channel.

Examples:

exten => 1,1,Set(DENOISE(rx)=on)

exten => 1,2,Set(DENOISE(tx)=off)

Syntax

```
DENOISE(channeldirection)
```

Arguments

- `channeldirection` - This can be either `rx` or `tx` the values that can be set to this are either `on` and `off`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_DEVICE_STATE

DEVICE_STATE()

Synopsis

Get or Set a device state.

Description

The DEVICE_STATE function can be used to retrieve the device state from any device state provider. For example:

NoOp(SIP/mypeer has state ${DEVICE_STATE(SIP/mypeer)})

NoOp(Conference number 1234 has state ${DEVICE_STATE(MeetMe:1234)})

The DEVICE_STATE function can also be used to set custom device state from the dialplan. The `Custom:` prefix must be used. For example:

Set(DEVICE_STATE(Custom:lamp1)=BUSY)

Set(DEVICE_STATE(Custom:lamp2)=NOT_INUSE)

You can subscribe to the status of a custom device state using a hint in the dialplan:

exten => 1234,hint,Custom:lamp1

The possible values for both uses of this function are:

UNKNOWN | NOT_INUSE | INUSE | BUSY | INVALID | UNAVAILABLE | RINGING | RINGINUSE | ONHOLD

Syntax

```
DEVICE_STATE(device)
```

Arguments

- `device`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_DIALGROUP

DIALGROUP()

Synopsis

Manages a group of users for dialing.

Description

Presents an interface meant to be used in concert with the Dial application, by presenting a list of channels which should be dialled when referenced.

When DIALGROUP is read from, the argument is interpreted as the particular *group* for which a dial should be attempted. When DIALGROUP is written to with no arguments, the entire list is replaced with the argument specified.

Functionality is similar to a queue, except that when no interfaces are available, execution may continue in the dialplan. This is useful when you want certain people to be the first to answer any calls, with immediate fallback to a queue when the front line people are busy or unavailable, but you still want front line people to log in and out of that group, just like a queue.

Example:

exten => 1,1,Set(DIALGROUP(mygroup,add)=SIP/10)

exten => 1,n,Set(DIALGROUP(mygroup,add)=SIP/20)

exten => 1,n,Dial(${DIALGROUP(mygroup)})

Syntax

```
DIALGROUP(group,op)
```

Arguments

- `group`
- `op` - The operation name, possible values are:
  add - add a channel name or interface (write-only)
  del - remove a channel name or interface (write-only)

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_DIALPLAN_EXISTS

DIALPLAN_EXISTS()

Synopsis

Checks the existence of a dialplan target.

Description

This function returns `1` if the target exits. Otherwise, it returns `0`.

Syntax

```
DIALPLAN_EXISTS(context,extension,priority)
```

Arguments

- `context`
- `extension`
- `priority`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_DUNDILOOKUP

DUNDILOOKUP()

Synopsis

Do a DUNDi lookup of a phone number.

Description

This will do a DUNDi lookup of the given phone number.

This function will return the Technology/Resource found in the first result in the DUNDi lookup. If no results were found, the result will be blank.

Syntax

```
DUNDILOOKUP(number,context,options)
```

Arguments

- `number`
- `context` - If not specified the default will be `e164`.
- `options`
    - b - Bypass the internal DUNDi cache

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_DUNDIQUERY

DUNDIQUERY()

Synopsis

Initiate a DUNDi query.

Description

This will do a DUNDi lookup of the given phone number.

The result of this function will be a numeric ID that can be used to retrieve the results with the `DUNDIRESULT` function.

Syntax

```
DUNDIQUERY(number,context,options)
```

Arguments

- `number`
- `context` - If not specified the default will be `e164`.
- `options`
    - b - Bypass the internal DUNDi cache

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_DUNDIRESULT

DUNDIRESULT()

Synopsis

Retrieve results from a DUNDIQUERY.

Description

This function will retrieve results from a previous use\n" of the `DUNDIQUERY` function.

Syntax

```
DUNDIRESULT(id,resultnum)
```

Arguments

- `id` - The identifier returned by the `DUNDIQUERY` function.
- `resultnum`
    - `number` - The number of the result that you want to retrieve, this starts at `1`
    - `getnum` - The total number of results that are available.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_ENUMLOOKUP

ENUMLOOKUP()

Synopsis

General or specific querying of NAPTR records for ENUM or ENUM-like DNS pointers.

Description

For more information see `doc/AST.pdf`.

Syntax

```
ENUMLOOKUP(number,method-type,options,record#,zone-suffix)
```

Arguments

- `number`
- `method-type` - If no *method-type* is given, the default will be `sip`.
- `options`
    - `c` - Returns an integer count of the number of NAPTRs of a certain RR type.
      Combination of `c` and Method-type of `ALL` will return a count of all NAPTRs for the record or -1 on error.
    - `u` - Returns the full URI and does not strip off the URI-scheme.
    - `s` - Triggers ISN specific rewriting.
    - `i` - Looks for branches into an Infrastructure ENUM tree.
    - `d` - for a direct DNS lookup without any flipping of digits.
- `record#` - If no *record#* is given, defaults to `1`.
- `zone-suffix` - If no *zone-suffix* is given, the default will be `e164.arpa`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_ENUMQUERY

ENUMQUERY()

Synopsis

Initiate an ENUM query.

Description

This will do a ENUM lookup of the given phone number.

Syntax

```
ENUMQUERY(number,method-type,zone-suffix)
```

Arguments

- `number`
- `method-type` - If no *method-type* is given, the default will be `sip`.
- `zone-suffix` - If no *zone-suffix* is given, the default will be `e164.arpa`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_ENUMRESULT

ENUMRESULT()

Synopsis

Retrieve results from a ENUMQUERY.

Description

This function will retrieve results from a previous use of the ENUMQUERY function.

Syntax

```
ENUMRESULT(id,resultnum)
```

Arguments

- `id` - The identifier returned by the ENUMQUERY function.
- `resultnum` - The number of the result that you want to retrieve.
  Results start at `1`. If this argument is specified as `getnum`, then it will return the total number of results that are available or `-1` on error.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_ENV

ENV()

Synopsis

Gets or sets the environment variable specified.

Description

Variables starting with `AST_` are reserved to the system and may not be set.

Syntax

```
ENV(varname)
```

Arguments

- `varname` - Environment variable name

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_EVAL

EVAL()

Synopsis

Evaluate stored variables

Description

Using EVAL basically causes a string to be evaluated twice. When a variable or expression is in the dialplan, it will be evaluated at runtime. However, if the results of the evaluation is in fact another variable or expression, using EVAL will have it evaluated a second time.

Example: If the `MYVAR` contains `OTHERVAR`, then the result of ${EVAL(`MYVAR`)} in the dialplan will be the contents of `OTHERVAR`. Normally just putting `MYVAR` in the dialplan the result would be `OTHERVAR`.

Syntax

```
EVAL(variable)
```

Arguments

- `variable`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_EXCEPTION

EXCEPTION()

Synopsis

Retrieve the details of the current dialplan exception.

Description

Retrieve the details (specified *field*) of the current dialplan exception.

Syntax

```
EXCEPTION(field)
```

Arguments

- `field` - The following fields are available for retrieval:
    - `reason` - INVALID, ERROR, RESPONSETIMEOUT, ABSOLUTETIMEOUT, or custom value set by the RaiseException() application
    - `context` - The context executing when the exception occurred.
    - `exten` - The extension executing when the exception occurred.
    - `priority` - The numeric priority executing when the exception occurred.

See Also

- Asterisk 12 Application_RaiseException

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_EXISTS

EXISTS()

Synopsis

Test the existence of a value.

Description

Returns `1` if exists, `0` otherwise.

Syntax

```
EXISTS(data)
```

Arguments

- `data`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_EXTENSION_STATE

EXTENSION_STATE()

Synopsis

Get an extension's state.

Description

The EXTENSION_STATE function can be used to retrieve the state from any hinted extension. For example:

NoOp(1234@default has state ${EXTENSION_STATE(1234)})

NoOp(4567@home has state ${EXTENSION_STATE(4567@home)})

The possible values returned by this function are:

UNKNOWN | NOT_INUSE | INUSE | BUSY | INVALID | UNAVAILABLE | RINGING | RINGINUSE | HOLDINUSE | ONHOLD

Syntax

```
EXTENSION_STATE(extension@context)
```

Arguments

- `extension`
- `context` - If it is not specified defaults to `default`.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_FAXOPT_res_fax

FAXOPT() - [res_fax]

Synopsis

Gets/sets various pieces of information about a fax session.

Description

FAXOPT can be used to override the settings for a FAX session listed in `res_fax.conf`, it can also be used to retreive information about a FAX session that has finished eg. pages/status.

Syntax

```
FAXOPT(item)
```

Arguments

- `item`
    - `ecm` - R/W Error Correction Mode (ECM) enable with 'yes', disable with 'no'.
    - `error` - R/O FAX transmission error code upon failure.
    - `filename` - R/O Filename of the first file of the FAX transmission.
    - `filenames` - R/O Filenames of all of the files in the FAX transmission (comma separated).
    - `headerinfo` - R/W FAX header information.
    - `localstationid` - R/W Local Station Identification.
    - `minrate` - R/W Minimum transfer rate set before transmission.
    - `maxrate` - R/W Maximum transfer rate set before transmission.
    - `modem` - R/W Modem type (v17/v27/v29).
    - `gateway` - R/W T38 fax gateway, with optional fax activity timeout in seconds (yes[,timeout]/no)
    - `faxdetect` - R/W Enable FAX detect with optional timeout in seconds (yes,t38,cng[,timeout]/no)
    - `pages` - R/O Number of pages transferred.
    - `rate` - R/O Negotiated transmission rate.
    - `remotestationid` - R/O Remote Station Identification after transmission.
    - `resolution` - R/O Negotiated image resolution after transmission.
    - `sessionid` - R/O Session ID of the FAX transmission.
    - `status` - R/O Result Status of the FAX transmission.
    - `statusstr` - R/O Verbose Result Status of the FAX transmission.

See Also

- Asterisk 12 Application_ReceiveFax

* Asterisk 12 Application_SendFax

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_FEATURE

FEATURE()

Synopsis

Get or set a feature option on a channel.

Description

When this function is used as a read, it will get the current value of the specified feature option for this channel. It will be the value of this option configured in features.conf if a channel specific value has not been set. This function can also be used to set a channel specific value for the supported feature options.

Syntax

```
FEATURE(option_name)
```

Arguments

* `option_name` - The allowed values are:
    * `inherit` - Inherit feature settings made in FEATURE or FEATUREMAP to child channels.
    * `featuredigittimeout` - Milliseconds allowed between digit presses when entering a feature code.
    * `transferdigittimeout` - Milliseconds allowed between digit presses when dialing a transfer destination
    * `atxfernoanswertimeout` - Milliseconds to wait for attended transfer destination to answer
    * `atxferdropcall` - Hang up the call entirely if the attended transfer fails
    * `atxferloopdelay` - Milliseconds to wait between attempts to re-dial transfer destination
    * `atxfercallbackretries` - Number of times to re-attempt dialing a transfer destination
    * `xfersound` - Sound to play to during transfer and transfer-like operations.
    * `xferfailsound` - Sound to play to a transferee when a transfer fails
    * `atxferabort` - Digits to dial to abort an attended transfer attempt
    * `atxfercomplete` - Digits to dial to complete an attended transfer
    * `atxferthreeway` - Digits to dial to change an attended transfer into a three-way call
    * `pickupexten` - Digits used for picking up ringing calls
    * `pickupsound` - Sound to play to picker when a call is picked up
    * `pickupfailsound` - Sound to play to picker when a call cannot be picked up
    * `courtesytone` - Sound to play when automon or automixmon is activated
    * `recordingfailsound` - Sound to play when automon or automixmon is attempted but fails to start

See Also

* Asterisk 12 Function_FEATUREMAP

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_FEATUREMAP

FEATUREMAP()

Synopsis

Get or set a feature map to a given value on a specific channel.

Description

When this function is used as a read, it will get the current digit sequence mapped to the specified feature for this channel. This value will be the one configured in features.conf if a channel specific value has not been set. This function can also be used to set a channel specific value for a feature mapping.

Syntax

```
FEATUREMAP(feature_name)
```

Arguments

- `feature_name` - The allowed values are:
    - `atxfer` - Attended Transfer
    - `blindxfer` - Blind Transfer
    - `automon` - Auto Monitor
    - `disconnect` - Call Disconnect
    - `parkcall` - Park Call
    - `automixmon` - Auto MixMonitor

See Also

- Asterisk 12 Function_FEATURE

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_FIELDNUM

FIELDNUM()

Synopsis

Return the 1-based offset of a field in a list

Description

Search the variable named *varname* for the string *value* delimited by *delim* and return a 1-based offset as to its location. If not found or an error occured, return `0`.

The delimiter may be specified as a special or extended ASCII character, by encoding it. The characters `\n`, `\r`, and `\t` are all recognized as the newline, carriage return, and tab characters, respectively. Also, octal and hexadecimal specifications are recognized by the patterns `\0nnn` a nd `\xHH`, respectively. For example, if you wanted to encode a comma as the delimiter, you could use either `\054` or `\x2C`.

Example: If ${example} contains `ex-amp-le`, then ${FIELDNUM(example,-,amp)} returns 2.

Syntax

```
FIELDNUM(varname,delim,value)
```

Arguments

- `varname`
- `delim`
- `value`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_FIELDQTY

FIELDQTY()

Synopsis

Count the fields with an arbitrary delimiter

Description

The delimiter may be specified as a special or extended ASCII character, by encoding it. The characters `\n`, `\r`, and `\t` are all recognized as the newline, carriage return, and tab characters, respectively. Also, octal and hexadecimal specifications are recognized by the patterns `\0nnn` a nd `\xHH`, respectively. For example, if you wanted to encode a comma as the delimiter, you could use either `\054` or `\x2C`.

Example: If ${example} contains `ex-amp-le`, then ${FIELDQTY(example,-)} returns 3.

Syntax

```
FIELDQTY(varname,delim)
```

Arguments

- `varname`
- `delim`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_FILE

FILE()

Synopsis

Read or write text file.

Description

Read and write text file in character and line mode.

Examples:

Read mode (byte):

;reads the entire content of the file.

Set(foo=${FILE(/tmp/test.txt)})

;reads from the 11th byte to the end of the file (i.e. skips the first 10).

Set(foo=${FILE(/tmp/test.txt,10)})

;reads from the 11th to 20th byte in the file (i.e. skip the first 10, then read 10 bytes).

Set(foo=${FILE(/tmp/test.txt,10,10)})

Read mode (line):

; reads the 3rd line of the file.

Set(foo=${FILE(/tmp/test.txt,3,1,l)})

; reads the 3rd and 4th lines of the file.

Set(foo=${FILE(/tmp/test.txt,3,2,l)})

; reads from the third line to the end of the file.

Set(foo=${FILE(/tmp/test.txt,3,,l)})

; reads the last three lines of the file.

Set(foo=${FILE(/tmp/test.txt,-3,,l)})

; reads the 3rd line of a DOS-formatted file.

Set(foo=${FILE(/tmp/test.txt,3,1,l,d)})

Write mode (byte):

; truncate the file and write "bar"

Set(FILE(/tmp/test.txt)=bar)

; Append "bar"

Set(FILE(/tmp/test.txt,,,a)=bar)

; Replace the first byte with "bar" (replaces 1 character with 3)

Set(FILE(/tmp/test.txt,0,1)=bar)

; Replace 10 bytes beginning at the 21st byte of the file with "bar"

Set(FILE(/tmp/test.txt,20,10)=bar)

; Replace all bytes from the 21st with "bar"

Set(FILE(/tmp/test.txt,20)=bar)

; Insert "bar" after the 4th character

Set(FILE(/tmp/test.txt,4,0)=bar)

Write mode (line):

; Replace the first line of the file with "bar"

Set(FILE(/tmp/foo.txt,0,1,l)=bar)

; Replace the last line of the file with "bar"

Set(FILE(/tmp/foo.txt,-1,,l)=bar)

; Append "bar" to the file with a newline

Set(FILE(/tmp/foo.txt,,,al)=bar)

Syntax

```
FILE(filename,offset,length,options,format)
```

Arguments

- `filename`
- `offset` - Maybe specified as any number. If negative, *offset* specifies the number of bytes back from the end of the file.
- `length` - If specified, will limit the length of the data read to that size. If negative, trims *length* bytes from the end of the file.
- `options`
  - `l` - Line mode: offset and length are assumed to be measured in lines, instead of byte offsets.
  - `a` - In write mode only, the append option is used to append to the end of the file, instead of overwriting the existing file.
  - `d` - In write mode and line mode only, this option does not automatically append a newline string to the end of a value. This is useful for deleting lines, instead of setting them to blank.
- `format` - The *format* parameter may be used to delimit the type of line terminators in line mode.
  - `u` - Unix newline format.
  - `d` - DOS newline format.
  - `m` - Macintosh newline format.

See Also

- Asterisk 12 Function_FILE_COUNT_LINE
- Asterisk 12 Function_FILE_FORMAT

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_FILE_COUNT_LINE

FILE_COUNT_LINE()

Synopsis

Obtains the number of lines of a text file.

Description

Returns the number of lines, or `-1` on error.

Syntax

```
FILE_COUNT_LINE(filename,format)
```

Arguments

- `filename`
- `format` - Format may be one of the following:
  - `u` - Unix newline format.
  - `d` - DOS newline format.
  - `m` - Macintosh newline format.

> **Note**
> If not specified, an attempt will be made to determine the newline format type.

See Also

- Asterisk 12 Function_FILE
- Asterisk 12 Function_FILE_FORMAT

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_FILE_FORMAT

FILE_FORMAT()

Synopsis

Return the newline format of a text file.

Description

Return the line terminator type:

'u' - Unix "\n" format

'd' - DOS "\r\n" format

'm' - Macintosh "\r" format

'x' - Cannot be determined

Syntax

```
FILE_FORMAT(filename)
```

Arguments

- `filename`

See Also

- Asterisk 12 Function_FILE
- Asterisk 12 Function_FILE_COUNT_LINE

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_FILTER

FILTER()

Synopsis

Filter the string to include only the allowed characters

Description

Permits all characters listed in *allowed-chars*, filtering all others outs. In addition to literally listing the characters, you may also use ranges of characters (delimited by a `-`

Hexadecimal characters started with a `\x`(i.e. \x20)

Octal characters started with a `\0` (i.e. \040)

Also `\t`,`\n` and `\r` are recognized.

> **Note**
> If you want the `-` character it needs to be prefixed with a {{}}

Syntax

```
FILTER(allowed-chars,string)
```

Arguments

- `allowed-chars`
- `string`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_FRAME_TRACE

FRAME_TRACE()

Synopsis

View internal ast_frames as they are read and written on a channel.

Description

Examples:

exten => 1,1,Set(FRAME_TRACE(white)=DTMF_BEGIN,DTMF_END); view only DTMF frames.

exten => 1,1,Set(FRAME_TRACE()=DTMF_BEGIN,DTMF_END); view only DTMF frames.

exten => 1,1,Set(FRAME_TRACE(black)=DTMF_BEGIN,DTMF_END); view everything except DTMF frames.

Syntax

```
FRAME_TRACE(filter list type)
```

Arguments

- `filter list type` - A filter can be applied to the trace to limit what frames are viewed. This filter can either be a `white` or `black` list of frame types. When no filter type is present, `white` is used. If no arguments are provided at all, all frames will be output.
  Below are the different types of frames that can be filtered.
    - `DTMF_BEGIN`
    - `DTMF_END`
    - `VOICE`
    - `VIDEO`
    - `CONTROL`
    - `NULL`
    - `IAX`
    - `TEXT`
    - `IMAGE`
    - `HTML`
    - `CNG`
    - `MODEM`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_GLOBAL

GLOBAL()

Synopsis

Gets or sets the global variable specified.

Description

Set or get the value of a global variable specified in *varname*

Syntax

```
GLOBAL(varname)
```

Arguments

- `varname` - Global variable name

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_GROUP

GROUP()

Synopsis

Gets or sets the channel group.

Description

category can be employed for more fine grained group management. Each channel can only be member of exactly one group per category.

Syntax

```
GROUP(category)
```

Arguments

- `category` - Category name.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_GROUP_COUNT

GROUP_COUNT()

Synopsis

Counts the number of channels in the specified group.

Description

Calculates the group count for the specified group, or uses the channel's current group if not specifed (and non-empty).

Syntax

```
GROUP_COUNT(groupname@category)
```

Arguments

- `groupname` - Group name.
- `category` - Category name

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_GROUP_LIST

GROUP_LIST()

Synopsis

Gets a list of the groups set on a channel.

Description

Gets a list of the groups set on a channel.

Syntax

```
GROUP_LIST()
```

Arguments

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_GROUP_MATCH_COUNT

GROUP_MATCH_COUNT()

Synopsis

Counts the number of channels in the groups matching the specified pattern.

Description

Calculates the group count for all groups that match the specified pattern. Note: category matching is applied after matching based on group. Uses standard regular expression matching on both (see regex(7)).

Syntax

```
GROUP_MATCH_COUNT(groupmatch@category)
```

Arguments

- `groupmatch` - A standard regular expression used to match a group name.
- `category` - A standard regular expression used to match a category name.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_HANGUPCAUSE

HANGUPCAUSE()

Synopsis

Gets per-channel hangupcause information from the channel.

Description

Gets technology-specific or translated Asterisk cause code information from the channel for the specified channel that resulted from a dial.

Syntax

```
HANGUPCAUSE(channel,type)
```

Arguments

- `channel` - The name of the channel for which to retreive cause information.
- `type` - Parameter describing which type of information is requested. Types are:
  - `tech` - Technology-specific cause information
  - `ast` - Translated Asterisk cause code

See Also

- Asterisk 12 Function_HANGUPCAUSE_KEYS
- Asterisk 12 Application_HangupCauseClear

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_HANGUPCAUSE_KEYS

HANGUPCAUSE_KEYS()

Synopsis

Gets the list of channels for which hangup causes are available.

Description

Returns a comma-separated list of channel names to be used with the HANGUPCAUSE function.

Syntax

See Also

- Asterisk 12 Function_HANGUPCAUSE
- Asterisk 12 Application_HangupCauseClear

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_HASH

HASH()

Synopsis

Implementation of a dialplan associative array

Description

In two arguments mode, gets and sets values to corresponding keys within a named associative array. The single-argument mode will only work when assigned to from a function defined by func_odbc

Syntax

```
HASH(hashname,hashkey)
```

Arguments

- `hashname`
- `hashkey`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_HASHKEYS

HASHKEYS()

Synopsis

Retrieve the keys of the HASH() function.

Description

Returns a comma-delimited list of the current keys of the associative array defined by the HASH() function. Note that if you iterate over the keys of the result, adding keys during iteration will cause the result of the HASHKEYS() function to change.

Syntax

```
HASHKEYS(hashname)
```

Arguments

- `hashname`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_HINT

HINT()

Synopsis

Get the devices set for a dialplan hint.

Description

The HINT function can be used to retrieve the list of devices that are mapped to a dialplan hint. For example:

NoOp(Hint for Extension 1234 is ${HINT(1234)})

Syntax

```
HINT(extension,options)
```

Arguments

- `extension`
    - `extension`
    - `context`
- `options`
    - n - Retrieve name on the hint instead of list of devices.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_IAXPEER

IAXPEER()

Synopsis

Gets IAX peer information.

Description

Gets information associated with the specified IAX2 peer.

Syntax

```
IAXPEER(peername,item)
```

Arguments

- `peername`
  - `CURRENTCHANNEL` - If *peername* is specified to this value, return the IP address of the endpoint of the current channel
- `item` - If *peername* is specified, valid items are:
  - `ip` - (default) The IP address.
  - `status` - The peer's status (if `qualify=yes`)
  - `mailbox` - The configured mailbox.
  - `context` - The configured context.
  - `expire` - The epoch time of the next expire.
  - `dynamic` - Is it dynamic? (yes/no).
  - `callerid_name` - The configured Caller ID name.
  - `callerid_num` - The configured Caller ID number.
  - `codecs` - The configured codecs.
  - `codecx` - Preferred codec index number *x* (beginning with `0`)

See Also

- Asterisk 12 Function_SIPPEER

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_IAXVAR

IAXVAR()

Synopsis

Sets or retrieves a remote variable.

Description

Gets or sets a variable that is sent to a remote IAX2 peer during call setup.

Syntax

```
IAXVAR(varname)
```

Arguments

- `varname`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_ICONV

ICONV()

Synopsis

Converts charsets of strings.

Description

Converts string from *in-charset* into *out-charset*. For available charsets, use `iconv -l` on your shell command line.

> **Note**
> Due to limitations within the API, ICONV will not currently work with charsets with embedded NULLs. If found, the string will terminate.

Syntax

```
ICONV(in-charset,out-charset,string)
```

Arguments

- `in-charset` - Input charset
- `out-charset` - Output charset
- `string` - String to convert, from *in-charset* to *out-charset*

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_IF

IF()

Synopsis

Check for an expresion.

Description

Returns the data following `?` if true, else the data following `:`

Syntax

```
IF(expresion?retvalue)
```

Arguments

- `expresion`
- `retvalue`
    - `true`
    - `false`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_IFMODULE

IFMODULE()

Synopsis

Checks if an Asterisk module is loaded in memory.

Description

Checks if a module is loaded. Use the full module name as shown by the list in `module list`. Returns `1` if module exists in memory, otherwise `0`

Syntax

```
IFMODULE(modulename.so)
```

Arguments

- `modulename.so` - Module name complete with `.so`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_IFTIME

IFTIME()

Synopsis

Temporal Conditional.

Description

Returns the data following `?` if true, else the data following `:`

Syntax

```
IFTIME(timespec?retvalue)
```

Arguments

- `timespec`
- `retvalue`
    - `true`
    - `false`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_IMPORT

IMPORT()

Synopsis

Retrieve the value of a variable from another channel.

Description

Syntax

```
IMPORT(channel,variable)
```

Arguments

- `channel`
- `variable`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_INC

INC()

Synopsis

Increments the value of a variable, while returning the updated value to the dialplan

Description

Increments the value of a variable, while returning the updated value to the dialplan

Example: INC(MyVAR) - Increments MyVar

Note: INC(${MyVAR}) - Is wrong, as INC expects the variable name, not its value

Syntax

```
INC(variable)
```

Arguments

- `variable` - The variable name to be manipulated, without the braces.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_ISNULL

ISNULL()

Synopsis

Check if a value is NULL.

Description

Returns `1` if NULL or `0` otherwise.

Syntax

```
ISNULL(data)
```

Arguments

- `data`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_JABBER_RECEIVE_res_jabber

JABBER_RECEIVE() - [res_jabber]

Synopsis

Reads XMPP messages.

Description

Receives a text message on the given *account* from the buddy identified by *jid* and returns the contents.

Example: ${JABBER_RECEIVE(asterisk,bob@domain.com)} returns an XMPP message sent from *bob@domain.com* (or nothing in case of a time out), to the *asterisk* XMPP account configured in jabber.conf.

Syntax

```
JABBER_RECEIVE(account,jid,timeout)
```

Arguments

- `account` - The local named account to listen on (specified in jabber.conf)
- `jid` - Jabber ID of the buddy to receive message from. It can be a bare JID (username@domain) or a full JID (username@domain/resource).

- `timeout` - In seconds, defaults to `20`.

See Also

- Asterisk 12 Function_JABBER_STATUS_res_jabber
- Asterisk 12 Application_JabberSend_res_jabber

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_JABBER_RECEIVE_res_xmpp

JABBER_RECEIVE() - [res_xmpp]

Synopsis

Reads XMPP messages.

Description

Receives a text message on the given *account* from the buddy identified by *jid* and returns the contents.

Example: ${JABBER_RECEIVE(asterisk,bob@domain.com)} returns an XMPP message sent from *bob@domain.com* (or nothing in case of a time out), to the *asterisk* XMPP account configured in xmpp.conf.

Syntax

```
JABBER_RECEIVE(account,jid,timeout)
```

Arguments

- `account` - The local named account to listen on (specified in xmpp.conf)
- `jid` - Jabber ID of the buddy to receive message from. It can be a bare JID (username@domain) or a full JID (username@domain/resource).
- `timeout` - In seconds, defaults to `20`.

See Also

- Asterisk 12 Function_JABBER_STATUS_res_xmpp
- Asterisk 12 Application_JabberSend_res_xmpp

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_JABBER_STATUS_res_jabber

JABBER_STATUS() - [res_jabber]

Synopsis

Retrieves a buddy's status.

Description

Retrieves the numeric status associated with the buddy identified by *jid*. If the buddy does not exist in the buddylist, returns 7.

Status will be 1-7.

1=Online, 2=Chatty, 3=Away, 4=XAway, 5=DND, 6=Offline

If not in roster variable will be set to 7.

Example: ${JABBER_STATUS(asterisk,bob@domain.com)} returns 1 if *bob@domain.com* is online. *asterisk* is the associated XMPP account configured in jabber.conf.

Syntax

```
JABBER_STATUS(account,jid)
```

Arguments

- `account` - The local named account to listen on (specified in jabber.conf)
- `jid` - Jabber ID of the buddy to receive message from. It can be a bare JID (username@domain) or a full JID (username@domain/resource).

See Also

- Asterisk 12 Function_JABBER_RECEIVE_res_jabber
- Asterisk 12 Application_JabberSend_res_jabber

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_JABBER_STATUS_res_xmpp

JABBER_STATUS() - [res_xmpp]

Synopsis

Retrieves a buddy's status.

Description

Retrieves the numeric status associated with the buddy identified by *jid*. If the buddy does not exist in the buddylist, returns 7.

Status will be 1-7.

1=Online, 2=Chatty, 3=Away, 4=XAway, 5=DND, 6=Offline

If not in roster variable will be set to 7.

Example: ${JABBER_STATUS(asterisk,bob@domain.com)} returns 1 if *bob@domain.com* is online. *asterisk* is the associated XMPP account configured in xmpp.conf.

Syntax

```
JABBER_STATUS(account,jid)
```

Arguments

- `account` - The local named account to listen on (specified in xmpp.conf)
- `jid` - Jabber ID of the buddy to receive message from. It can be a bare JID (username@domain) or a full JID (username@domain/resource).

See Also

- Asterisk 12 Function_JABBER_RECEIVE_res_xmpp
- Asterisk 12 Application_JabberSend_res_xmpp

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_JITTERBUFFER

JITTERBUFFER()

Synopsis

Add a Jitterbuffer to the Read side of the channel. This dejitters the audio stream before it reaches the Asterisk core. This is a write only function.

Description

max_size: Defaults to 200 ms

Length in milliseconds of buffer.

resync_threshold: Defaults to 1000ms

The length in milliseconds over which a timestamp difference will result in resyncing the jitterbuffer.

target_extra: Defaults to 40ms

This option only affects the adaptive jitterbuffer. It represents the amount time in milliseconds by which the new jitter buffer will pad its size.

Examples:

exten => 1,1,Set(JITTERBUFFER(fixed)=default);Fixed with defaults.

exten => 1,1,Set(JITTERBUFFER(fixed)=200);Fixed with max size 200ms, default resync threshold and target extra.

exten => 1,1,Set(JITTERBUFFER(fixed)=200,1500);Fixed with max size 200ms resync threshold 1500.

exten => 1,1,Set(JITTERBUFFER(adaptive)=default);Adaptive with defaults.

exten => 1,1,Set(JITTERBUFFER(adaptive)=200,,60);Adaptive with max size 200ms, default resync threshold and 40ms target extra.

exten => 1,n,Set(JITTERBUFFER(disabled)=);Remove previously applied jitterbuffer

> **Note**
> If a channel specifies a jitterbuffer due to channel driver configuration and the JITTERBUFFER function has set a jitterbuffer for that channel, the jitterbuffer set by the JITTERBUFFER function will take priority and the jitterbuffer set by the channel configuration will not be applied.

Syntax

```
JITTERBUFFER(jitterbuffer type)
```

Arguments

- `jitterbuffer type` - Jitterbuffer type can be `fixed`, `adaptive`, or `disabled`.
  Used as follows.
  Set(JITTERBUFFER(type)=max_size[,resync_threshold[,target_extra]])
  Set(JITTERBUFFER(type)=default)

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_KEYPADHASH

KEYPADHASH()

Synopsis

Hash the letters in string into equivalent keypad numbers.

Description

Example: ${KEYPADHASH(Les)} returns "537"

Syntax

```
KEYPADHASH(string)
```

Arguments

- `string`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_LEN

LEN()

Synopsis

Return the length of the string given.

Description

Example: ${LEN(example)} returns 7

Syntax

```
LEN(string)
```

Arguments

- `string`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_LISTFILTER

LISTFILTER()

Synopsis

Remove an item from a list, by name.

Description

Remove *value* from the list contained in the *varname* variable, where the list delimiter is specified by the *delim* parameter. This is very useful for removing a single channel name from a list of channels, for example.

Syntax

```
LISTFILTER(varname,delim,value)
```

Arguments

- `varname`
- `delim`
- `value`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_LOCAL

LOCAL()

Synopsis

Manage variables local to the gosub stack frame.

Description

Read and write a variable local to the gosub stack frame, once we Return() it will be lost (or it will go back to whatever value it had before the Gosub()).

Syntax

```
LOCAL(varname)
```

Arguments

- `varname`

See Also

- Asterisk 12 Application_Gosub
- Asterisk 12 Application_GosubIf
- Asterisk 12 Application_Return

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_LOCAL_PEEK

LOCAL_PEEK()

Synopsis

Retrieve variables hidden by the local gosub stack frame.

Description

Read a variable *varname* hidden by *n* levels of gosub stack frames. Note that ${LOCAL_PEEK(0,foo)} is the same as `foo`, since the value of *n* peeks under 0 levels of stack frames; in other words, 0 is the current level. If *n* exceeds the available number of stack frames, then an empty string is returned.

Syntax

```
LOCAL_PEEK(n,varname)
```

Arguments

- `n`

- `varname`

See Also

- Asterisk 12 Application_Gosub
- Asterisk 12 Application_GosubIf
- Asterisk 12 Application_Return

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_LOCK

LOCK()

Synopsis

Attempt to obtain a named mutex.

Description

Attempts to grab a named lock exclusively, and prevents other channels from obtaining the same lock. LOCK will wait for the lock to become available. Returns `1` if the lock was obtained or `0` on error.

> **Note**
> To avoid the possibility of a deadlock, LOCK will only attempt to obtain the lock for 3 seconds if the channel already has another lock.

Syntax

```
LOCK(lockname)
```

Arguments

- `lockname`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_MAILBOX_EXISTS

MAILBOX_EXISTS()

Synopsis

Tell if a mailbox is configured.

Description

> **Note**

DEPRECATED. Use VM_INFO(mailbox[@context],exists) instead.

Returns a boolean of whether the corresponding *mailbox* exists. If *context* is not specified, defaults to the `default` context.

Syntax

```
MAILBOX_EXISTS(mailbox@context)
```

Arguments

- `mailbox`
- `context`

See Also

- Asterisk 12 Function_VM_INFO

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_MASTER_CHANNEL

MASTER_CHANNEL()

Synopsis

Gets or sets variables on the master channel

Description

Allows access to the channel which created the current channel, if any. If the channel is already a master channel, then accesses local channel variables.

Syntax

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_MATH

MATH()

Synopsis

Performs Mathematical Functions.

Description

Performs mathematical functions based on two parameters and an operator. The returned value type is *type*

Example: Set(i=${MATH(123%16,int)}) - sets var i=11

Syntax

```
MATH(expression,type)
```

Arguments

- `expression` - Is of the form: *number1opnumber2* where the possible values for *op* are: +,-,/,*,%,<<,>>,^,AND,OR,XOR,<,>,<=,>=,== (and behave as their C equivalents)
- `type` - Wanted type of result:
  f, float - float(default)
  i, int - integer
  h, hex - hex
  c, char - char

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_MD5

MD5()

Synopsis

Computes an MD5 digest.

Description

Computes an MD5 digest.

Syntax

```
MD5(data)
```

Arguments

- `data`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_MEETME_INFO

MEETME_INFO()

Synopsis

Query a given conference of various properties.

Description

Syntax

```
MEETME_INFO(keyword,confno)
```

Arguments

- `keyword` - Options:
    - `lock` - Boolean of whether the corresponding conference is locked.
    - `parties` - Number of parties in a given conference
    - `activity` - Duration of conference in seconds.
    - `dynamic` - Boolean of whether the corresponding conference is dynamic.
- `confno` - Conference number to retrieve information from.

See Also

- Asterisk 12 Application_MeetMe
- Asterisk 12 Application_MeetMeCount
- Asterisk 12 Application_MeetMeAdmin
- Asterisk 12 Application_MeetMeChannelAdmin

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_MESSAGE

MESSAGE()

Synopsis

Create a message or read fields from a message.

Description

This function will read from or write a value to a text message. It is used both to read the data out of an incoming message, as well as modify or create a message that will be sent outbound.

Syntax

```
MESSAGE(argument)
```

Arguments

- `argument` - Field of the message to get or set.
    - `to` - Read-only. The destination of the message. When processing an incoming message, this will be set to the destination listed as the recipient of the message that was received by Asterisk.
    - `from` - Read-only. The source of the message. When processing an incoming message, this will be set to the source of the message.
    - `custom_data` - Write-only. Mark or unmark all message headers for an outgoing message. The following values can be set:
        - `mark_all_outbound` - Mark all headers for an outgoing message.
        - `clear_all_outbound` - Unmark all headers for an outgoing message.
    - `body` - Read/Write. The message body. When processing an incoming message, this includes the body of the message that Asterisk received. When MessageSend() is executed, the contents of this field are used as the body of the outgoing message. The body will always be UTF-8.

See Also

- Asterisk 12 Application_MessageSend

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_MESSAGE_DATA

MESSAGE_DATA()

Synopsis

Read or write custom data attached to a message.

Description

This function will read from or write a value to a text message. It is used both to read the data out of an incoming message, as well as modify a message that will be sent outbound.

> **Note**
> If you want to set an outbound message to carry data in the current message, do Set(MESSAGE_DATA(*key*)=${MESSAGE_DATA(*key*)}).

Syntax

```
MESSAGE_DATA(argument)
```

Arguments

- `argument` - Field of the message to get or set.

See Also

- Asterisk 12 Application_MessageSend

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_MINIVMACCOUNT

MINIVMACCOUNT()

Synopsis

Gets MiniVoicemail account information.

Description

Syntax

```
MINIVMACCOUNT(account:item)
```

Arguments

- `account`

- `item` - Valid items are:
    - `path` - Path to account mailbox (if account exists, otherwise temporary mailbox).
    - `hasaccount` - 1 is static Minivm account exists, 0 otherwise.
    - `fullname` - Full name of account owner.
    - `email` - Email address used for account.
    - `etemplate` - Email template for account (default template if none is configured).
    - `ptemplate` - Pager template for account (default template if none is configured).
    - `accountcode` - Account code for the voicemail account.
    - `pincode` - Pin code for voicemail account.
    - `timezone` - Time zone for voicemail account.
    - `language` - Language for voicemail account.
    - `<channel variable name>` - Channel variable value (set in configuration for account).

See Also

- Asterisk 12 Application_MinivmRecord
- Asterisk 12 Application_MinivmGreet
- Asterisk 12 Application_MinivmNotify
- Asterisk 12 Application_MinivmDelete
- Asterisk 12 Application_MinivmAccMess
- Asterisk 12 Application_MinivmMWI
- Asterisk 12 Function_MINIVMCOUNTER

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_MINIVMCOUNTER

MINIVMCOUNTER()

Synopsis

Reads or sets counters for MiniVoicemail message.

Description

The operation is atomic and the counter is locked while changing the value. The counters are stored as text files in the minivm account directories. It might be better to use realtime functions if you are using a database to operate your Asterisk.

Syntax

```
MINIVMCOUNTER(account:name:operand)
```

Arguments

- `account` - If account is given and it exists, the counter is specific for the account.
  If account is a domain and the domain directory exists, counters are specific for a domain.
- `name` - The name of the counter is a string, up to 10 characters.
- `operand` - The counters never goes below zero. Valid operands for changing the value of a counter when assigning a value are:
    - `i` - Increment by value.
    - `d` - Decrement by value.
    - `s` - Set to value.

See Also

- Asterisk 12 Application_MinivmRecord

- Asterisk 12 Application_MinivmGreet
- Asterisk 12 Application_MinivmNotify
- Asterisk 12 Application_MinivmDelete
- Asterisk 12 Application_MinivmAccMess
- Asterisk 12 Application_MinivmMWI
- Asterisk 12 Function_MINIVMACCOUNT

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_MUTEAUDIO

MUTEAUDIO()

Synopsis

Muting audio streams in the channel

Description

The MUTEAUDIO function can be used to mute inbound (to the PBX) or outbound audio in a call.

Examples:

MUTEAUDIO(in)=on

MUTEAUDIO(in)=off

Syntax

```
MUTEAUDIO(direction)
```

Arguments

- `direction` - Must be one of
  - `in` - Inbound stream (to the PBX)
  - `out` - Outbound stream (from the PBX)
  - `all` - Both streams

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_ODBC

ODBC()

Synopsis

Controls ODBC transaction properties.

Description

The ODBC() function allows setting several properties to influence how a connected database processes transactions.

Syntax

```
ODBC(property[,argument])
```

Arguments

- `property`
  - `transaction` - Gets or sets the active transaction ID. If set, and the transaction ID does not exist and a *database name* is specified as an argument, it will be created.
  - `forcecommit` - Controls whether a transaction will be automatically committed when the channel hangs up. Defaults to false. If a *transaction ID* is specified in the optional argument, the property will be applied to that ID, otherwise to the current active ID.
  - `isolation` - Controls the data isolation on uncommitted transactions. May be one of the following: `read_committed`, `read_uncommitted`, `repeatable_read`, or `serializable`. Defaults to the database setting in `res_odbc.conf` or `read_committed` if not specified. If a *transaction ID* is specified as an optional argument, it will be applied to that ID, otherwise the current active ID.
- `argument`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_ODBC_FETCH

ODBC_FETCH()

Synopsis

Fetch a row from a multirow query.

Description

For queries which are marked as mode=multirow, the original query returns a *result-id* from which results may be fetched. This function implements the actual fetch of the results.

This also sets `ODBC_FETCH_STATUS`.

- `ODBC_FETCH_STATUS`
  - SUCESS - If rows are available.
  - FAILURE - If no rows are available.

Syntax

```
ODBC_FETCH(result-id)
```

Arguments

- `result-id`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_PASSTHRU

PASSTHRU()

Synopsis

Pass the given argument back as a value.

Description

Literally returns the given *string*. The intent is to permit other dialplan functions which take a variable name as an argument to be able to take a literal string, instead.

> **Note**
> The functions which take a variable name need to be passed var and not ${var}. Similarly, use PASSTHRU() and not ${PASSTHRU()}.

Example: ${CHANNEL} contains SIP/321-1

${CUT(PASSTHRU(${CUT(CHANNEL,-,1)}),/,2)}) will return 321

Syntax

```
PASSTHRU([string])
```

Arguments

- `string`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_PITCH_SHIFT

PITCH_SHIFT()

Synopsis

Pitch shift both tx and rx audio streams on a channel.

Description

Examples:

exten => 1,1,Set(PITCH_SHIFT(tx)=highest); raises pitch an octave

exten => 1,1,Set(PITCH_SHIFT(rx)=higher) ; raises pitch more

exten => 1,1,Set(PITCH_SHIFT(both)=high) ; raises pitch

exten => 1,1,Set(PITCH_SHIFT(rx)=low) ; lowers pitch

exten => 1,1,Set(PITCH_SHIFT(tx)=lower) ; lowers pitch more exten => 1,1,Set(PITCH_SHIFT(both)=lowest) ; lowers pitch an octave

exten => 1,1,Set(PITCH_SHIFT(rx)=0.8) ; lowers pitch

exten => 1,1,Set(PITCH_SHIFT(tx)=1.5) ; raises pitch

Syntax

```
PITCH_SHIFT(channel direction)
```

Arguments

- `channel direction` - Direction can be either `rx`, `tx`, or `both`. The direction can either be set to a valid floating point number between 0.1 and 4.0 or one of the enum values listed below. A value of 1.0 has no effect. Greater than 1 raises the pitch. Lower than 1 lowers the pitch.
  The pitch amount can also be set by the following values
  - `highest`
  - `higher`
  - `high`
  - `low`
  - `lower`
  - `lowest`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_PJSIP_DIAL_CONTACTS

PJSIP_DIAL_CONTACTS()

Synopsis

Return a dial string for dialing all contacts on an AOR.

Description

Returns a properly formatted dial string for dialing all contacts on an AOR.

Syntax

```
PJSIP_DIAL_CONTACTS(endpoint[,aor[,request_user]])
```

Arguments

- `endpoint` - Name of the endpoint
- `aor` - Name of an AOR to use, if not specified the configured AORs on the endpoint are used
- `request_user` - Optional request user to use in the request URI

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_PJSIP_MEDIA_OFFER

PJSIP_MEDIA_OFFER()

Synopsis

Media and codec offerings to be set on an outbound SIP channel prior to dialing.

Description

Returns the codecs offered based upon the media choice

Syntax

```
PJSIP_MEDIA_OFFER(media)
```

Arguments

- `media` - types of media offered

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_POP

POP()

Synopsis

Removes and returns the last item off of a variable containing delimited text

Description

Example:

exten => s,1,Set(array=one,two,three)

exten => s,n,While($["${SET(var=${POP(array)})}" != ""])

exten => s,n,NoOp(var is ${var})

exten => s,n,EndWhile

This would iterate over each value in array, right to left, and would result in NoOp(var is three), NoOp(var is two), and NoOp(var is one) being executed.

Syntax

```
POP(varname[,delimiter])
```

Arguments

- `varname`
- `delimiter`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_PP_EACH_EXTENSION

PP_EACH_EXTENSION()

Synopsis

Execute specified template for each extension.

Description

Output the specified template for each extension associated with the specified MAC address.

Syntax

```
PP_EACH_EXTENSION(mac,template)
```

Arguments

- `mac`
- `template`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_PP_EACH_USER

PP_EACH_USER()

Synopsis

Generate a string for each phoneprov user.

Description

Pass in a string, with phoneprov variables you want substituted in the format of %{VARNAME}, and you will get the string rendered for each user in phoneprov excluding ones with MAC address *exclude_mac*. Probably not useful outside of res_phoneprov.

Example: ${PP_EACH_USER(<item><fn>%{DISPLAY_NAME}</fn></item>|${MAC})

Syntax

```
PP_EACH_USER(string,exclude_mac)
```

Arguments

- `string`
- `exclude_mac`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_PRESENCE_STATE

PRESENCE_STATE()

Synopsis

Get or Set a presence state.

Description

The PRESENCE_STATE function can be used to retrieve the presence from any presence provider. For example:

NoOp(SIP/mypeer has presence ${PRESENCE_STATE(SIP/mypeer,value)})

NoOp(Conference number 1234 has presence message ${PRESENCE_STATE(MeetMe:1234,message)})

The PRESENCE_STATE function can also be used to set custom presence state from the dialplan. The `CustomPresence:` prefix must be used. For example:

Set(PRESENCE_STATE(CustomPresence:lamp1)=away,temporary,Out to lunch)

Set(PRESENCE_STATE(CustomPresence:lamp2)=dnd,,Trying to get work done)

Set(PRESENCE_STATE(CustomPresence:lamp3)=xa,T24gdmFjYXRpb24=,,e)

Set(BASE64_LAMP3_PRESENCE=${PRESENCE_STATE(CustomPresence:lamp3,subtype,e)})

You can subscribe to the status of a custom presence state using a hint in the dialplan:

exten => 1234,hint,CustomPresence:lamp1

The possible values for both uses of this function are:

not_set | unavailable | available | away | xa | chat | dnd

Syntax

```
PRESENCE_STATE(provider,field[,options])
```

Arguments

- `provider` - The provider of the presence, such as `CustomPresence`

- `field` - Which field of the presence state information is wanted.
    - `value` - The current presence, such as `away`
    - `subtype` - Further information about the current presence
    - `message` - A custom message that may indicate further details about the presence
- `options`
    - `e` - On Write - Use this option when the subtype and message provided are Base64 encoded. On Read - Retrieves message/subtype in Base64 encoded form.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_PUSH

PUSH()

Synopsis

Appends one or more values to the end of a variable containing delimited text

Description

Example: Set(PUSH(array)=one,two,three) would append one, two, and three to the end of the values stored in the variable "array".

Syntax

```
PUSH(varname[,delimiter])
```

Arguments

- `varname`
- `delimiter`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_QUEUE_EXISTS

QUEUE_EXISTS()

Synopsis

Check if a named queue exists on this server

Description

Returns 1 if the specified queue exists, 0 if it does not

Syntax

```
QUEUE_EXISTS(queuename)
```

Arguments

- `queuename`

See Also

- Asterisk 12 Application_Queue
- Asterisk 12 Application_QueueLog
- Asterisk 12 Application_AddQueueMember
- Asterisk 12 Application_RemoveQueueMember
- Asterisk 12 Application_PauseQueueMember
- Asterisk 12 Application_UnpauseQueueMember
- Asterisk 12 Function_QUEUE_VARIABLES
- Asterisk 12 Function_QUEUE_MEMBER
- Asterisk 12 Function_QUEUE_MEMBER_COUNT
- Asterisk 12 Function_QUEUE_EXISTS
- Asterisk 12 Function_QUEUE_WAITING_COUNT
- Asterisk 12 Function_QUEUE_MEMBER_LIST
- Asterisk 12 Function_QUEUE_MEMBER_PENALTY

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_QUEUE_MEMBER

QUEUE_MEMBER()

Synopsis

Count number of members answering a queue.

Description

Allows access to queue counts [R] and member information [R/W].

queuename is required for all operations *interface* is required for all member operations.

Syntax

```
QUEUE_MEMBER(queuename,option[,interface])
```

Arguments

- `queuename`
- `option`
    - `logged` - Returns the number of logged-in members for the specified queue.
    - `free` - Returns the number of logged-in members for the specified queue that either can take calls or are currently wrapping up after a previous call.
    - `ready` - Returns the number of logged-in members for the specified queue that are immediately available to answer a call.
    - `count` - Returns the total number of members for the specified queue.
    - `penalty` - Gets or sets queue member penalty.
    - `paused` - Gets or sets queue member paused status.
    - `ringinuse` - Gets or sets queue member ringinuse.

* `interface`

See Also

* Asterisk 12 Application_Queue
* Asterisk 12 Application_QueueLog
* Asterisk 12 Application_AddQueueMember
* Asterisk 12 Application_RemoveQueueMember
* Asterisk 12 Application_PauseQueueMember
* Asterisk 12 Application_UnpauseQueueMember
* Asterisk 12 Function_QUEUE_VARIABLES
* Asterisk 12 Function_QUEUE_MEMBER
* Asterisk 12 Function_QUEUE_MEMBER_COUNT
* Asterisk 12 Function_QUEUE_EXISTS
* Asterisk 12 Function_QUEUE_WAITING_COUNT
* Asterisk 12 Function_QUEUE_MEMBER_LIST
* Asterisk 12 Function_QUEUE_MEMBER_PENALTY

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_QUEUE_MEMBER_COUNT

QUEUE_MEMBER_COUNT()

Synopsis

Count number of members answering a queue.

Description

Returns the number of members currently associated with the specified *queuename*.

Warning
This function has been deprecated in favor of the `QUEUE_MEMBER()` function

Syntax

```
QUEUE_MEMBER_COUNT(queuename)
```

Arguments

* `queuename`

See Also

* Asterisk 12 Application_Queue
* Asterisk 12 Application_QueueLog
* Asterisk 12 Application_AddQueueMember
* Asterisk 12 Application_RemoveQueueMember
* Asterisk 12 Application_PauseQueueMember
* Asterisk 12 Application_UnpauseQueueMember
* Asterisk 12 Function_QUEUE_VARIABLES
* Asterisk 12 Function_QUEUE_MEMBER
* Asterisk 12 Function_QUEUE_MEMBER_COUNT
* Asterisk 12 Function_QUEUE_EXISTS
* Asterisk 12 Function_QUEUE_WAITING_COUNT

* Asterisk 12 Function_QUEUE_MEMBER_LIST
* Asterisk 12 Function_QUEUE_MEMBER_PENALTY

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_QUEUE_MEMBER_LIST

QUEUE_MEMBER_LIST()

Synopsis

Returns a list of interfaces on a queue.

Description

Returns a comma-separated list of members associated with the specified *queuename*.

Syntax

```
QUEUE_MEMBER_LIST(queuename)
```

Arguments

* `queuename`

See Also

* Asterisk 12 Application_Queue
* Asterisk 12 Application_QueueLog
* Asterisk 12 Application_AddQueueMember
* Asterisk 12 Application_RemoveQueueMember
* Asterisk 12 Application_PauseQueueMember
* Asterisk 12 Application_UnpauseQueueMember
* Asterisk 12 Function_QUEUE_VARIABLES
* Asterisk 12 Function_QUEUE_MEMBER
* Asterisk 12 Function_QUEUE_MEMBER_COUNT
* Asterisk 12 Function_QUEUE_EXISTS
* Asterisk 12 Function_QUEUE_WAITING_COUNT
* Asterisk 12 Function_QUEUE_MEMBER_LIST
* Asterisk 12 Function_QUEUE_MEMBER_PENALTY

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_QUEUE_MEMBER_PENALTY

QUEUE_MEMBER_PENALTY()

Synopsis

Gets or sets queue members penalty.

Description

Gets or sets queue members penalty.

Warning
This function has been deprecated in favor of the `QUEUE_MEMBER()` function

Syntax

```
QUEUE_MEMBER_PENALTY(queuename,interface)
```

Arguments

- `queuename`
- `interface`

See Also

- Asterisk 12 Application_Queue
- Asterisk 12 Application_QueueLog
- Asterisk 12 Application_AddQueueMember
- Asterisk 12 Application_RemoveQueueMember
- Asterisk 12 Application_PauseQueueMember
- Asterisk 12 Application_UnpauseQueueMember
- Asterisk 12 Function_QUEUE_VARIABLES
- Asterisk 12 Function_QUEUE_MEMBER
- Asterisk 12 Function_QUEUE_MEMBER_COUNT
- Asterisk 12 Function_QUEUE_EXISTS
- Asterisk 12 Function_QUEUE_WAITING_COUNT
- Asterisk 12 Function_QUEUE_MEMBER_LIST
- Asterisk 12 Function_QUEUE_MEMBER_PENALTY

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_QUEUE_VARIABLES

QUEUE_VARIABLES()

Synopsis

Return Queue information in variables.

Description

Makes the following queue variables available.

Returns `0` if queue is found and setqueuevar is defined, `-1` otherwise.

Syntax

```
QUEUE_VARIABLES(queuename)
```

Arguments

- `queuename`
    - `QUEUEMAX` - Maxmimum number of calls allowed.
    - `QUEUESTRATEGY` - The strategy of the queue.

- `QUEUECALLS` - Number of calls currently in the queue.
- `QUEUEHOLDTIME` - Current average hold time.
- `QUEUECOMPLETED` - Number of completed calls for the queue.
- `QUEUEABANDONED` - Number of abandoned calls.
- `QUEUESRVLEVEL` - Queue service level.
- `QUEUESRVLEVELPERF` - Current service level performance.

See Also

- Asterisk 12 Application_Queue
- Asterisk 12 Application_QueueLog
- Asterisk 12 Application_AddQueueMember
- Asterisk 12 Application_RemoveQueueMember
- Asterisk 12 Application_PauseQueueMember
- Asterisk 12 Application_UnpauseQueueMember
- Asterisk 12 Function_QUEUE_VARIABLES
- Asterisk 12 Function_QUEUE_MEMBER
- Asterisk 12 Function_QUEUE_MEMBER_COUNT
- Asterisk 12 Function_QUEUE_EXISTS
- Asterisk 12 Function_QUEUE_WAITING_COUNT
- Asterisk 12 Function_QUEUE_MEMBER_LIST
- Asterisk 12 Function_QUEUE_MEMBER_PENALTY

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_QUEUE_WAITING_COUNT

QUEUE_WAITING_COUNT()

Synopsis

Count number of calls currently waiting in a queue.

Description

Returns the number of callers currently waiting in the specified *queuename*.

Syntax

```
QUEUE_WAITING_COUNT(queuename)
```

Arguments

- `queuename`

See Also

- Asterisk 12 Application_Queue
- Asterisk 12 Application_QueueLog
- Asterisk 12 Application_AddQueueMember
- Asterisk 12 Application_RemoveQueueMember
- Asterisk 12 Application_PauseQueueMember
- Asterisk 12 Application_UnpauseQueueMember
- Asterisk 12 Function_QUEUE_VARIABLES
- Asterisk 12 Function_QUEUE_MEMBER
- Asterisk 12 Function_QUEUE_MEMBER_COUNT
- Asterisk 12 Function_QUEUE_EXISTS

- Asterisk 12 Function_QUEUE_WAITING_COUNT
- Asterisk 12 Function_QUEUE_MEMBER_LIST
- Asterisk 12 Function_QUEUE_MEMBER_PENALTY

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_QUOTE

QUOTE()

Synopsis

Quotes a given string, escaping embedded quotes as necessary

Description

Example: ${QUOTE(ab"c"de)} will return "abcde"

Syntax

```
QUOTE(string)
```

Arguments

- `string`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_RAND

RAND()

Synopsis

Choose a random number in a range.

Description

Choose a random number between *min* and *max*. *min* defaults to `0`, if not specified, while *max* defaults to `RAND_MAX` (2147483647 on many systems).

Example: Set(junky=${RAND(1,8)}); Sets junky to a random number between 1 and 8, inclusive.

Syntax

```
RAND(min,max)
```

Arguments

- `min`
- `max`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_REALTIME

REALTIME()

Synopsis

RealTime Read/Write Functions.

Description

This function will read or write values from/to a RealTime repository. REALTIME(....) will read names/values from the repository, and REALTIME(....)= will write a new value/field to the repository. On a read, this function returns a delimited text string. The name/value pairs are delimited by *delim1*, and the name and value are delimited between each other with delim2. If there is no match, NULL will be returned by the function. On a write, this function will always return NULL.

Syntax

```
REALTIME(family,fieldmatch,matchvalue,delim1|field,delim2)
```

Arguments

- `family`
- `fieldmatch`
- `matchvalue`
- `delim1|field` - Use *delim1* with *delim2* on read and *field* without *delim2* on write
  If we are reading and *delim1* is not specified, defaults to `,`
- `delim2` - Parameter only used when reading, if not specified defaults to =

See Also

- Asterisk 12 Function_REALTIME_STORE
- Asterisk 12 Function_REALTIME_DESTROY
- Asterisk 12 Function_REALTIME_FIELD
- Asterisk 12 Function_REALTIME_HASH

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_REALTIME_DESTROY

REALTIME_DESTROY()

Synopsis

RealTime Destroy Function.

Description

This function acts in the same way as REALTIME(....) does, except that it destroys the matched record in the RT engine.

Syntax

```
REALTIME_DESTROY(family,fieldmatch,matchvalue,delim1,delim2)
```

Arguments

- `family`
- `fieldmatch`
- `matchvalue`
- `delim1`
- `delim2`

See Also

- Asterisk 12 Function_REALTIME
- Asterisk 12 Function_REALTIME_STORE
- Asterisk 12 Function_REALTIME_FIELD
- Asterisk 12 Function_REALTIME_HASH

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_REALTIME_FIELD

REALTIME_FIELD()

Synopsis

RealTime query function.

Description

This function retrieves a single item, *fieldname* from the RT engine, where *fieldmatch* contains the value *matchvalue*. When written to, the REALTIME_FIELD() function performs identically to the REALTIME() function.

Syntax

```
REALTIME_FIELD(family,fieldmatch,matchvalue,fieldname)
```

Arguments

- `family`
- `fieldmatch`
- `matchvalue`
- `fieldname`

See Also

- Asterisk 12 Function_REALTIME
- Asterisk 12 Function_REALTIME_STORE
- Asterisk 12 Function_REALTIME_DESTROY

* Asterisk 12 Function_REALTIME_HASH

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_REALTIME_HASH

REALTIME_HASH()

Synopsis

RealTime query function.

Description

This function retrieves a single record from the RT engine, where *fieldmatch* contains the value *matchvalue* and formats the output suitably, such that it can be assigned to the HASH() function. The HASH() function then provides a suitable method for retrieving each field value of the record.

Syntax

```
REALTIME_HASH(family,fieldmatch,matchvalue)
```

Arguments

* `family`
* `fieldmatch`
* `matchvalue`

See Also

* Asterisk 12 Function_REALTIME
* Asterisk 12 Function_REALTIME_STORE
* Asterisk 12 Function_REALTIME_DESTROY
* Asterisk 12 Function_REALTIME_FIELD

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_REALTIME_STORE

REALTIME_STORE()

Synopsis

RealTime Store Function.

Description

This function will insert a new set of values into the RealTime repository. If RT engine provides an unique ID of the stored record, REALTIME_STORE(...)=.. creates channel variable named RTSTOREID, which contains value of unique ID. Currently, a maximum of 30 field/value pairs is supported.

Syntax

```
REALTIME_STORE(family,field1,fieldN[,...],field30)
```

Arguments

- `family`
- `field1`
- `fieldN`
- `field30`

See Also

- Asterisk 12 Function_REALTIME
- Asterisk 12 Function_REALTIME_DESTROY
- Asterisk 12 Function_REALTIME_FIELD
- Asterisk 12 Function_REALTIME_HASH

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_REDIRECTING

REDIRECTING()

Synopsis

Gets or sets Redirecting data on the channel.

Description

Gets or sets Redirecting data on the channel.

The allowable values for the *reason* and *orig-reason* fields are the following:

- `unknown` - Unknown
- `cfb` - Call Forwarding Busy
- `cfnr` - Call Forwarding No Reply
- `unavailable` - Callee is Unavailable
- `time_of_day` - Time of Day
- `dnd` - Do Not Disturb
- `deflection` - Call Deflection
- `follow_me` - Follow Me
- `out_of_order` - Called DTE Out-Of-Order
- `away` - Callee is Away
- `cf_dte` - Call Forwarding By The Called DTE
- `cfu` - Call Forwarding Unconditional
  The allowable values for the *xxx-name-charset* field are the following:

- `unknown` - Unknown
- `iso8859-1` - ISO8859-1
- `withdrawn` - Withdrawn
- `iso8859-2` - ISO8859-2
- `iso8859-3` - ISO8859-3
- `iso8859-4` - ISO8859-4
- `iso8859-5` - ISO8859-5
- `iso8859-7` - ISO8859-7
- `bmp` - ISO10646 Bmp String
- `utf8` - ISO10646 UTF-8 String

Syntax

```
REDIRECTING(datatype,i)
```

Arguments

- `datatype` - The allowable datatypes are:
    - `orig-all`
    - `orig-name`
    - `orig-name-valid`
    - `orig-name-charset`
    - `orig-name-pres`
    - `orig-num`
    - `orig-num-valid`
    - `orig-num-plan`
    - `orig-num-pres`
    - `orig-subaddr`
    - `orig-subaddr-valid`
    - `orig-subaddr-type`
    - `orig-subaddr-odd`
    - `orig-tag`
    - `orig-reason`
    - `from-all`
    - `from-name`
    - `from-name-valid`
    - `from-name-charset`
    - `from-name-pres`
    - `from-num`
    - `from-num-valid`
    - `from-num-plan`
    - `from-num-pres`
    - `from-subaddr`
    - `from-subaddr-valid`
    - `from-subaddr-type`
    - `from-subaddr-odd`
    - `from-tag`
    - `to-all`
    - `to-name`
    - `to-name-valid`
    - `to-name-charset`
    - `to-name-pres`
    - `to-num`
    - `to-num-valid`
    - `to-num-plan`
    - `to-num-pres`
    - `to-subaddr`
    - `to-subaddr-valid`
    - `to-subaddr-type`
    - `to-subaddr-odd`
    - `to-tag`
    - `priv-orig-all`
    - `priv-orig-name`
    - `priv-orig-name-valid`
    - `priv-orig-name-charset`
    - `priv-orig-name-pres`
    - `priv-orig-num`
    - `priv-orig-num-valid`
    - `priv-orig-num-plan`

- `priv-orig-num-pres`
- `priv-orig-subaddr`
- `priv-orig-subaddr-valid`
- `priv-orig-subaddr-type`
- `priv-orig-subaddr-odd`
- `priv-orig-tag`
- `priv-from-all`
- `priv-from-name`
- `priv-from-name-valid`
- `priv-from-name-charset`
- `priv-from-name-pres`
- `priv-from-num`
- `priv-from-num-valid`
- `priv-from-num-plan`
- `priv-from-num-pres`
- `priv-from-subaddr`
- `priv-from-subaddr-valid`
- `priv-from-subaddr-type`
- `priv-from-subaddr-odd`
- `priv-from-tag`
- `priv-to-all`
- `priv-to-name`
- `priv-to-name-valid`
- `priv-to-name-charset`
- `priv-to-name-pres`
- `priv-to-num`
- `priv-to-num-valid`
- `priv-to-num-plan`
- `priv-to-num-pres`
- `priv-to-subaddr`
- `priv-to-subaddr-valid`
- `priv-to-subaddr-type`
- `priv-to-subaddr-odd`
- `priv-to-tag`
- `reason`
- `count`

- `i` - If set, this will prevent the channel from sending out protocol messages because of the value being set

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_REGEX

REGEX()

Synopsis

Check string against a regular expression.

Description

Return `1` on regular expression match or `0` otherwise

Please note that the space following the double quotes separating the regex from the data is optional and if present, is skipped. If a space is desired at the beginning of the data, then put two spaces there; the second will not be skipped.

Syntax

```
REGEX("regular expression" string)
```

Arguments

- `"regular expression"`
- `string`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_REPLACE

REPLACE()

Synopsis

Replace a set of characters in a given string with another character.

Description

Iterates through a string replacing all the *find-chars* with *replace-char*. *replace-char* may be either empty or contain one character. If empty, all *find-chars* will be deleted from the output.

> **Note**
> The replacement only occurs in the output. The original variable is not altered.

Syntax

```
REPLACE(varname,find-chars[,replace-char])
```

Arguments

- `varname`
- `find-chars`
- `replace-char`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_SET

SET()

Synopsis

SET assigns a value to a channel variable.

Description

Syntax

```
SET(varname=value)
```

Arguments

- `varname`
- `value`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_SHA1

SHA1()

Synopsis

Computes a SHA1 digest.

Description

Generate a SHA1 digest via the SHA1 algorythm.

Example: Set(sha1hash=${SHA1(junky)})

Sets the asterisk variable sha1hash to the string `60fa5675b9303eb62f99a9cd47f9f5837d18f9a0` which is known as his hash

Syntax

```
SHA1(data)
```

Arguments

- `data` - Input string

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_SHARED

SHARED()

Synopsis

Gets or sets the shared variable specified.

Description

Implements a shared variable area, in which you may share variables between channels.

The variables used in this space are separate from the general namespace of the channel and thus `SHARED(foo)` and `foo` represent two completely different variables, despite sharing the same name.

Finally, realize that there is an inherent race between channels operating at the same time, fiddling with each others' internal variables, which is why this special variable namespace exists; it is to remind you that variables in the SHARED namespace may change at any time, without warning. You should therefore take special care to ensure that when using the SHARED namespace, you retrieve the variable and store it in a regular channel variable before using it in a set of calculations (or you might be surprised by the result).

Syntax

```
SHARED(varname,channel)
```

Arguments

- `varname` - Variable name
- `channel` - If not specified will default to current channel. It is the complete channel name: `SIP/12-abcd1234` or the prefix only `SIP/12` .

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_SHELL

SHELL()

Synopsis

Executes a command using the system shell and captures its output.

Description

Collects the output generated by a command executed by the system shell

Example: `Set(foo=${SHELL(echo \bar)})`

> **Note**
> The command supplied to this function will be executed by the system's shell, typically specified in the SHELL environment variable. There are many different system shells available with somewhat different behaviors, so the output generated by this function may vary between platforms.

Syntax

```
SHELL(command)
```

Arguments

- `command` - The command that the shell should execute.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_SHIFT

SHIFT()

Synopsis

Removes and returns the first item off of a variable containing delimited text

Description

Example:

exten => s,1,Set(array=one,two,three)

exten => s,n,While($["${SET(var=${SHIFT(array)})}" != ""])

exten => s,n,NoOp(var is ${var})

exten => s,n,EndWhile

This would iterate over each value in array, left to right, and would result in NoOp(var is one), NoOp(var is two), and NoOp(var is three) being executed.

Syntax

```
SHIFT(varname[,delimiter])
```

Arguments

- `varname`
- `delimiter`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_SIP_HEADER

SIP_HEADER()

Synopsis

Gets the specified SIP header from an incoming INVITE message.

Description

Since there are several headers (such as Via) which can occur multiple times, SIP_HEADER takes an optional second argument to specify which header with that name to retrieve. Headers start at offset `1`.

Please observe that contents of the SDP (an attachment to the SIP request) can't be accessed with this function.

Syntax

```
SIP_HEADER(name,number)
```

Arguments

- `name`
- `number` - If not specified, defaults to `1`.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_SIPCHANINFO

SIPCHANINFO()

Synopsis

Gets the specified SIP parameter from the current channel.

Description

Syntax

```
SIPCHANINFO(item)
```

Arguments

- `item`
    - `peerip` - The IP address of the peer.
    - `recvip` - The source IP address of the peer.
    - `from` - The SIP URI from the `From:` header.
    - `uri` - The SIP URI from the `Contact:` header.
    - `useragent` - The Useragent header used by the peer.
    - `peername` - The name of the peer.
    - `t38passthrough` - 1 if T38 is offered or enabled in this channel, otherwise 0.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_SIPPEER

SIPPEER()

Synopsis

Gets SIP peer information.

Description

Syntax

```
SIPPEER(peername,item)
```

Arguments

- `peername`
- `item`
    - `ip` - (default) The IP address.
    - `port` - The port number.
    - `mailbox` - The configured mailbox.
    - `context` - The configured context.
    - `expire` - The epoch time of the next expire.
    - `dynamic` - Is it dynamic? (yes/no).
    - `callerid_name` - The configured Caller ID name.
    - `callerid_num` - The configured Caller ID number.
    - `callgroup` - The configured Callgroup.
    - `pickupgroup` - The configured Pickupgroup.
    - `namedcallgroup` - The configured Named Callgroup.
    - `namedpickupgroup` - The configured Named Pickupgroup.
    - `codecs` - The configured codecs.
    - `status` - Status (if qualify=yes).
    - `regexten` - Extension activated at registration.
    - `limit` - Call limit (call-limit).
    - `busylevel` - Configured call level for signalling busy.
    - `curcalls` - Current amount of calls. Only available if call-limit is set.
    - `language` - Default language for peer.
    - `accountcode` - Account code for this peer.
    - `useragent` - Current user agent header used by peer.
    - `maxforwards` - The value used for SIP loop prevention in outbound requests
    - `chanvarname` - A channel variable configured with setvar for this peer.
    - `codecx` - Preferred codec index number *x* (beginning with zero).

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_SMDI_MSG

SMDI_MSG()

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.

Syntax

```
SMDI_MSG(message_id,component)
```

Arguments

- `message_id`
- `component` - Valid message components are:
    - `number` - The message desk number
    - `terminal` - The message desk terminal
    - `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:
      Options:
        - `D` - Direct Calls
        - `A` - Forward All Calls
        - `B` - Forward Busy Calls
        - `N` - Forward No Answer Calls

See Also

- Asterisk 12 Function_SMDI_MSG_RETRIEVE

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_SMDI_MSG_RETRIEVE

SMDI_MSG_RETRIEVE()

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.

Syntax

```
SMDI_MSG_RETRIEVE(smdi port,search key,timeout,options)
```

Arguments

- `smdi port`
- `search key`
- `timeout`
- `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.

See Also

- Asterisk 12 Function_SMDI_MSG

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_SORT

SORT()

Synopsis

Sorts a list of key/vals into a list of keys, based upon the vals.

Description

Takes a comma-separated list of keys and values, each separated by a colon, and returns a comma-separated list of the keys, sorted by their values. Values will be evaluated as floating-point numbers.

Syntax

```
SORT(keyval,keyvaln[,...])
```

Arguments

- `keyval`
    - `key1`
    - `val1`
- `keyvaln`
    - `key2`
    - `val2`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_SPEECH

SPEECH()

Synopsis

Gets information about speech recognition results.

Description

Gets information about speech recognition results.

Syntax

```
SPEECH(argument)
```

Arguments

- `argument`
    - `status` - Returns `1` upon speech object existing, or `0` if not
    - `spoke` - Returns `1` if spoker spoke, or `0` if not
    - `results` - Returns number of results that were recognized.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_SPEECH_ENGINE

SPEECH_ENGINE()

Synopsis

Get or change a speech engine specific attribute.

Description

Changes a speech engine specific attribute.

Syntax

```
SPEECH_ENGINE(name)
```

Arguments

- `name`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_SPEECH_GRAMMAR

SPEECH_GRAMMAR()

Synopsis

Gets the matched grammar of a result if available.

Description

Gets the matched grammar of a result if available.

Syntax

```
SPEECH_GRAMMAR(nbest_number/result_number)
```

Arguments

- `nbest_number`
- `result_number`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_SPEECH_RESULTS_TYPE

SPEECH_RESULTS_TYPE()

Synopsis

Sets the type of results that will be returned.

Description

Sets the type of results that will be returned. Valid options are normal or nbest.

Syntax

```
SPEECH_RESULTS_TYPE()
```

Arguments

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_SPEECH_SCORE

SPEECH_SCORE()

Synopsis

Gets the confidence score of a result.

Description

Gets the confidence score of a result.

Syntax

```
SPEECH_SCORE(nbest_number/result_number)
```

Arguments

- `nbest_number`
- `result_number`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_SPEECH_TEXT

SPEECH_TEXT()

Synopsis

Gets the recognized text of a result.

Description

Gets the recognized text of a result.

Syntax

```
SPEECH_TEXT(nbest_number/result_number)
```

Arguments

- `nbest_number`
- `result_number`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_SPRINTF

SPRINTF()

Synopsis

Format a variable according to a format string.

Description

Parses the format string specified and returns a string matching that format. Supports most options found in **sprintf(3)**. Returns a shortened string if a format specifier is not recognized.

Syntax

```
SPRINTF(format,arg1,arg2[,...],argN)
```

Arguments

- `format`
- `arg1`
- `arg2`
- `argN`

See Also

- `sprintf(3)`

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_SQL_ESC

SQL_ESC()

Synopsis

Escapes single ticks for use in SQL statements.

Description

Used in SQL templates to escape data which may contain single ticks `'` which are otherwise used to delimit data.

Example: SELECT foo FROM bar WHERE baz='${SQL_ESC(${ARG1})}'

Syntax

```
SQL_ESC(string)
```

Arguments

- `string`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_SRVQUERY

SRVQUERY()

Synopsis

Initiate an SRV query.

Description

This will do an SRV lookup of the given service.

Syntax

```
SRVQUERY(service)
```

Arguments

- `service` - The service for which to look up SRV records. An example would be something like `_sip._udp.example.com`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_SRVRESULT

SRVRESULT()

Synopsis

Retrieve results from an SRVQUERY.

Description

This function will retrieve results from a previous use of the SRVQUERY function.

Syntax

```
SRVRESULT(id,resultnum)
```

Arguments

- `id` - The identifier returned by the SRVQUERY function.
- `resultnum` - The number of the result that you want to retrieve.
  Results start at `1`. If this argument is specified as `getnum`, then it will return the total number of results that are available.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_STACK_PEEK

STACK_PEEK()

Synopsis

View info about the location which called Gosub

Description

Read the calling {{c}}ontext, {{e}}xtension, {{p}}riority, or {{l}}abel, as specified by *which*, by going up *n* frames in the Gosub stack. If *suppress* is true, then if the number of available stack frames is exceeded, then no error message will be printed.

Syntax

```
STACK_PEEK(n,which[,suppress])
```

Arguments

- `n`
- `which`
- `suppress`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_STAT

STAT()

Synopsis

Does a check on the specified file.

Description

Syntax

```
STAT(flag,filename)
```

Arguments

- `flag` - Flag may be one of the following:
  d - Checks if the file is a directory.
  e - Checks if the file exists.
  f - Checks if the file is a regular file.
  m - Returns the file mode (in octal)
  s - Returns the size (in bytes) of the file
  A - Returns the epoch at which the file was last accessed.
  C - Returns the epoch at which the inode was last changed.
  M - Returns the epoch at which the file was last modified.
- `filename`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_STRFTIME

STRFTIME()

Synopsis

Returns the current date/time in the specified format.

Description

STRFTIME supports all of the same formats as the underlying C function **strftime(3)**. It also supports the following format: `%[n]q` - fractions of a second, with leading zeros.

Example: `%3q` will give milliseconds and `%1q` will give tenths of a second. The default is set at milliseconds (n=3). The common case is to use it in combination with %S, as in `%S.%3q`.

Syntax

```
STRFTIME(epoch,timezone,format)
```

Arguments

- `epoch`
- `timezone`
- `format`

See Also

- `strftime(3)`

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_STRPTIME

STRPTIME()

Synopsis

Returns the epoch of the arbitrary date/time string structured as described by the format.

Description

This is useful for converting a date into `EPOCH` time, possibly to pass to an application like SayUnixTime or to calculate the difference between the two date strings

Example: ${STRPTIME(2006-03-01 07:30:35,America/Chicago,%Y-%m-%d %H:%M:%S)} returns 1141219835

Syntax

```
STRPTIME(datetime,timezone,format)
```

Arguments

- `datetime`
- `timezone`
- `format`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_STRREPLACE

STRREPLACE()

Synopsis

Replace instances of a substring within a string with another string.

Description

Searches for all instances of the *find-string* in provided variable and replaces them with *replace-string*. If *replace-string* is an empty string, this will effecively delete that substring. If *max-replacements* is specified, this function will stop after performing replacements *max-replacements* times.

> **Note**
> The replacement only occurs in the output. The original variable is not altered.

Syntax

```
STRREPLACE(varname,find-string[,replace-string[,max-replacements]])
```

Arguments

- `varname`
- `find-string`
- `replace-string`
- `max-replacements`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_SYSINFO

SYSINFO()

Synopsis

Returns system information specified by parameter.

Description

Returns information from a given parameter.

Syntax

```
SYSINFO(parameter)
```

Arguments

- `parameter`
  - `loadavg` - System load average from past minute.
  - `numcalls` - Number of active calls currently in progress.
  - `uptime` - System uptime in hours.

    > **Note**
    > This parameter is dependant upon operating system.

  - `totalram` - Total usable main memory size in KiB.

    > **Note**
    > This parameter is dependant upon operating system.

  - `freeram` - Available memory size in KiB.

    > **Note**
    > This parameter is dependant upon operating system.

  - `bufferram` - Memory used by buffers in KiB.

    > **Note**
    > This parameter is dependant upon operating system.

  - `totalswap` - Total swap space still available in KiB.

    > **Note**
    > This parameter is dependant upon operating system.

  - `freeswap` - Free swap space still available in KiB.

    > **Note**
    > This parameter is dependant upon operating system.

  - `numprocs` - Number of current processes.

    > **Note**
    > This parameter is dependant upon operating system.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_TESTTIME

TESTTIME()

Synopsis

Sets a time to be used with the channel to test logical conditions.

Description

To test dialplan timing conditions at times other than the current time, use this function to set an alternate date and time. For example, you may wish to evaluate whether a location will correctly identify to callers that the area is closed on Christmas Day, when Christmas would otherwise fall on a day when the office is normally open.

Syntax

```
TESTTIME(date,time[,zone])
```

Arguments

- `date` - Date in ISO 8601 format
- `time` - Time in HH:MM:SS format (24-hour time)
- `zone` - Timezone name

See Also

- Asterisk 12 Application_GotoIfTime

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_TIMEOUT

TIMEOUT()

Synopsis

Gets or sets timeouts on the channel. Timeout values are in seconds.

Description

The timeouts that can be manipulated are:

absolute: The absolute maximum amount of time permitted for a call. Setting of 0 disables the timeout.

digit: The maximum amount of time permitted between digits when the user is typing in an extension. When this timeout expires, after the user has started to type in an extension, the extension will be considered complete, and will be interpreted. Note that if an extension typed in is valid, it will not have to timeout to be tested, so typically at the expiry of this timeout, the extension will be considered invalid (and thus control would be passed to the i extension, or if it doesn't exist the call would be terminated). The default timeout is 5 seconds.

response: The maximum amount of time permitted after falling through a series of priorities for a channel in which the user may begin typing an extension. If the user does not type an extension in this amount of time, control will pass to the t extension if it exists, and if not the call would be terminated. The default timeout is 10 seconds.

Syntax

```
TIMEOUT(timeouttype)
```

Arguments

- `timeouttype` - The timeout that will be manipulated. The possible timeout types are: `absolute`, `digit` or `response`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_TOLOWER

TOLOWER()

Synopsis

Convert string to all lowercase letters.

Description

Example: ${TOLOWER(Example)} returns "example"

Syntax

```
TOLOWER(string)
```

Arguments

- `string`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_TOUPPER

TOUPPER()

Synopsis

Convert string to all uppercase letters.

Description

Example: ${TOUPPER(Example)} returns "EXAMPLE"

Syntax

```
TOUPPER(string)
```

Arguments

- `string`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_TRYLOCK

TRYLOCK()

Synopsis

Attempt to obtain a named mutex.

Description

Attempts to grab a named lock exclusively, and prevents other channels from obtaining the same lock. Returns `1` if the lock was available or `0` otherwise.

Syntax

```
TRYLOCK(lockname)
```

Arguments

- `lockname`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_TXTCIDNAME

TXTCIDNAME()

Synopsis

TXTCIDNAME looks up a caller name via DNS.

Description

This function looks up the given phone number in DNS to retrieve the caller id name. The result will either be blank or be the value found in the TXT record in DNS.

Syntax

```
TXTCIDNAME(number,zone-suffix)
```

Arguments

- `number`
- `zone-suffix` - If no *zone-suffix* is given, the default will be `e164.arpa`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_UNLOCK

UNLOCK()

Synopsis

Unlocks a named mutex.

Description

Unlocks a previously locked mutex. Returns `1` if the channel had a lock or `0` otherwise.

> **Note**
> It is generally unnecessary to unlock in a hangup routine, as any locks held are automatically freed when the channel is destroyed.

Syntax

```
UNLOCK(lockname)
```

Arguments

- `lockname`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_UNSHIFT

UNSHIFT()

Synopsis

Inserts one or more values to the beginning of a variable containing delimited text

Description

Example: Set(UNSHIFT(array)=one,two,three) would insert one, two, and three before the values stored in the variable "array".

Syntax

```
UNSHIFT(varname[,delimiter])
```

Arguments

- `varname`
- `delimiter`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_URIDECODE

URIDECODE()

Synopsis

Decodes a URI-encoded string according to RFC 2396.

Description

Returns the decoded URI-encoded *data* string.

Syntax

```
URIDECODE(data)
```

Arguments

- `data` - Input string to be decoded.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_URIENCODE

URIENCODE()

Synopsis

Encodes a string to URI-safe encoding according to RFC 2396.

Description

Returns the encoded string defined in *data*.

Syntax

```
URIENCODE(data)
```

Arguments

- `data` - Input string to be encoded.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_VALID_EXTEN

VALID_EXTEN()

Synopsis

Determine whether an extension exists or not.

Description

Returns a true value if the indicated *context*, *extension*, and *priority* exist.

> **Warning**
> This function has been deprecated in favor of the `DIALPLAN_EXISTS()` function

Syntax

```
VALID_EXTEN(context,extension,priority)
```

Arguments

- `context` - Defaults to the current context
- `extension`
- `priority` - Priority defaults to `1`.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_VERSION

VERSION()

Synopsis

Return the Version info for this Asterisk.

Description

If there are no arguments, return the version of Asterisk in this format: SVN-branch-1.4-r44830M

Example: Set(junky=${VERSION()};

Sets junky to the string `SVN-branch-1.6-r74830M`, or possibly, `SVN-trunk-r45126M`.

Syntax

```
VERSION(info)
```

Arguments

- `info` - The possible values are:
    - `ASTERISK_VERSION_NUM` - A string of digits is returned, e.g. 10602 for 1.6.2 or 100300 for 10.3.0, or 999999 when using an SVN build.
    - `BUILD_USER` - The string representing the user's name whose account was used to configure Asterisk, is returned.
    - `BUILD_HOSTNAME` - The string representing the name of the host on which Asterisk was configured, is returned.
    - `BUILD_MACHINE` - The string representing the type of machine on which Asterisk was configured, is returned.
    - `BUILD_OS` - The string representing the OS of the machine on which Asterisk was configured, is returned.
    - `BUILD_DATE` - The string representing the date on which Asterisk was configured, is returned.
    - `BUILD_KERNEL` - The string representing the kernel version of the machine on which Asterisk was configured, is returned.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_VM_INFO

VM_INFO()

Synopsis

Returns the selected attribute from a mailbox.

Description

Returns the selected attribute from the specified *mailbox*. If *context* is not specified, defaults to the `default` context. Where the *folder* can be specified, common folders include `INBOX`, `Old`, `Work`, `Family` and `Friends`.

Syntax

```
VM_INFO(mailbox,attribute[,folder])
```

Arguments

- `mailbox`
    - `mailbox`
    - `context`
- `attribute`
    - `count` - Count of messages in specified *folder*. If *folder* is not specified, defaults to `INBOX`.
    - `email` - E-mail address associated with the mailbox.
    - `exists` - Returns a boolean of whether the corresponding *mailbox* exists.
    - `fullname` - Full name associated with the mailbox.

        - `language` - Mailbox language if overridden, otherwise the language of the channel.
        - `locale` - Mailbox locale if overridden, otherwise global locale.
        - `pager` - Pager e-mail address associated with the mailbox.
        - `password` - Mailbox access password.
        - `tz` - Mailbox timezone if overridden, otherwise global timezone
- `folder` - If not specified, `INBOX` is assumed.

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_VMCOUNT

VMCOUNT()

Synopsis

Count the voicemails in a specified mailbox.

Description

Count the number of voicemails in a specified mailbox, you could also specify the *context* and the mailbox *folder*.

Example: `exten => s,1,Set(foo=${VMCOUNT(125)})`

Syntax

```
VMCOUNT(vmbox[,folder])
```

Arguments

- `vmbox`
    - `vmbox`
    - `context` - If not specified, defaults to `default`.
- `folder` - If not specified, defaults to `INBOX`

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Function_VOLUME

VOLUME()

Synopsis

Set the TX or RX volume of a channel.

Description

The VOLUME function can be used to increase or decrease the `tx` or `rx` gain of any channel.

For example:

Set(VOLUME(TX)=3)

Set(VOLUME(RX)=2)

Set(VOLUME(TX,p)=3)

Set(VOLUME(RX,p)=3)

Syntax

```
VOLUME(direction,options)
```

Arguments

- `direction` - Must be TX or RX.
- `options`
  - p - Enable DTMF volume control

See Also

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Module Configuration

Asterisk 12 Configuration_app_agent_pool

Agent pool applications

This configuration documentation is for functionality provided by `app_agent_pool`.

Overview

> **Note**
> Option changes take effect on agent login or after an agent disconnects from a call.

agents.conf

global

Unused, but reserved.

agent-id

Configure an agent for the pool.

Configuration Option Reference

Option Name	Type	Default Value	Regular Expression	Description

ackcall	Boolean	no	false	Enable to require the agent to acknowledge a call.
acceptdtmf	String	#	false	DTMF key sequence the agent uses to acknowledge a call.
autologoff	Unsigned Integer	0	false	Time the agent has to acknowledge a call before being logged off.
wrapuptime	Unsigned Integer	0	false	Minimum time the agent has between calls.
musiconhold	String	default	false	Music on hold class the agent listens to between calls.
recordagentcalls	Boolean	no	false	Enable to automatically record calls the agent takes.
custom_beep	String	beep	false	Sound file played to alert the agent when a call is present.
fullname	String		false	A friendly name for the agent used in log messages.

Configuration Option Descriptions

ackcall

Enable to require the agent to give a DTMF acknowledgement when the agent receives a call.

> **Note**
> The option is overridden by `AGENTACKCALL` on agent login.

> **Note**
> Option changes take effect on agent login or after an agent disconnects from a call.

acceptdtmf

> **Note**
> The option is overridden by `AGENTACCEPTDTMF` on agent login.

> **Note**
> The option is ignored unless the ackcall option is enabled.

> **Note**
> Option changes take effect on agent login or after an agent disconnects from a call.

autologoff

Set how many seconds a call for the agent has to wait for the agent to acknowledge the call before the agent is automatically logged off. If set to zero then the call will wait forever for the agent to acknowledge.

> **Note**
> The option is overridden by `AGENTAUTOLOGOFF` on agent login.

> **Note**
> The option is ignored unless the ackcall option is enabled.

> **Note**
> Option changes take effect on agent login or after an agent disconnects from a call.

wrapuptime

Set the minimum amount of time in milliseconds after disconnecting a call before the agent can receive a new call.

> **Note**

The option is overridden by `AGENTWRAPUPTIME` on agent login.

Note
Option changes take effect on agent login or after an agent disconnects from a call.

musiconhold

Note
Option changes take effect on agent login or after an agent disconnects from a call.

recordagentcalls

Enable recording calls the agent takes automatically by invoking the automixmon DTMF feature when the agent connects to a caller. See `features.conf.sample` for information about the automixmon feature.

Note
Option changes take effect on agent login or after an agent disconnects from a call.

custom_beep

Note
Option changes take effect on agent login or after an agent disconnects from a call.

fullname

Note
Option changes take effect on agent login or after an agent disconnects from a call.

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Configuration_app_confbridge

Conference Bridge Application

This configuration documentation is for functionality provided by `app_confbridge`.

confbridge.conf

global

Unused, but reserved.

user_profile

A named profile to apply to specific callers.

Configuration Option Reference

Option Name	Type	Default Value	Regular Expression	Description
`type`	`None`		`false`	Define this configuration category as a user profile.
`admin`	`Boolean`	`no`	`false`	Sets if the user is an admin or not
`marked`	`Boolean`	`no`	`false`	Sets if this is a marked user or not
`startmuted`	`Boolean`	`no`	`false`	Sets if all users should start out muted
`music_on_hold_when_empty`	`Boolean`	`no`	`false`	Play MOH when user is alone or waiting on a marked user
`quiet`	`Boolean`	`no`	`false`	Silence enter/leave prompts and user intros for this user
`announce_user_count`	`Boolean`	`no`	`false`	Sets if the number of users should be announced to the user

announce_user_count_all	Custom	no	false	Announce user count to all the other users when this user joins
announce_only_user	Boolean	yes	false	Announce to a user when they join an empty conference
wait_marked	Boolean	no	false	Sets if the user must wait for a marked user to enter before joining a conference
end_marked	Boolean	no	false	Kick the user from the conference when the last marked user leaves
talk_detection_events	Boolean	no	false	Set whether or not notifications of when a user begins and ends talking should be sent out as events over AMI
dtmf_passthrough	Boolean	no	false	Sets whether or not DTMF should pass through the conference

announce_join_leave	Boolean	no	false	Prompt user for their name when joining a conference and play it to the conference when they enter
pin	String		false	Sets a PIN the user must enter before joining the conference
music_on_hold_class	String		false	The MOH class to use for this user
announcement	String		false	Sound file to play to the user when they join a conference
denoise	Boolean	no	false	Apply a denoise filter to the audio before mixing
dsp_drop_silence	Boolean	no	false	Drop what Asterisk detects as silence from audio sent to the bridge
dsp_silence_threshold	Unsigned Integer	2500	false	The number of milliseconds of detected silence necessary to trigger silence detection

dsp_talking_threshold	Unsigned Integer	160	false	The number of milliseconds of detected non-silence necessary to triger talk detection
jitterbuffer	Boolean	no	false	Place a jitter buffer on the user's audio stream before audio mixing is performed
template	Custom		false	When using the CONFBRIDGE dialplan function, use a user profile as a template for creating a new temporary profile

Configuration Option Descriptions

type

The type parameter determines how a context in the configuration file is interpreted.

- `user` - Configure the context as a *user_profile*
- `bridge` - Configure the context as a *bridge_profile*
- `menu` - Configure the context as a *menu*

announce_user_count_all

Sets if the number of users should be announced to all the other users in the conference when this user joins. This option can be either set to 'yes' or a number. When set to a number, the announcement will only occur once the user count is above the specified number.

denoise

Sets whether or not a denoise filter should be applied to the audio before mixing or not. Off by default. Requires `codec_speex` to be built and installed. Do not confuse this option with *drop_sil* *ence*. Denoise is useful if there is a lot of background noise for a user as it attempts to remove the noise while 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.

dsp_drop_silence

This option 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. Highly recommended for large conferences due to its performance enhancements.

dsp_silence_threshold

The time in milliseconds of sound falling within the what the dsp has established as baseline silence before a user is considered be silent. This value affects several operations and should not be changed unless the impact on call quality is fully understood.

What this value affects internally:

1. When talk 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 the user has stopped talking may get falsely 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 conference bridge after the user stops talking. If this value is set too low the user's audio stream may sound choppy to the other participants. This is caused by the user transitioning constantly from silence to talking during mid sentence.

The best way to approach this option is to set it slightly above the maximum amount of ms of silence a user may generate during natural speech.

By default this value is 2500ms. Valid values are 1 through 2^31.

dsp_talking_threshold

The time in milliseconds of sound above what the dsp has established as base line silence for a user before a user is considered to be talking. This value affects several operations and should not be changed unless the impact on call quality is fully understood.

What this value affects internally:

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 talk detection AMI events are enabled, this value determines when talking has begun which results in an AMI event to fire. If this value is set too tight AMI events may be falsely triggered by variants in room noise.

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.

By default this value is 160 ms. Valid values are 1 through 2^31

jitterbuffer

Enabling this option places a jitterbuffer on the user's audio stream before audio mixing is performed. This is highly recommended but will add a slight delay to the audio. This option is using the `JITTERBUFFER` dialplan function's default adaptive jitterbuffer. For a more fine tuned jitterbuffer, disable this option and use the `JITTERBUFFER` dialplan function on the user before entering the ConfBridge application.

bridge_profile

A named profile to apply to specific bridges.

Configuration Option Reference

Option Name	Type	Default Value	Regular Expression	Description
`type`	`None`		`false`	Define this configuration category as a bridge profile
`jitterbuffer`	`Boolean`	`no`	`false`	Place a jitter buffer on the conference's audio stream
`internal_sample_rate`	`Unsigned Integer`	`0`	`false`	Set the internal native sample rate for mixing the conference
`mixing_interval`	`Custom`	`20`	`false`	Sets the internal mixing interval in milliseconds for the bridge

record_conference	Boolean	no	false	Record the conference starting with the first active user's entrance and ending with the last active user's exit
record_file	String	confbridge-name of conference bridge-start time.wav	false	The filename of the conference recording
record_file_append	Boolean	yes	false	Append record file when starting/stopping on same conference recording
video_mode	Custom		false	Sets how confbridge handles video distribution to the conference participants
max_members	Unsigned Integer	0	false	Limit the maximum number of participants for a single conference
sound_	Custom		true	Override the various conference bridge sound files

`template`	`Custom`		`false`	When using the CONFBRIDGE dialplan function, use a bridge profile as a template for creating a new temporary profile

Configuration Option Descriptions

type

The type parameter determines how a context in the configuration file is interpreted.

- `user` - Configure the context as a *user_profile*
- `bridge` - Configure the context as a *bridge_profile*
- `menu` - Configure the context as a *menu*

internal_sample_rate

Sets the internal native sample rate the conference is mixed at. This is set to automatically adjust the sample rate to the best quality by default. Other values can be anything from 8000-192000. If a sample rate is set that Asterisk does not support, the closest sample rate Asterisk does support to the one requested will be used.

mixing_interval

Sets the internal mixing interval in milliseconds for the bridge. This number reflects how tight or loose the mixing will be for the conference. In order to improve performance a larger mixing interval such as 40ms may be chosen. Using a larger mixing interval comes at the cost of introducing larger amounts of delay into the bridge. Valid values here are 10, 20, 40, or 80.

record_conference

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 slinear. This file will be located in the configured monitoring directory in `asterisk.conf`.

record_file

When *record_conference* is set to yes, the specific name of the record file can be set using this option. Note that since multiple conferences may use the same bridge profile, this may cause issues depending on the configuration. It is recommended to only use this option dynamically with the `CONFBRIDGE()` dialplan function. This allows the record name to be specified and a unique name to be chosen. By default, the record_file is stored in Asterisk's spool/monitor directory with a unique filename starting with the 'confbridge' prefix.

record_file_append

When *record_file_append* is set to yes, stopping and starting recording on a conference adds the new portion to end of current record_file. When this is set to no, a new *record_file* is generated every time you start then stop recording on a conference.

video_mode

Sets how confbridge handles video distribution to the conference participants. Note that participants wanting to view and be the source of a video feed **MUST** be sharing the same video codec. Also, using video in conjunction with with the jitterbuffer currently results in the audio being slightly out of sync with the video. This is a result of the jitterbuffer only working on the audio stream. It is recommended to disable the jitterbuffer when video is used.

- `none` - No video sources are set by default in the conference. It is still possible for a user to be set as a video source via AMI or DTMF action at any time.
- `follow_talker` - The video feed will follow whoever is talking and providing video.
- `last_marked` - The last marked user to join the conference with video capabilities will be the single source of video distributed to all participants. If multiple marked users are capable of video, the last one to join is always the source, when that user leaves it goes to the one who joined before them.
- `first_marked` - The first marked user to join the conference with video capabilities is the single source of video distribution among all participants. If that user leaves, the marked user to join after them becomes the source.

max_members

This option 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. Note however that an Admin user will always be alowed to join the conference regardless if this limit is reached or not.

sound_

All sounds in the conference are customizable using the bridge profile options below. Simply state the option followed by the filename or full path of the filename after the option. Example: `sound_had_joined=conf-hasjoin` This will play the `conf-hasjoin` sound file found in the sounds directory when announcing someone's name is joining the conference.

- `sound_join` - The sound played to everyone when someone enters the conference.
- `sound_leave` - The sound played to everyone when someone leaves the conference.
- `sound_has_joined` - The sound played before announcing someone's name has joined the conference. This is used for user intros. Example `"_____ has joined the conference"`
- `sound_has_left` - The sound played when announcing someone's name has left the conference. This is used for user intros. Example `"_____ has left the conference"`
- `sound_kicked` - The sound played to a user who has been kicked from the conference.
- `sound_muted` - The sound played when the mute option it toggled on.
- `sound_unmuted` - The sound played when the mute option it toggled off.
- `sound_only_person` - The sound played when the user is the only person in the conference.

- `sound_only_one` - The sound played to a user when there is only one other person is in the conference.
- `sound_there_are` - The sound played when announcing how many users there are in a conference.
- `sound_other_in_party` - This file is used in conjunction with `sound_there_are` when announcing how many users there are in the conference. The sounds are stringed together like this. `"sound_there_are" ${number of participants} "sound_other_in_party"`
- `sound_place_into_conference` - The sound played when someone is placed into the conference after waiting for a marked user.
- `sound_wait_for_leader` - The sound played when a user is placed into a conference that can not start until a marked user enters.
- `sound_leader_has_left` - The sound played when the last marked user leaves the conference.
- `sound_get_pin` - The sound played when prompting for a conference pin number.
- `sound_invalid_pin` - The sound played when an invalid pin is entered too many times.
- `sound_locked` - The sound played to a user trying to join a locked conference.
- `sound_locked_now` - The sound played to an admin after toggling the conference to locked mode.
- `sound_unlocked_now` - The sound played to an admin after toggling the conference to unlocked mode.
- `sound_error_menu` - The sound played when an invalid menu option is entered.

menu

A conference user menu

Configuration Option Reference

Option Name	Type	Default Value	Regular Expression	Description
`type`	`None`		`false`	Define this configuration category as a menu
`0-9A-D*#`	`Custom`		`true`	DTMF sequences to assign various confbridge actions to

Configuration Option Descriptions

type

The type parameter determines how a context in the configuration file is interpreted.

- `user` - Configure the context as a *user_profile*
- `bridge` - Configure the context as a *bridge_profile*
- `menu` - Configure the context as a *menu*

0-9A-D*#

The ConfBridge application also has the ability to apply custom DTMF menus to each channel using the application. Like the User and Bridge profiles a menu is passed in to ConfBridge as an argument in the dialplan.

Below is a list of menu actions that can be assigned to a DTMF sequence.

Note

A single DTMF sequence can have multiple actions associated with it. This is accomplished by stringing the actions together and using a `,` as the delimiter. Example: Both listening and talking volume is reset when 5 is pressed. `5=reset_talking_volume, reset_listening_volume`

- `playback(filename&filename2&...)` - `playback` will play back an audio file to a channel and then immediately return to the conference. This file can not be interupted by DTMF. Multiple files can be chained together using the & character.
- `playback_and_continue(filename&filename2&...)` - `playback_and_continue` will play back a prompt while continuing to collect the dtmf sequence. This is useful when using a menu prompt that describes all the menu options. Note however that any DTMF during this action will terminate the prompts playback. Prompt files can be chained together using the & character as a delimiter.
- `toggle_mute` - Toggle turning on and off mute. Mute will make the user silent to everyone else, but the user will still be able to listen in. continue to collect the dtmf sequence.
- `no_op` - This action does nothing (No Operation). Its only real purpose exists for being able to reserve a sequence in the config as a menu exit sequence.
- `decrease_listening_volume` - Decreases the channel's listening volume.
- `increase_listening_volume` - Increases the channel's listening volume.
- `reset_listening_volume` - Reset channel's listening volume to default level.
- `decrease_talking_volume` - Decreases the channel's talking volume.
- `increase_talking_volume` - Increases the channel's talking volume.
- `reset_talking_volume` - Reset channel's talking volume to default level.
- `dialplan_exec(context,exten,priority)` - The `dialplan_exec` action allows a user to escape from the conference and execute commands in the dialplan. Once the dialplan exits the user will be put back into the conference. The possibilities are endless!
- `leave_conference` - This action allows a user to exit the conference and continue execution in the dialplan.
- `admin_kick_last` - This action allows an Admin to kick the last participant from the conference. This action will only work for admins which allows a single menu to be used for both users and admins.
- `admin_toggle_conference_lock` - This action allows an Admin to toggle locking and unlocking the conference. Non admins can not use this action even if it is in their menu.
- `set_as_single_video_src` - This action allows any user to set themselves as the single video source distributed to all participants. This will make the video feed stick to them regardless of what the `video_mode` is set to.
- `release_as_single_video_src` - This action allows a user to release themselves as the video source. If `video_mode` is not set to `none` this action will result in the conference returning to whatever video mode the bridge profile is using.
  Note that this action will have no effect if the user is not currently the video source. Also, the user is not guaranteed by using this action that they will not become the video source again. The bridge will return to whatever operation the `video_mode` option is set to upon release of the video src.
- `admin_toggle_mute_participants` - This action allows an administrator to toggle the mute state for all non-admins within a conference. All admin users are unaffected by this option. Note that all users, regardless of their admin status, are notified that the conference is muted.
- `participant_count` - This action plays back the number of participants currently in a conference

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Configuration_app_skel

This configuration documentation is for functionality provided by `app_skel`.

app_skel.conf

globals

Options that apply globally to app_skel

Configuration Option Reference

Option Name	Type	Default Value	Regular Expression	Description
`games`				The number of games a single execution of SkelGuessNumber will play
`cheat`				Should the computer cheat?

Configuration Option Descriptions

cheat

If enabled, the computer will ignore winning guesses.

sounds

Prompts for SkelGuessNumber to play

Configuration Option Reference

Option Name	Type	Default Value	Regular Expression	Description
`prompt`		`please-enter-yournumberqueue-less-than`		A prompt directing the user to enter a number less than the max number
`wrong_guess`		`vm-pls-try-again`		The sound file to play when a wrong guess is made
`right_guess`		`auth-thankyou`		The sound file to play when a correct guess is made
`too_low`				The sound file to play when a guess is too low

too_high				The sound file to play when a guess is too high
lose		vm-goodbye		The sound file to play when a player loses

level

Defined levels for the SkelGuessNumber game

Configuration Option Reference

Option Name	Type	Default Value	Regular Expression	Description
max_number				The maximum in the range of numbers to guess (1 is the implied minimum)
max_guesses				The maximum number of guesses before a game is considered lost

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Configuration_cdr

Call Detail Record configuration

This configuration documentation is for functionality provided by `cdr`.

Overview

CDR is Call Detail Record, which provides logging services via a variety of pluggable backend modules. Detailed call information can be recorded to databases, files, etc. Useful for billing, fraud prevention, compliance with Sarbanes-Oxley aka The Enron Act, QOS evaluations, and more.

cdr.conf

general

Global settings applied to the CDR engine.

Configuration Option Reference

Option Name	Type	Default Value	Regular Expression	Description
`debug`	`Boolean`		`false`	Enable/disable verbose CDR debugging.
`enable`	`Boolean`	`1`	`false`	Enable/disable CDR logging.
`unanswered`	`Boolean`	`0`	`false`	Log calls that are never answered.
`congestion`	`Boolean`		`false`	Log congested calls.
`endbeforehexten`	`Boolean`	`0`	`false`	Don't produce CDRs while executing hangup logic
`initiatedseconds`	`Boolean`	`0`	`false`	Count microseconds for billsec purposes
`batch`	`Boolean`	`0`	`false`	Submit CDRs to the backends for processing in batches
`size`	`Unsigned Integer`	`100`	`false`	The maximum number of CDRs to accumulate before triggering a batch

time	Unsigned Integer	300	false	The maximum time to accumulate CDRs before triggering a batch
scheduleronly	Boolean	0	false	Post batched CDRs on their own thread instead of the scheduler
safeshutdown	Boolean	1	false	Block shutdown of Asterisk until CDRs are submitted

Configuration Option Descriptions

debug

When set to `True`, verbose updates of changes in CDR information will be logged. Note that this is only of use when debugging CDR behavior.

enable

Define whether or not to use CDR logging. Setting this to "no" will override any loading of backend CDR modules. Default is "yes".

unanswered

Define whether or not to log unanswered calls. Setting this to "yes" will report every attempt to ring a phone in dialing attempts, when it was not answered. For example, if you try to dial 3 extensions, and this option is "yes", you will get 3 CDR's, one for each phone that was rung. Some find this information horribly useless. Others find it very valuable. Note, in "yes" mode, you will see one CDR, with one of the call targets on one side, and the originating channel on the other, and then one CDR for each channel attempted. This may seem redundant, but cannot be helped.

In brief, this option controls the reporting of unanswered calls which only have an A party. Calls which get offered to an outgoing line, but are unanswered, are still logged, and that is the intended behavior. (It also results in some B side CDRs being output, as they have the B side channel as their source channel, and no destination channel.)

congestion

Define whether or not to log congested calls. Setting this to "yes" will report each call that fails to complete due to congestion conditions.

endbeforehexten

As each CDR for a channel is finished, its end time is updated and the CDR is finalized. When a channel is hung up and hangup logic is present (in the form of a hangup handler or the h extensi on), a new CDR is generated for the channel. Any statistics are gathered from this new CDR. By enabling this option, no new CDR is created for the dialplan logic that is executed in h extensions or attached hangup handler subroutines. The default value is `no`, indicating that a CDR will be generated during hangup logic.

initiatedseconds

Normally, the `billsec` field logged to the CDR backends is simply the end time (hangup time) minus the answer time in seconds. Internally, asterisk stores the time in terms of microseconds and seconds. By setting initiatedseconds to `yes`, you can force asterisk to report any seconds that were initiated (a sort of round up method). Technically, this is when the microsecond part of the end time is greater than the microsecond part of the answer time, then the billsec time is incremented one second.

batch

Define the CDR batch mode, where instead of posting the CDR at the end of every call, the data will be stored in a buffer to help alleviate load on the asterisk server.

> **Warning**
> Use of batch mode may result in data loss after unsafe asterisk termination, i.e., software crash, power failure, kill -9, etc.

size

Define the maximum number of CDRs to accumulate in the buffer before posting them to the backend engines. batch must be set to `yes`.

time

Define the maximum time to accumulate CDRs before posting them in a batch to the backend engines. If this time limit is reached, then it will post the records, regardless of the value defined for size. batch must be set to `yes`.

> **Note**
> Time is expressed in seconds.

scheduleronly

The CDR engine uses the internal asterisk scheduler to determine when to post records. Posting can either occur inside the scheduler thread, or a new thread can be spawned for the submission of every batch. For small batches, it might be acceptable to just use the scheduler thread, so set this to `yes`. For large batches, say anything over size=10, a new thread is recommended, so set this to no.

safeshutdown

When shutting down asterisk, you can block until the CDRs are submitted. If you don't, then data will likely be lost. You can always check the size of the CDR batch buffer with the CLI `cdr status` command. To enable blocking on submission of CDR data during asterisk shutdown, set this to `yes`.

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Configuration_cel

This configuration documentation is for functionality provided by `cel`.

cel.conf

general

Options that apply globally to Channel Event Logging (CEL)

Configuration Option Reference

Option Name	Type	Default Value	Regular Expression	Description
`enable`	`Boolean`	`no`	`false`	Determines whether CEL is enabled
`dateformat`	`String`		`false`	The format to be used for dates when logging
`apps`	`Custom`		`false`	List of apps for CEL to track
`events`	`Custom`		`false`	List of events for CEL to track

Configuration Option Descriptions

apps

A case-insensitive, comma-separated list of applications to track when one or both of APP_START and APP_END events are flagged for tracking

events

A case-sensitive, comma-separated list of event names to track. These event names do not include the leading `AST_CEL`.

- `ALL` - Special value which tracks all events.
- `CHAN_START`
- `CHAN_END`
- `ANSWER`
- `HANGUP`
- `APP_START`
- `APP_END`
- `PARK_START`
- `PARK_END`
- `USER_DEFINED`
- `BRIDGE_ENTER`
- `BRIDGE_EXIT`
- `BLINDTRANSFER`
- `ATTENDEDTRANSFER`
- `PICKUP`
- `FORWARD`
- `LINKEDID_END`
- `LOCAL_OPTIMIZE`

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Configuration_chan_motif

Jingle Channel Driver

This configuration documentation is for functionality provided by `chan_motif`.

Overview

Transports

There are three different transports and protocol derivatives supported by `chan_motif`. They are in order of preference: Jingle using ICE-UDP, Google Jingle, and Google-V1.

Jingle as defined in XEP-0166 supports the widest range of features. It is referred to as `ice-udp` . This is the specification that Jingle clients implement.

Google Jingle follows the Jingle specification for signaling but uses a custom transport for media. It is supported by the Google Talk Plug-in in Gmail and by some other Jingle clients. It is referred to as `google` in this file.

Google-V1 is the original Google Talk signaling protocol which uses an initial preliminary version of Jingle. It also uses the same custom transport as Google Jingle for media. It is supported by Google Voice, some other Jingle clients, and the Windows Google Talk client. It is referred to as `google-v1` in this file.

Incoming sessions will automatically switch to the correct transport once it has been determined.

Outgoing sessions are capable of determining if the target is capable of Jingle or a Google transport if the target is in the roster. Unfortunately it is not possible to differentiate between a Google Jingle or Google-V1 capable resource until a session initiate attempt occurs. If a resource is determined to use a Google transport it will initially use Google Jingle but will fall back to Google-V1 if required.

If an outgoing session attempt fails due to failure to support the given transport `chan_motif` will fall back in preference order listed previously until all transports have been exhausted.

Dialing and Resource Selection Strategy

Placing a call through an endpoint can be accomplished using the following dial string:

Motif/endpoint name/target

When placing an outgoing call through an endpoint the requested target is searched for in the roster list. If present the first Jingle or Google Jingle capable resource is specifically targeted. Since the capabilities of the resource are known the outgoing session initiation will disregard the configured transport and use the determined one.

If the target is not found in the roster the target will be used as-is and a session will be initiated using the transport specified in this configuration file. If no transport has been specified the endpoint defaults to `ice-udp`.

Video Support

Support for video does not need to be explicitly enabled. Configuring any video codec on your endpoint will automatically enable it.

DTMF

The only supported method for DTMF is RFC2833. This is always enabled on audio streams and negotiated if possible.

Incoming Calls

Incoming calls will first look for the extension matching the name of the endpoint in the configured context. If no such extension exists the call will automatically fall back to the `s` extensi on.

CallerID

The incoming caller id number is populated with the username of the caller and the name is populated with the full identity of the caller. If you would like to perform authentication or filtering of incoming calls it is recommended that you use these fields to do so.

Outgoing caller id can **not** be set.

> **Warning**
> Multiple endpoints using the same connection is **NOT** supported. Doing so may result in broken calls.

motif.conf

endpoint

The configuration for an endpoint.

Configuration Option Reference

Option Name	Type	Default Value	Regular Expression	Description
`context`	`String`	`default`	`false`	Default dialplan context that incoming sessions will be routed to
`callgroup`	`Custom`		`false`	A callgroup to assign to this endpoint.
`pickupgroup`	`Custom`		`false`	A pickup group to assign to this endpoint.
`language`	`String`		`false`	The default language for this endpoint.
`musicclass`	`String`		`false`	Default music on hold class for this endpoint.
`parkinglot`	`String`		`false`	Default parking lot for this endpoint.
`accountcode`	`String`		`false`	Accout code for CDR purposes
`allow`	`Codec`	`ulaw,alaw`	`false`	Codecs to allow

disallow	Codec	all	false	Codecs to disallow
connection	Custom		false	Connection to accept traffic on and on which to send traffic out
transport	Custom		false	The transport to use for the endpoint.
maxicecandidates	Unsigned Integer	10	false	Maximum number of ICE candidates to offer
maxpayloads	Unsigned Integer	30	false	Maximum number of pyaloads to offer

Configuration Option Descriptions

transport

The default outbound transport for this endpoint. Inbound messages are inferred. Allowed transports are `ice-udp`, `google`, or `google-v1`. Note that `chan_motif` will fall back to transport preference order if the transport value chosen here fails.

- `ice-udp` - The Jingle protocol, as defined in XEP 0166.
- `google` - The Google Jingle protocol, which follows the Jingle specification for signaling but uses a custom transport for media.
- `google-v1` - Google-V1 is the original Google Talk signaling protocol which uses an initial preliminary version of Jingle. It also uses the same custom transport as `google` for media.

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Configuration_core

Bucket file API

This configuration documentation is for functionality provided by `core`.

bucket

bucket

Configuration Option Reference

Option Name	Type	Default Value	Regular Expression	Description
scheme	String		false	Scheme in use for bucket
created	Custom		false	Time at which the bucket was created
modified	Custom		false	Time at which the bucket was last modified

file

Configuration Option Reference

Option Name	Type	Default Value	Regular Expression	Description
scheme	String		false	Scheme in use for file
created	Custom		false	Time at which the file was created
modified	Custom		false	Time at which the file was last modified

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Configuration_features

Features Configuration

This configuration documentation is for functionality provided by `features`.

features.conf

globals

Configuration Option Reference

Option Name	Type	Default Value	Regular Expression	Description

featuredigittimeout	Custom	1000	false	Milliseconds allowed between digit presses when entering a feature code.
courtesytone	Custom		false	Sound to play when automon or automixmon is activated
recordingfailsound	Custom		false	Sound to play when automon or automixmon is attempted but fails to start
transferdigittimeout	Custom	3000	false	Milliseconds allowed between digit presses when dialing a transfer destination
atxfernoanswertimeout	Custom	15000	false	Milliseconds to wait for attended transfer destination to answer
atxferdropcall	Custom	0	false	Hang up the call entirely if the attended transfer fails
atxferloopdelay	Custom	10000	false	Milliseconds to wait between attempts to re-dial transfer destination

atxfercallbackretries	Custom	2	false	Number of times to re-attempt dialing a transfer destination
xfersound	Custom	beep	false	Sound to play to during transfer and transfer-like operations.
xferfailsound	Custom	beeperr	false	Sound to play to a transferee when a transfer fails
atxferabort	Custom	*1	false	Digits to dial to abort an attended transfer attempt
atxfercomplete	Custom	*2	false	Digits to dial to complete an attended transfer
atxferthreeway	Custom	*3	false	Digits to dial to change an attended transfer into a three-way call
atxferswap	Custom	*4	false	Digits to dial to toggle who the transferrer is currently bridged to during an attended transfer
pickupexten	Custom	*8	false	Digits used for picking up ringing calls
pickupsound	Custom		false	Sound to play to picker when a call is picked up

`pickupfailsound`	`Custom`		`false`	Sound to play to picker when a call cannot be picked up

Configuration Option Descriptions

atxferdropcall

When this option is set to `no`, then Asterisk will attempt to re-call the transferrer if the call to the transfer target fails. If the call to the transferrer fails, then Asterisk will wait *atxferloopdelay* milliseconds and then attempt to dial the transfer target again. This process will repeat until *atxfercallbackretries* attempts to re-call the transferrer have occurred.

When this option is set to `yes`, then Asterisk will not attempt to re-call the transferrer if the call to the transfer target fails. Asterisk will instead hang up all channels involved in the transfer.

xfersound

This sound will play to the transferrer and transfer target channels when an attended transfer completes. This sound is also played to channels when performing an AMI `Bridge` action.

atxferabort

This option is only available to the transferrer during an attended transfer operation. Aborting a transfer results in the transfer being cancelled and the original parties in the call being re-bridged.

atxfercomplete

This option is only available to the transferrer during an attended transfer operation. Completing the transfer with a DTMF sequence is functionally equivalent to hanging up the transferrer channel during an attended transfer. The result is that the transfer target and transferees are bridged.

atxferthreeway

This option is only available to the transferrer during an attended transfer operation. Pressing this DTMF sequence will result in the transferrer, the transferees, and the transfer target all being in a single bridge together.

atxferswap

This option is only available to the transferrer during an attended transfer operation. Pressing this DTMF sequence will result in the transferrer swapping which party he is bridged with. For instance, if the transferrer is currently bridged with the transfer target, then pressing this DTMF sequence will cause the transferrer to be bridged with the transferees.

pickupexten

In order for the pickup attempt to be successful, the party attempting to pick up the call must either have a *namedpickupgroup* in common with a ringing party's *namedcallgroup* or must have a *pickupgroup* in common with a ringing party's *callgroup*.

featuremap

DTMF options that can be triggered during bridged calls

Configuration Option Reference

Option Name	Type	Default Value	Regular Expression	Description
`atxfer`	`Custom`		`false`	DTMF sequence to initiate an attended transfer
`blindxfer`	`Custom`	`#`	`false`	DTMF sequence to initiate a blind transfer
`disconnect`	`Custom`	`*`	`false`	DTMF sequence to disconnect the current call
`parkcall`	`Custom`		`false`	DTMF sequence to park a call
`automon`	`Custom`		`false`	DTMF sequence to start or stop monitoring a call
`automixmon`	`Custom`		`false`	DTMF sequence to start or stop mixmonitoring a call

Configuration Option Descriptions

atxfer

The transferee parties will be placed on hold and the transferrer may dial an extension to reach a transfer target. During an attended transfer, the transferrer may consult with the transfer target before completing the transfer. Once the transferrer has hung up or pressed the *atxfercomplete* DTMF sequence, then the transferees and transfer target will be bridged.

blindxfer

The transferee parties will be placed on hold and the transferrer may dial an extension to reach a transfer target. During a blind transfer, as soon as the transfer target is dialed, the transferrer is hung up.

disconnect

Entering this DTMF sequence will cause the bridge to end, no matter the number of parties present

parkcall

The parking lot used to park the call is determined by using either the *PARKINGLOT* channel variable or a configured value on the channel (provided by the channel driver) if the variable is not present. If no configured value on the channel is present, then `"default"` is used. The call is parked in the next available space in the parking lot.

automon

This will cause the channel that pressed the DTMF sequence to be monitored by the `Monitor` a pplication. The format for the recording is determined by the *TOUCH_MONITOR_FORMAT* chan nel variable. If this variable is not specified, then `wav` is the default. The filename is constructed in the following manner:

prefix-timestamp-filename

where prefix is either the value of the *TOUCH_MONITOR_PREFIX* channel variable or `auto` if the variable is not set. The timestamp is a UNIX timestamp. The filename is either the value of the *TOUCH_MONITOR* channel variable or the callerID of the channels if the variable is not set.

automixmon

Operation of the automixmon is similar to the {{ automon }} feature, with the following exceptions: *TOUCH_MIXMONITOR* is used in place of *TOUCH_MONITOR* *TOUCH_MIXMONITOR_FORM AT* is used in place of *TOUCH_MIXMONITOR* There is no equivalent for *TOUCH_MONITOR_P REFIX*. `"auto"` is always how the filename begins.

applicationmap

Section for defining custom feature invocations during a call

Configuration Option Reference

Option Name	Type	Default Value	Regular Expression	Description
`.*`	`Custom`		`true`	A custom feature to invoke during a bridged call

Configuration Option Descriptions

.*

Each item listed here is a comma-separated list of parameters that determine how a feature may be invoked during a call

Example:

eggs = *5,self,Playback(hello-world),default

This would create a feature called `eggs` that could be invoked during a call by pressing the `*5`. The party that presses the DTMF sequence would then trigger the `Playback` application to play the `hello-world` file. The application invocation would happen on the party that pressed the DTMF sequence since `self` is specified. The other parties in the bridge would hear the `default` music on hold class during the playback.

In addition to the syntax outlined in this documentation, a backwards-compatible alternative is also allowed. The following applicationmap lines are functionally identical:

eggs = *5,self,Playback(hello-world),default

eggs = *5,self,Playback,hello-world,default

eggs = *5,self,Playback,"hello-world",default

featuregroup

Groupings of items from the applicationmap

Configuration Option Reference

Option Name	Type	Default Value	Regular Expression	Description
`.*`	`Custom`		`true`	Applicationmap item to place in the feature group

Configuration Option Descriptions

.*

Each item here must be a name of an item in the applicationmap. The argument may either be a new DTMF sequence to use for the item or it may be left blank in order to use the DTMF sequence specified in the applicationmap. For example:

eggs => *1

bacon =>

would result in the applicationmap items `eggs` and `bacon` being in the featuregroup. The former would have its default DTMF trigger overridden with `*1` and the latter would have the DTMF value specified in the applicationmap.

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Configuration_named_acl

This configuration documentation is for functionality provided by `named_acl`.

named_acl.conf

named_acl

Options for configuring a named ACL

Configuration Option Reference

Option Name	Type	Default Value	Regular Expression	Description
`permit`	`ACL`		`false`	An address/subnet from which to allow access
`deny`	`ACL`		`false`	An address/subnet from which to disallow access

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Configuration_res_ari

HTTP binding for the Stasis API

This configuration documentation is for functionality provided by `res_ari`.

ari.conf

general

General configuration settings

Configuration Option Reference

Option Name	Type	Default Value	Regular Expression	Description

enabled	Boolean	yes	false	Enable/disable the ARI module
pretty	Custom	no	false	Responses from ARI are formatted to be human readable
auth_realm	String	Asterisk REST Interface	false	Realm to use for authentication. Defaults to Asterisk REST Interface.
allowed_origins	String		false	Comma separated list of allowed origins, for Cross-Origin Resource Sharing. May be set to * to allow all origins.

user

Per-user configuration settings

Configuration Option Reference

Option Name	Type	Default Value	Regular Expression	Description
type	None		false	Define this configuration section as a user.
read_only	Boolean	no	false	When set to yes, user is only authorized for read-only requests

password	String		false	Crypted or plaintext password (see password_format)
password_format	Custom	plain	false	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. When set to plain, the password is in plaintext

Configuration Option Descriptions

type

- `user` - Configure this section as a *user*

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Configuration_res_parking

This configuration documentation is for functionality provided by `res_parking`.

res_parking.conf

globals

Options that apply to every parking lot

Configuration Option Reference

Option Name	Type	Default Value	Regular Expression	Description

parkeddynamic	Boolean	no	false	Enables dynamically created parkinglots.

parking_lot

Defined parking lots for res_parking to use to park calls on

Configuration Option Reference

Option Name	Type	Default Value	Regular Expression	Description
context	String	parkedcalls	false	The name of the context where calls are parked and picked up from.
parkext	String		false	Extension to park calls to this parking lot.
parkext_exclusive	Boolean	no	false	If yes, the extension registered as parkext will park exclusively to this parking lot.
parkpos	Custom	701-750	false	Numerical range of parking spaces which can be used to retrieve parked calls.
parkinghints	Boolean	no	false	If yes, this parking lot will add hints automatically for parking spaces.

parkingtime	Unsigned Integer	45	false	Amount of time a call will remain parked before giving up (in seconds).
parkedmusicclass	String		false	Which music class to use for parked calls. They will use the default if unspecified.
comebacktoorigin	Boolean	yes	false	Determines what should be done with the parked channel if no one picks it up before it times out.
comebackdialtime	Unsigned Integer	30	false	Timeout for the Dial extension created to call back the parker when a parked call times out.
comebackcontext	String	parkedcallstimeout	false	Context where parked calls will enter the PBX on timeout when comebacktoorigin=no
courtesytone	String		false	If the name of a sound file is provided, use this as the courtesy tone

`parkedplay`	`Custom`	`caller`	`false`	Who we should play the courtesytone to on the pickup of a parked call from this lot
`parkedcalltransfers`	`Custom`	`no`	`false`	Apply the DTMF transfer features to the caller and/or callee when parked calls are picked up.
`parkedcallreparking`	`Custom`	`no`	`false`	Apply the DTMF parking feature to the caller and/or callee when parked calls are picked up.
`parkedcallhangup`	`Custom`	`no`	`false`	Apply the DTMF Hangup feature to the caller and/or callee when parked calls are picked up.
`parkedcallrecording`	`Custom`	`no`	`false`	Apply the DTMF recording features to the caller and/or callee when parked calls are picked up

`findslot`	`Custom`	`first`	`false`	Rule to use when trying to figure out which parking space a call should be parked with.
`courtesytone`				If set, the sound set will be played to whomever is set by parkedplay

Configuration Option Descriptions

context

This option is only used if parkext is set.

parkext

If this option is used, this extension will automatically be created to place calls into parking lots. In addition, if parkext_exclusive is set for this parking lot, the name of the parking lot will be included in the application's arguments so that it only parks to this parking lot. The extension will be created in `context`. Using this option also creates extensions for retrieving parked calls from the parking spaces in the same context.

parkpos

If parkext is set, these extensions will automatically be mapped in `context` in order to pick up calls parked to these parking spaces.

comebacktoorigin

Valid Options:

- `yes` - Automatically have the parked channel dial the device that parked the call with dial timeout set by the `parkingtime` option. When the call times out an extension to dial the PARKER will automatically be created in the `park-dial` context with an extension of the flattened parker device name. If the call is not answered, the parked channel that is timing out will continue in the dial plan at that point if there are more priorities in the extension (which won't be the case unless the dialplan deliberately includes such priorities in the `park-dial` context through pattern matching or deliberately written flattened peer extensions).
- `no` - Place the call into the PBX at `comebackcontext` instead. The extension will still be set as the flattened peer name. If an extension the flattened peer name isn't available then it will fall back to the `s` extension. If that also is unavailable it will attempt to fall back to `s@default`. The normal dial extension will still be created in the `park-dial` context with the extension also being the flattened peer name.

> **Note**
> Flattened Peer Names - Extensions can not include slash characters since those are used for pattern matching. When a peer name is flattened, slashes become

underscores. For example if the parker of a call is called `SIP/0004F2040001` then flattened peer name and therefor the extensions created and used on timeouts will be `SIP_0004F204001`.

> **Note**
> When parking times out and the channel returns to the dial plan, the following variables are set:

- `PARKINGSLOT` - extension that the call was parked in prior to timing out.
- `PARKEDLOT` - name of the lot that the call was parked in prior to timing out.
- `PARKER` - The device that parked the call

comebackcontext

The extension the call enters will prioritize the flattened peer name in this context. If the flattened peer name extension is unavailable, then the 's' extension in this context will be used. If that also is unavailable, the 's' extension in the 'default' context will be used.

courtesytone

By default, this tone is only played to the caller of a parked call. Who receives the tone can be changed using the `parkedplay` option.

parkedplay

- `no` - Apply to neither side.
- `caller` - Apply to only to the caller picking up the parked call.
- `callee` - Apply to only to the parked call being picked up.
- `both` - Apply to both the caller and the callee.

> **Note**
> If courtesy tone is not specified then this option will be ignored.

parkedcalltransfers

- `no` - Apply to neither side.
- `caller` - Apply to only to the caller picking up the parked call.
- `callee` - Apply to only to the parked call being picked up.
- `both` - Apply to both the caller and the callee.

parkedcallreparking

- `no` - Apply to neither side.
- `caller` - Apply to only to the caller picking up the parked call.
- `callee` - Apply to only to the parked call being picked up.
- `both` - Apply to both the caller and the callee.

parkedcallhangup

- `no` - Apply to neither side.
- `caller` - Apply to only to the caller picking up the parked call.
- `callee` - Apply to only to the parked call being picked up.

- `both` - Apply to both the caller and the callee.

parkedcallrecording

- `no` - Apply to neither side.
- `caller` - Apply to only to the caller picking up the parked call.
- `callee` - Apply to only to the parked call being picked up.
- `both` - Apply to both the caller and the callee.

findslot

- `first` - Always try to place in the lowest available space in the parking lot
- `next` - Track the last parking space used and always attempt to use the one immediately after.

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Configuration_res_pjsip

SIP Resource using PJProject

This configuration documentation is for functionality provided by `res_pjsip`.

pjsip.conf

endpoint

Endpoint

Configuration Option Reference

Option Name	Type	Default Value	Regular Expression	Description
`100rel`	`Custom`	`yes`	`false`	Allow support for RFC3262 provisional ACK tags
`aggregate_mwi`	`Boolean`	`yes`	`false`	
`allow`	`Codec`		`false`	Media Codec(s) to allow
`aors`	`String`		`false`	AoR(s) to be used with the endpoint

auth	Custom		false	Authentication Object(s) associated with the endpoint
callerid	Custom		false	CallerID information for the endpoint
callerid_privacy	Custom		false	Default privacy level
callerid_tag	Custom		false	Internal id_tag for the endpoint
context	String	default	false	Dialplan context for inbound sessions
direct_media_glare_mitigation	Custom	none	false	Mitigation of direct media (re)INVITE glare
direct_media_method	Custom	invite	false	Direct Media method type
connected_line_method	Custom	invite	false	Connected line method type
direct_media	Boolean	yes	false	Determines whether media may flow directly between endpoints.
disable_direct_media_on_nat	Boolean	no	false	Disable direct media session refreshes when NAT obstructs the media session
disallow	Codec		false	Media Codec(s) to disallow

dtmfmode	Custom	rfc4733	false	DTMF mode
external_media_address	String		false	IP used for External Media handling
force_rport	Boolean	yes	false	Force use of return port
ice_support	Boolean	no	false	Enable the ICE mechanism to help traverse NAT
identify_by	Custom	username	false	Way(s) for Endpoint to be identified
mailboxes	String		false	Mailbox(es) to be associated with
mohsuggest	String	default	false	Default Music On Hold class
outbound_auth	Custom		false	Authentication object used for outbound requests
outbound_proxy	String		false	Proxy through which to send requests
rewrite_contact	Boolean	no	false	Allow Contact header to be rewritten with the source IP address-port
rtp_ipv6	Boolean	no	false	Allow use of IPv6 for RTP traffic

rtp_symmetric	Boolean	no	false	Enforce that RTP must be symmetric
send_diversion	Boolean	yes	false	Send the Diversion header, conveying the diversion information to the called user agent
send_pai	Boolean	no	false	Send the P-Asserted-Identity header
send_rpid	Boolean	no	false	Send the Remote-Party-ID header
timers_min_se	Unsigned Integer	90	false	Minimum session timers expiration period
timers	Custom	yes	false	Session timers for SIP packets
timers_sess_expires	Unsigned Integer	1800	false	Maximum session timer expiration period
transport	String		false	Desired transport configuration
trust_id_inbound	Boolean	no	false	Accept identification information received from this endpoint

trust_id_outbound	Boolean	no	false	Send private identification details to the endpoint.
type	None		false	Must be of type 'endpoint'.
use_ptime	Boolean	no	false	Use Endpoint's requested packetisation interval
use_avpf	Boolean	no	false	Determines whether res_pjsip will use and enforce usage of AVPF for this endpoint.
media_encryption	Custom	no	false	Determines whether res_pjsip will use and enforce usage of media encryption for this endpoint.
inband_progress	Boolean	no	false	Determines whether chan_pjsip will indicate ringing using inband progress.
callgroup	Custom		false	The numeric pickup groups for a channel.
pickupgroup	Custom		false	The numeric pickup groups that a channel can pickup.

namedcallgroup	Custom		false	The named pickup groups for a channel.
namedpickupgroup	Custom		false	The named pickup groups that a channel can pickup.
devicestate_busy_at	Unsigned Integer	0	false	The number of in-use channels which will cause busy to be returned as device state
t38udptl	Boolean	no	false	Whether T.38 UDPTL support is enabled or not
t38udptl_ec	Custom	none	false	T.38 UDPTL error correction method
t38udptl_maxdatagram	Unsigned Integer	0	false	T.38 UDPTL maximum datagram size
faxdetect	Boolean	no	false	Whether CNG tone detection is enabled
t38udptl_nat	Boolean	no	false	Whether NAT support is enabled on UDPTL sessions
t38udptl_ipv6	Boolean	no	false	Whether IPv6 is used for UDPTL Sessions

tonezone	String		false	Set which country's indications to use for channels created for this endpoint.
language	String		false	Set the default language to use for channels created for this endpoint.
one_touch_recording	Boolean	no	false	Determines whether one-touch recording is allowed for this endpoint.
recordonfeature	String	automixmon	false	The feature to enact when one-touch recording is turned on.
recordofffeature	String	automixmon	false	The feature to enact when one-touch recording is turned off.
rtpengine	String	asterisk	false	Name of the RTP engine to use for channels created for this endpoint
allowtransfer	Boolean	yes	false	Determines whether SIP REFER transfers are allowed for this endpoint

sdpowner	String	-	false	String placed as the username portion of an SDP origin (o=) line.
sdpsession	String	Asterisk	false	String used for the SDP session (s=) line.
tos_audio	Unsigned Integer	0	false	DSCP TOS bits for audio streams
tos_video	Unsigned Integer	0	false	DSCP TOS bits for video streams
cos_audio	Unsigned Integer	0	false	Priority for audio streams
cos_video	Unsigned Integer	0	false	Priority for video streams
allowsubscribe	Boolean	yes	false	Determines if endpoint is allowed to initiate subscriptions with Asterisk.
subminexpiry	Unsigned Integer	0	false	The minimum allowed expiry time for subscriptions initiated by the endpoint.
fromuser	String		false	Username to use in From header for requests to this endpoint.

mwifromuser	String		false	Username to use in From header for unsolicited MWI NOTIFYs to this endpoint.
fromdomain	String		false	Domain to user in From header for requests to this endpoint.
dtlsverify	Custom		false	Verify that the provided peer certificate is valid
dtlsrekey	Custom		false	Interval at which to renegotiate the TLS session and rekey the SRTP session
dtlscertfile	Custom		false	Path to certificate file to present to peer
dtlsprivatekey	Custom		false	Path to private key for certificate file
dtlscipher	Custom		false	Cipher to use for DTLS negotiation
dtlscafile	Custom		false	Path to certificate authority certificate
dtlscapath	Custom		false	Path to a directory containing certificate authority certificates

dtlssetup	Custom		false	Whether we are willing to accept connections, connect to the other party, or both.
srtp_tag_32	Boolean	no	false	Determines whether 32 byte tags should be used instead of 80 byte tags.

Configuration Option Descriptions

100rel

- no
- required
- yes

aggregate_mwi

When enabled, *aggregate_mwi* condenses message waiting notifications from multiple mailboxes into a single NOTIFY. If it is disabled, individual NOTIFYs are sent for each mailbox.

aors

List of comma separated AoRs that the endpoint should be associated with.

auth

This is a comma-delimited list of *auth* sections defined in `pjsip.conf` to be used to verify inbound connection attempts.

Endpoints without an `authentication` object configured will allow connections without vertification.

callerid

Must be in the format Name <Number>, or only <Number>.

callerid_privacy

- allowed_not_screened
- allowed_passed_screened
- allowed_failed_screened
- allowed
- prohib_not_screened
- prohib_passed_screened
- prohib_failed_screened
- prohib

* `unavailable`

direct_media_glare_mitigation

This setting attempts to avoid creating INVITE glare scenarios by disabling direct media reINVITEs in one direction thereby allowing designated servers (according to this option) to initiate direct media reINVITEs without contention and significantly reducing call setup time.

A more detailed description of how this option functions can be found on the Asterisk wiki https://wiki.asterisk.org/wiki/display/AST/SIP+Direct+Media+Reinvite+Glare+Avoidance

* `none`
* `outgoing`
* `incoming`

direct_media_method

Method for setting up Direct Media between endpoints.

* `invite`
* `reinvite` - Alias for the `invite` value.
* `update`

connected_line_method

Method used when updating connected line information.

* `invite`
* `reinvite` - Alias for the `invite` value.
* `update`

dtmfmode

This setting allows to choose the DTMF mode for endpoint communication.

* `rfc4733` - DTMF is sent out of band of the main audio stream.This supercedes the older **RFC-2833** used within the older `chan_sip`.
* `inband` - DTMF is sent as part of audio stream.
* `info` - DTMF is sent as SIP INFO packets.

identify_by

An endpoint can be identified in multiple ways. Currently, the only supported option is `username`, which matches the endpoint based on the username in the From header.

> **Note**
> Endpoints can also be identified by IP address; however, that method of identification is not handled by this configuration option. See the documentation for the `identify` configu ration section for more details on that method of endpoint identification. If this option is set to `username` and an `identify` configuration section exists for the endpoint, then the endpoint can be identified in multiple ways.

* `username`

rewrite_contact

On inbound SIP messages from this endpoint, the Contact header will be changed to have the source IP address and port. This option does not affect outbound messages send to this endpoint.

timers_min_se

Minimium session timer expiration period. Time in seconds.

timers

- `forced`
- `no`
- `required`
- `yes`

timers_sess_expires

Maximium session timer expiration period. Time in seconds.

transport

This will set the desired transport configuration to send SIP data through.

> **Warning**
> Not specifying a transport will **DEFAULT** to the first configured transport in `pjsip.conf` which is valid for the URI we are trying to contact.

> **Warning**
> Transport configuration is not affected by reloads. In order to change transports, a full Asterisk restart is required

trust_id_inbound

This option determines whether Asterisk will accept identification from the endpoint from headers such as P-Asserted-Identity or Remote-Party-ID header. This option applies both to calls originating from the endpoint and calls originating from Asterisk. If `no`, the configured Caller-ID from pjsip.conf will always be used as the identity for the endpoint.

trust_id_outbound

This option determines whether res_pjsip will send private identification information to the endpoint. If `no`, private Caller-ID information will not be forwarded to the endpoint. "Private" in this case refers to any method of restricting identification. Example: setting *callerid_privacy* to any `prohib` variation. Example: If *trust_id_inbound* is set to `yes`, the presence of a `Privacy: id` header in a SIP request or response would indicate the identification provided in the request is private.

use_avpf

If set to `yes`, res_pjsip will use use the AVPF or SAVPF RTP profile for all media offers on outbound calls and media updates and will decline media offers not using the AVPF or SAVPF profile.

If set to `no`, res_pjsip will use use the AVP or SAVP RTP profile for all media offers on outbound calls and media updates and will decline media offers not using the AVP or SAVP profile.

media_encryption

- `no` - res_pjsip will offer no encryption and allow no encryption to be setup.
- `sdes` - res_pjsip will offer standard SRTP setup via in-SDP keys. Encrypted SIP transport should be used in conjunction with this option to prevent exposure of media encryption keys.
- `dtls` - res_pjsip will offer DTLS-SRTP setup.

inband_progress

If set to `yes`, chan_pjsip will send a 183 Session Progress when told to indicate ringing and will immediately start sending ringing as audio.

If set to `no`, chan_pjsip will send a 180 Ringing when told to indicate ringing and will NOT send it as audio.

callgroup

Can be set to a comma separated list of numbers or ranges between the values of 0-63 (maximum of 64 groups).

pickupgroup

Can be set to a comma separated list of numbers or ranges between the values of 0-63 (maximum of 64 groups).

namedcallgroup

Can be set to a comma separated list of case sensitive strings limited by supported line length.

namedpickupgroup

Can be set to a comma separated list of case sensitive strings limited by supported line length.

devicestate_busy_at

When the number of in-use channels for the endpoint matches the devicestate_busy_at setting the PJSIP channel driver will return busy as the device state instead of in use.

t38udptl

If set to yes T.38 UDPTL support will be enabled, and T.38 negotiation requests will be accepted and relayed.

t38udptl_ec

- `none` - No error correction should be used.
- `fec` - Forward error correction should be used.
- `redundancy` - Redundacy error correction should be used.

t38udptl_maxdatagram

This option can be set to override the maximum datagram of a remote endpoint for broken endpoints.

faxdetect

This option can be set to send the session to the fax extension when a CNG tone is detected.

t38udptl_nat

When enabled the UDPTL stack will send UDPTL packets to the source address of received packets.

t38udptl_ipv6

When enabled the UDPTL stack will use IPv6.

recordonfeature

When an INFO request for one-touch recording arrives with a Record header set to "on", this feature will be enabled for the channel. The feature designated here can be any built-in or dynamic feature defined in features.conf.

> **Note**
> This setting has no effect if the endpoint's one_touch_recording option is disabled

recordofffeature

When an INFO request for one-touch recording arrives with a Record header set to "off", this feature will be enabled for the channel. The feature designated here can be any built-in or dynamic feature defined in features.conf.

> **Note**
> This setting has no effect if the endpoint's one_touch_recording option is disabled

tos_audio

See https://wiki.asterisk.org/wiki/display/AST/IP+Quality+of+Service for more information about QoS settings

tos_video

See https://wiki.asterisk.org/wiki/display/AST/IP+Quality+of+Service for more information about

QoS settings

cos_audio

See https://wiki.asterisk.org/wiki/display/AST/IP+Quality+of+Service for more information about QoS settings

cos_video

See https://wiki.asterisk.org/wiki/display/AST/IP+Quality+of+Service for more information about QoS settings

dtlsverify

This option only applies if *media_encryption* is set to `dtls`.

dtlsrekey

This option only applies if *media_encryption* is set to `dtls`.

If this is not set or the value provided is 0 rekeying will be disabled.

dtlscertfile

This option only applies if *media_encryption* is set to `dtls`.

dtlsprivatekey

This option only applies if *media_encryption* is set to `dtls`.

dtlscipher

This option only applies if *media_encryption* is set to `dtls`.

Many options for acceptable ciphers. See link for more:
http://www.openssl.org/docs/apps/ciphers.html#CIPHER_STRINGS

dtlscafile

This option only applies if *media_encryption* is set to `dtls`.

dtlscapath

This option only applies if *media_encryption* is set to `dtls`.

dtlssetup

This option only applies if *media_encryption* is set to `dtls`.

- `active` - res_pjsip will make a connection to the peer.
- `passive` - res_pjsip will accept connections from the peer.
- `actpass` - res_pjsip will offer and accept connections from the peer.

srtp_tag_32

This option only applies if *media_encryption* is set to `sdes` or `dtls`.

auth

Authentication type

Configuration Option Reference

Option Name	Type	Default Value	Regular Expression	Description
auth_type	Custom	userpass	false	Authentication type
nonce_lifetime	Unsigned Integer	32	false	Lifetime of a nonce associated with this authentication config.
md5_cred	String		false	MD5 Hash used for authentication.
password	String		false	PlainText password used for authentication.
realm	String	asterisk	false	SIP realm for endpoint
type	None		false	Must be 'auth'
username	String		false	Username to use for account

Configuration Option Descriptions

auth_type

This option specifies which of the password style config options should be read when trying to authenticate an endpoint inbound request. If set to `userpass` then we'll read from the 'password' option. For `md5` we'll read from 'md5_cred'.

- `md5`
- `userpass`

md5_cred

Only used when auth_type is `md5`.

password

Only used when auth_type is `userpass`.

domain_alias

Domain Alias

Configuration Option Reference

Option Name	Type	Default Value	Regular Expression	Description
`type`	`None`		`false`	Must be of type 'domain_alias'.
`domain`	`String`		`false`	Domain to be aliased

transport

SIP Transport

Configuration Option Reference

Option Name	Type	Default Value	Regular Expression	Description
`async_operations`	`Unsigned Integer`	`1`	`false`	Number of simultaneous Asynchronous Operations
`bind`	`Custom`		`false`	IP Address and optional port to bind to for this transport
`ca_list_file`	`String`		`false`	File containing a list of certificates to read (TLS ONLY)

cert_file	String		false	Certificate file for endpoint (TLS ONLY)
cipher	Custom		false	Preferred Cryptography Cipher (TLS ONLY)
domain	String		false	Domain the transport comes from
external_media_address	String		false	External Address to use in RTP handling
external_signaling_address	String		false	External address for SIP signalling
external_signaling_port	Unsigned Integer	0	false	External port for SIP signalling
method	Custom		false	Method of SSL transport (TLS ONLY)
localnet	Custom		false	Network to consider local (used for NAT purposes).
password	String		false	Password required for transport
privkey_file	String		false	Private key file (TLS ONLY)
protocol	Custom	udp	false	Protocol to use for SIP traffic

require_client_cert	Custom		false	Require client certificate (TLS ONLY)
type	None		false	Must be of type 'transport'.
verify_client	Custom		false	Require verification of client certificate (TLS ONLY)
verify_server	Custom		false	Require verification of server certificate (TLS ONLY)
tos	Unsigned Integer	0	false	Enable TOS for the signalling sent over this transport
cos	Unsigned Integer	0	false	Enable COS for the signalling sent over this transport

Configuration Option Descriptions

cipher

Many options for acceptable ciphers see link for more: http://www.openssl.org/docs/apps/ciphers.html#CIPHER_STRINGS

method

- `default`
- `unspecified`
- `tlsv1`
- `sslv2`
- `sslv3`
- `sslv23`

localnet

This must be in CIDR or dotted decimal format with the IP and mask separated with a slash ('/').

protocol

- `udp`

- `tcp`
- `tls`
- `ws`
- `wss`

tos

See `https://wiki.asterisk.org/wiki/display/AST/IP+Quality+of+Service` for more information on this parameter.

> **Note**
> This option does not apply to the *ws* or the *wss* protocols.

cos

See `https://wiki.asterisk.org/wiki/display/AST/IP+Quality+of+Service` for more information on this parameter.

> **Note**
> This option does not apply to the *ws* or the *wss* protocols.

contact

A way of creating an aliased name to a SIP URI

Configuration Option Reference

Option Name	Type	Default Value	Regular Expression	Description
`type`	`None`		`false`	Must be of type 'contact'.
`uri`	`String`		`false`	SIP URI to contact peer
`expiration_time`	`Custom`		`false`	Time to keep alive a contact
`qualify_frequency`	`Unsigned Integer`	`0`	`false`	Interval at which to qualify a contact

Configuration Option Descriptions

expiration_time

Time to keep alive a contact. String style specification.

qualify_frequency

Interval between attempts to qualify the contact for reachability. If `0` never qualify. Time in seconds.

aor

The configuration for a location of an endpoint

Configuration Option Reference

Option Name	Type	Default Value	Regular Expression	Description
`contact`	`Custom`		`false`	Permanent contacts assigned to AoR
`default_expiration`	`Unsigned Integer`	`3600`	`false`	Default expiration time in seconds for contacts that are dynamically bound to an AoR.
`mailboxes`	`String`		`false`	Mailbox(es) to be associated with
`maximum_expiration`	`Unsigned Integer`	`7200`	`false`	Maximum time to keep an AoR
`max_contacts`	`Unsigned Integer`	`0`	`false`	Maximum number of contacts that can bind to an AoR
`minimum_expiration`	`Unsigned Integer`	`60`	`false`	Minimum keep alive time for an AoR
`remove_existing`	`Boolean`	`no`	`false`	Determines whether new contacts replace existing ones.

type	None		false	Must be of type 'aor'.
qualify_frequency	Unsigned Integer	0	false	Interval at which to qualify an AoR
authenticate_qualify	Boolean	no	false	Authenticates a qualify request if needed

Configuration Option Descriptions

contact

Contacts specified will be called whenever referenced by `chan_pjsip`.

Use a separate "contact=" entry for each contact required. Contacts are specified using a SIP URI.

mailboxes

This option applies when an external entity subscribes to an AoR for message waiting indications. The mailboxes specified will be subscribed to. More than one mailbox can be specified with a comma-delimited string.

maximum_expiration

Maximium time to keep a peer with explicit expiration. Time in seconds.

max_contacts

Maximum number of contacts that can associate with this AoR. This value does not affect the number of contacts that can be added with the "contact" option. It only limits contacts added through external interaction, such as registration.

> **Note**
> This should be set to `1` and *remove_existing* set to `yes` if you wish to stick with the older `chan_sip` behaviour.

minimum_expiration

Minimum time to keep a peer with an explict expiration. Time in seconds.

remove_existing

On receiving a new registration to the AoR should it remove the existing contact that was registered against it?

Note
This should be set to `yes` and *max_contacts* set to `1` if you wish to stick with the older `chan_sip` behaviour.

qualify_frequency

Interval between attempts to qualify the AoR for reachability. If `0` never qualify. Time in seconds.

authenticate_qualify

If true and a qualify request receives a challenge or authenticate response authentication is attempted before declaring the contact available.

system

Options that apply to the SIP stack as well as other system-wide settings

Configuration Option Reference

Option Name	Type	Default Value	Regular Expression	Description
`timert1`	`Unsigned Integer`	`500`	`false`	Set transaction timer T1 value (milliseconds).
`timerb`	`Unsigned Integer`	`32000`	`false`	Set transaction timer B value (milliseconds).
`compactheaders`	`Boolean`	`no`	`false`	Use the short forms of common SIP header names.
`threadpool_initial_size`	`Unsigned Integer`	`0`	`false`	Initial number of threads in the res_pjsip threadpool.
`threadpool_auto_increment`	`Unsigned Integer`	`5`	`false`	The amount by which the number of threads is incremented when necessary.

threadpool_idle_timeout	Unsigned Integer	60	false	Number of seconds before an idle thread should be disposed of.
threadpool_max_size	Unsigned Integer	0	false	Maximum number of threads in the res_pjsip threadpool. A value of 0 indicates no maximum.
type	None		false	Must be of type 'system'.

Configuration Option Descriptions

timert1

Timer T1 is the base for determining how long to wait before retransmitting requests that receive no response when using an unreliable transport (e.g. UDP). For more information on this timer, see RFC 3261, Section 17.1.1.1.

timerb

Timer B determines the maximum amount of time to wait after sending an INVITE request before terminating the transaction. It is recommended that this be set to 64 * Timer T1, but it may be set higher if desired. For more information on this timer, see RFC 3261, Section 17.1.1.1.

global

Options that apply globally to all SIP communications

Configuration Option Reference

Option Name	Type	Default Value	Regular Expression	Description
maxforwards	Unsigned Integer	70	false	Value used in Max-Forwards header for SIP requests.

`type`	`None`		`false`	Must be of type 'global'.
`useragent`	`String`	`Asterisk PBX SVN-branch-12-r397924`	`false`	Value used in User-Agent header for SIP requests and Server header for SIP responses.

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Configuration_res_pjsip_acl

SIP ACL module

This configuration documentation is for functionality provided by `res_pjsip_acl`.

Overview

ACL

The ACL module used by `res_pjsip`. This module is independent of `endpoints` and operates on all inbound SIP communication using res_pjsip.

It should be noted that this module can also reference ACLs from `acl.conf`.

There are two main ways of creating an access list: `IP-Domain` and `Contact Header`. It is possible to create a combined ACL using both IP and Contact.

pjsip.conf

acl

Access Control List

Configuration Option Reference

Option Name	Type	Default Value	Regular Expression	Description
`acl`	`Custom`		`false`	Name of IP ACL
`contactacl`	`Custom`		`false`	Name of Contact ACL

contactdeny	Custom		false	List of Contact Header addresses to Deny
contactpermit	Custom		false	List of Contact Header addresses to Permit
deny	Custom		false	List of IP-domains to deny access from
permit	Custom		false	List of IP-domains to allow access from
type	None		false	Must be of type 'security'.

Configuration Option Descriptions

acl

This matches sections configured in `acl.conf`

contactacl

This matches sections configured in `acl.conf`

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Configuration_res_pjsip_endpoint_identifier_ip

Module that identifies endpoints via source IP address

This configuration documentation is for functionality provided by `res_pjsip_endpoint_identifier_ip`.

pjsip.conf

identify

Configuration Option Reference

Option Name	Type	Default Value	Regular Expression	Description
`endpoint`	`String`		`false`	Name of Endpoint
`match`	`Custom`		`false`	IP addresses or networks to match against
`type`	`None`		`false`	Must be of type 'identify'.

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Configuration_res_pjsip_outbound_registration

SIP resource for outbound registrations

This configuration documentation is for functionality provided by `res_pjsip_outbound_registration`.

Overview

Outbound Registration

This module allows `res_pjsip` to register to other SIP servers.

pjsip.conf

registration

The configuration for outbound registration

Configuration Option Reference

Option Name	Type	Default Value	Regular Expression	Description
`auth_rejection_permanent`	`Boolean`	`yes`	`false`	Determines whether failed authentication challenges are treated as permanent failures.

client_uri	String		false	Client SIP URI used when attemping outbound registration
contact_user	String		false	Contact User to use in request
expiration	Unsigned Integer	3600	false	Expiration time for registrations in seconds
max_retries	Unsigned Integer	10	false	Maximum number of registration attempts.
outbound_auth	Custom		false	Authentication object to be used for outbound registrations.
outbound_proxy	String		false	Outbound Proxy used to send registrations
retry_interval	Unsigned Integer	60	false	Interval in seconds between retries if outbound registration is unsuccessful
server_uri	String		false	SIP URI of the server to register against
transport	String		false	Transport used for outbound authentication
type	None		false	Must be of type 'registration'.

Configuration Option Descriptions

auth_rejection_permanent

If this option is enabled and an authentication challenge fails, registration will not be attempted again until the configuration is reloaded.

transport

> **Note**
> A *transport* configured in `pjsip.conf`. As with other `res_pjsip` modules, this will use the first available transport of the appropriate type if unconfigured.

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Configuration_res_statsd

Statsd client.

This configuration documentation is for functionality provided by `res_statsd`.

statsd.conf

global

Global configuration settings

Configuration Option Reference

Option Name	Type	Default Value	Regular Expression	Description
`enabled`	`Boolean`	`no`	`false`	Enable/disable the statsd module
`server`	`IP Address`	`127.0.0.1`	`false`	Address of the statsd server
`prefix`	`String`		`false`	Prefix to prepend to every metric

add_newline	Boolean	no	false	Append a newline to every event. This is useful if you want to fake out a server using netcat (nc -lu 8125)

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Configuration_res_xmpp

XMPP Messaging

This configuration documentation is for functionality provided by `res_xmpp`.

xmpp.conf

global

Global configuration settings

Configuration Option Reference

Option Name	Type	Default Value	Regular Expression	Description
debug	Custom	no	false	Enable/disable XMPP message debugging
autoprune	Custom	no	false	Auto-remove users from buddy list.
autoregister	Custom	yes	false	Auto-register users from buddy list
collection_nodes	Custom	no	false	Enable support for XEP-0248 for use with distributed device state

pubsub_autocreate	Custom	no	false	Whether or not the PubSub server supports/is using auto-create for nodes
auth_policy	Custom	accept	false	Whether to automatically accept or deny users' subscription requests

Configuration Option Descriptions

autoprune

Auto-remove users from buddy list. Depending on the setup (e.g., using your personal Gtalk account for a test) this could cause loss of the contact list.

client

Configuration options for an XMPP client

Configuration Option Reference

Option Name	Type	Default Value	Regular Expression	Description
username	String		false	XMPP username with optional resource
secret	String		false	XMPP password
serverhost	String		false	Route to server, e.g. talk.google.com
statusmessage	String	Online and Available	false	Custom status message

pubsub_node	String		false	Node for publishing events via PubSub
context	String	default	false	Dialplan context to send incoming messages to
priority	Unsigned Integer	1	false	XMPP resource priority
port	Unsigned Integer	5222	false	XMPP server port
timeout	Unsigned Integer	5	false	Timeout in seconds to hold incoming messages
debug	Custom	no	false	Enable debugging
type	Custom	client	false	Connection is either a client or a component
distribute_events	Custom	no	false	Whether or not to distribute events using this connection
usetls	Custom	yes	false	Whether to use TLS for the connection or not
usesasl	Custom	yes	false	Whether to use SASL for the connection or not
forceoldssl	Custom	no	false	Force the use of old-style SSL for the connection

keepalive	Custom	yes	false	If enabled, periodically send an XMPP message from this client with an empty message
autoprune	Custom	no	false	Auto-remove users from buddy list.
autoregister	Custom	yes	false	Auto-register users bfrom buddy list
auth_policy	Custom	accept	false	Whether to automatically accept or deny users' subscription requests
sendtodialplan	Custom	no	false	Send incoming messages into the dialplan
status	Custom	available	false	Default XMPP status for the client
buddy	Custom		false	Manual addition of buddy to list

Configuration Option Descriptions

timeout

Timeout (in seconds) on the message stack. Messages stored longer than this value will be deleted by Asterisk. This option applies to incoming messages only which are intended to be processed by the `JABBER_RECEIVE` dialplan function.

autoprune

Auto-remove users from buddy list. Depending on the setup (e.g., using your personal Gtalk account for a test) this could cause loss of the contact list.

status

Can be one of the following XMPP statuses:

- `chat`
- `available`
- `away`
- `xaway`
- `dnd`

buddy

Manual addition of buddy to the buddy list. For distributed events, these budies are automatically added in the whitelist as 'owners' of the node(s).

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Configuration_stasis

Stasis message bus configuration.

This configuration documentation is for functionality provided by `stasis`.

stasis.conf

threadpool

Threadpool configuration.

Configuration Option Reference

Option Name	Type	Default Value	Regular Expression	Description
`initial_size`	`Integer`	`0`	`false`	Initial number of threads in the message bus threadpool.
`idle_timeout_sec`	`Integer`	`20`	`false`	Number of seconds for an idle thread to be disposed of.
`max_size`	`Integer`	`200`	`false`	Maximum number of threads in the threadpool.

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Configuration_udptl

This configuration documentation is for functionality provided by `udptl`.

udptl.conf

global

Global options for configuring UDPTL

Configuration Option Reference

Option Name	Type	Default Value	Regular Expression	Description
`udptlstart`	`Unsigned Integer`	`4000`	`false`	The start of the UDPTL port range
`udptlend`	`Unsigned Integer`	`4999`	`false`	The end of the UDPTL port range
`udptlchecksums`	`Boolean`	`yes`	`false`	Whether to enable or disable UDP checksums on UDPTL traffic
`udptlfecentries`	`Unsigned Integer`		`false`	The number of error correction entries in a UDPTL packet
`udptlfecspan`	`Unsigned Integer`		`false`	The span over which parity is calculated for FEC in a UDPTL packet
`use_even_ports`	`Boolean`	`no`	`false`	Whether to only use even-numbered UDPTL ports
`t38faxudpec`	`Custom`		`false`	Removed

t38faxmaxdatagram	Custom		false	Removed

Import Version

This documentation was imported from Asterisk Version Unknown

Asterisk 12 Installation and Configuration

The following pages contain installation and configuration information specific to Asterisk 12.

For general Asterisk configuration and installation instructions, see Installing Asterisk and Configuration and Operation.

Configuring res_pjsip

Overview

This page is intended to help an Asterisk administrator understand configuration fundamentals for the new SIP resources and channel driver included with Asterisk 12. It covers an explanation of configuration for pjsip.conf which configures the SIP resource modules utilized by the chan_pjsip driver.

If you are looking for info on how to configure sip.conf (the config used by the older SIP channel driver for Asterisk), then you'll want to go here Creating SIP Accounts for some basic info or check out the sip.conf sample file included in the /configs directory of your Asterisk source files.

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 (implying you installed their dependencies as well)
- You understand basic Asterisk concepts. Including the role of extensions.conf (dialplan) in your overall Asterisk configuration.

If you don't know anything about Asterisk yet, then you should probably start at the Getting Started section.

Quick Start

If you like to play before you read or 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 SIP trunks or phones.
- Look at the Full res_pjsip configuration examples by scenario 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 this page, the sample pjsip.conf, or the reference documentation if you get confused.

pjsip.conf Explained

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

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.

> **Option Values and Defaults**
> **How do I know what values I can use for an option?** Use the built-in configuration help at the CLI or view the wiki section listing all config option help text. You can use "config show help res_pjsip <configobject> <configoption>" to get help on a particular option. The output will typically describe the **default** value for an option as well. **Link to list of config options goes here, once we have them pulled onto the wiki**
>
> **Defaults:** For many config options, it's very helpful to understand their default behavior. For example, endpoint's "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.

ENDPOINT (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 (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.

> **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=
privkey_file=
ca_list_file=
cipher=
method=
```

AUTH (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 (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 (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
```

DOMAIN_ALIAS (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 (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 (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]
endpoint=6001
match=203.0.113.1
```

CONTACT (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.

Config Section Help and Defaults

Once we have the XML configuration help pulled onto the wiki we'll put a link here to that wiki section.

In the meantime use the built-in configuration help to your advantage. You can use "config show help res_pjsip <configobject> <configoption>" to get help on a particular option. That help will typically describe the default value for an option as well.

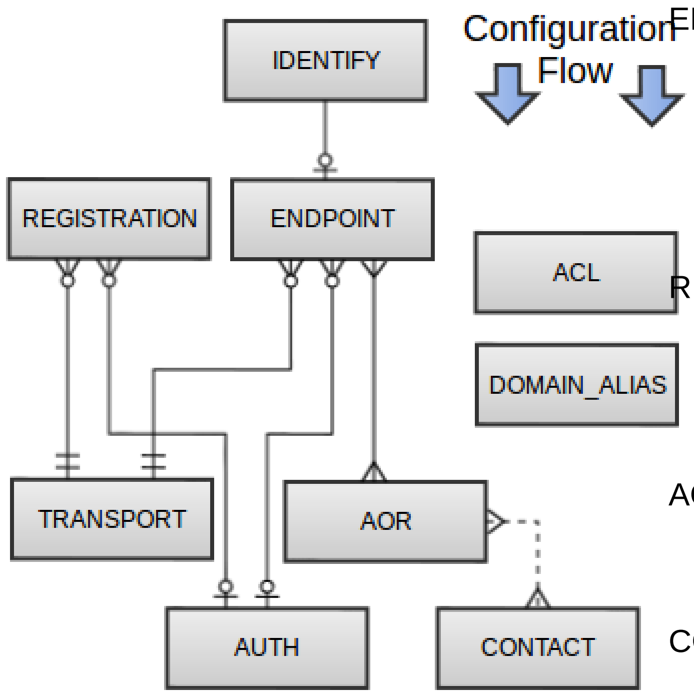
Relationships of Configuration Objects in pjsip.conf

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.

Full res_pjsip configuration examples by scenario

Below are some sample configurations to demonstrate various scenarios with complete pjsip.conf files.

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
transport=simpletrans
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

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
transport=simpletrans
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:203.0.113.1:5060

[mytrunk]
type=endpoint
transport=simpletrans
context=from-external
disallow=all
allow=ulaw
outbound_auth=mytrunk
aors=mytrunk

[mytrunk]
type=identify
endpoint=mytrunk
match=203.0.113.1
```

- "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

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
transport=simpletrans
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.

Dialing using chan_pjsip

Dialing from dialplan in extensions.conf

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 contacts associated.

```
exten => _6XXX,1,Dial(PJSIP/${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)
```

Old to New - sip.conf to pjsip.conf example comparison

We want to provide some examples of what similar configurations would look like between the old sip.conf and the new pjsip.conf

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

There is also a script available to provide a basic conversion of a sip.conf config to a pjsip.conf config. ADD **A LINK TO SIP.CONF to PJSIP.CONF SCRIPT WHEN READY**

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
<pre>[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</pre>	<pre>[simpletrans] type=transport protocol=udp bind=0.0.0.0 [6001] type = endpoint transport = simpletrans 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 transport = simpletrans 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</pre>

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
<pre>[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</pre>	<pre>[simpletrans] type=transport protocol=udp bind=0.0.0.0 [mytrunk] type=registration transport=simpletrans outbound_auth=mytrunk server_uri=sip:myaccountname@203.0.11 3.1:5060 client_uri=sip:myaccountname@192.0.2. 1:5060 [mytrunk] type=auth auth_type=userpass password=1234567890 username=myaccountname [mytrunk] type=aor contact=sip:203.0.113.1:5060 [mytrunk] type=endpoint transport=simpletrans context=from-external disallow=all allow=ulaw outbound_auth=mytrunk aors=mytrunk [mytrunk] type=identify endpoint=mytrunk match=203.0.113.1</pre>

Installing pjproject

Asterisk 12 contains a new SIP stack that is based on pjproject. Some changes in pjproject were needed for Asterisk 12. The largest change were modifications made to the pjproject build system that allowed it to build shared object libraries. Unlike Asterisk 11 - which embedded pjproject in order to utilize its ICE/STUN/TURN libraries in its RTP engine - Asterisk 12 dynamically links to pjproject.

As a result, currently, versions of pjproject that can be downloaded from www.pjsip.org will **not** w ork with Asterisk 12. This includes the version of pjproject embedded in Asterisk 11.

The Asterisk 12 compatible version of pjproject is available on github, or - depending on your Linux distribution - available as a package. This wiki page provides detailed instructions on building and install pjproject for Asterisk 12.

> 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.

Building and Installing pjproject from Source

Downloading pjproject

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

   > Downloading and installing `git` are beyond the scope of these instructions.

2. Checkout pjproject from the github repo:

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

   And that's it!

Building and Installing pjproject

pjproject embeds a number of third party libraries which can conflict with installed versions of those libraries. Thus, building pjproject is highly dependent on your distribution of Linux as well as what third party libraries are already installed on your system. A number of configuration options are available to disable these libraries in pjproject and custom tailor it to your system. The table below outlines common ones that may be needed for a typical installation.

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.

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.

1. In the pjproject source directory:

```
# ./configure --prefix=/usr --enable-shared
```

> You **must** specify `--enable-shared` in order to build the shared objects for Asterisk. `--prefix` should be set to the base path of the `lib` directory where the shared objects will be installed.

2. Build pjproject:

```
# make dep
  ...
# make
  ...
     output/sample-x86_64-unknown-linux-gnu/vid_streamutil.o
-L/home/mjordan/projects/pjproject/pjlib/lib
-L/home/mjordan/projects/pjproject/pjlib-util/lib
-L/home/mjordan/projects/pjproject/pjnath/lib
-L/home/mjordan/projects/pjproject/pjmedia/lib
-L/home/mjordan/projects/pjproject/pjsip/lib
-L/home/mjordan/projects/pjproject/third_party/lib        -lpjsua
-lpjsip-ua -lpjsip-simple -lpjsip -lpjmedia-codec -lpjmedia
-lpjmedia-videodev -lpjmedia-audiodev -lpjnath -lpjlib-util
-lmilenage -lsrtp -lresample -lgsmcodec -lspeex -lilbccodec
-lg7221codec -lportaudio  -lpj -lm -luuid -lm -lnsl -lrt -lpthread
-lasound -lcrypto -lssl
make[2]: Leaving directory
`/home/mjordan/projects/pjproject/pjsip-apps/build'
make[1]: Leaving directory
`/home/mjordan/projects/pjproject/pjsip-apps/build'
```

3. Install pjproject

```
# make install
...
dev -lpjnath -lpjlib-util  -lmilenage -lsrtp -lresample -lgsmcodec
-lspeex -lilbccodec -lg7221codec -lportaudio  -lpj -lm -luuid -lm
-lnsl -lrt -lpthread  -lasound -lcrypto -lssl!" | \
 sed -e "s!@PJ_INSTALL_CFLAGS@!-I/usr/include -DPJ_AUTOCONF=1 -O2
-DPJ_IS_BIG_ENDIAN=0 -DPJ_IS_LITTLE_ENDIAN=1 -fPIC!" >
//usr/lib/pkgconfig/libpjproject.pc
```

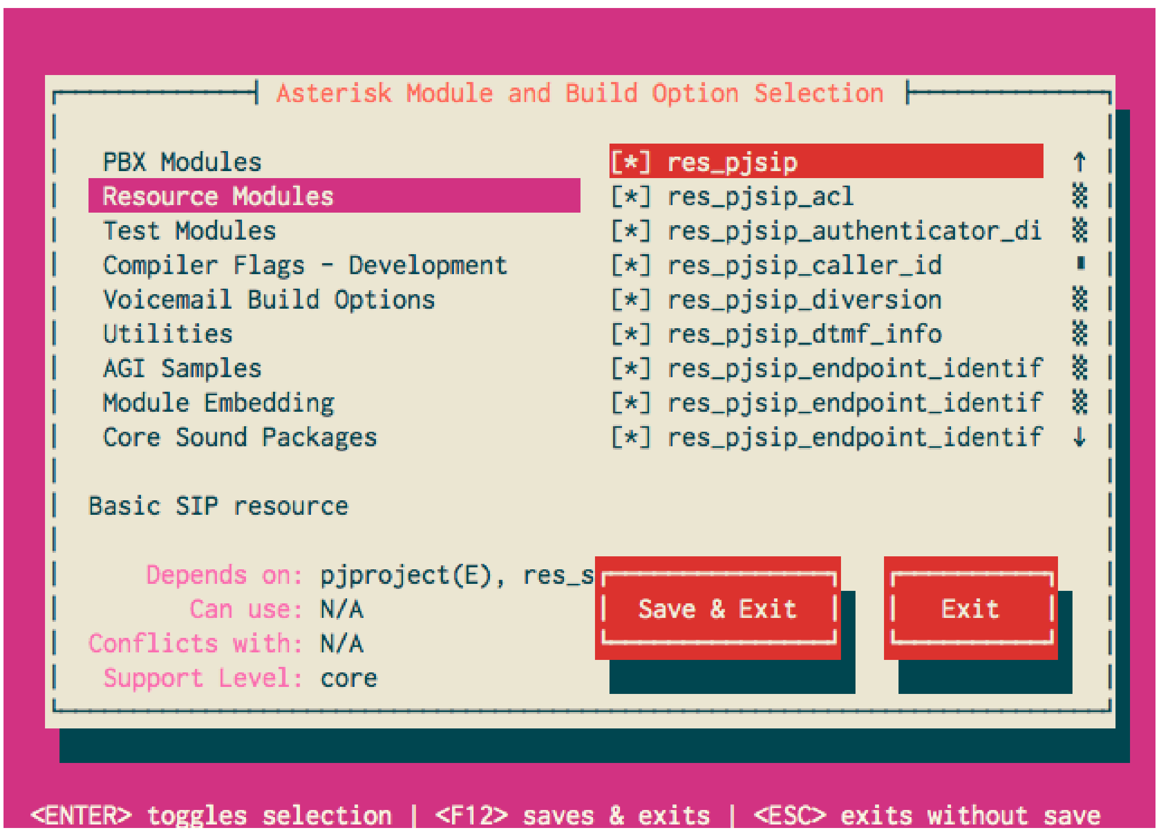
4. Update shared library links and verify that pjproject has been installed in the target location:

```
# ldconfig
# 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
```

5. Finally, verify that Asterisk detects the pjproject libraries. In your Asterisk 12 source directory:

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

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



Issues and Workarounds

<table>
<tr><th>Issue</th><th>Resolution</th></tr>
<tr><td>When building pjproject, errors about an ARM codec are displayed:

<pre>
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
...
</pre>
</td><td>You already have the AMR codec installed. Run <code>configure</code> with the <code>--disable-opencore-amr</code> option specified.</td></tr>
</table>

<table>
<tr><td>When building pjproject, linker errors referring to various video methods are displayed:
<pre>/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'</pre></td><td>Run <code>configure</code> with either or both <code>--disable-video</code> or <code>--disable-v4l2</code></td></tr>
<tr><td>After building pjproject, the dump provided by <code>ldconfig -p</code> doesn't display any libraries.</td><td>Run <code>ldconfig</code> to re-configure dynamic linker run-time bindings</td></tr>
</table>

<table>
<tr>
<td>After building and installing pjproject, Asterisk fails to detect any of the libraries (the entries cannot be selected in menuselect)</td>
<td>

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
```

1. Make sure you have `pkg-config` installed on your system.
2. 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.

</td>
</tr>
</table>

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, it is recommended that any static libraries be removed prior to installing the compatible version of pjproject.

pjproject provides an `uninstall` make target that will remove previous installations:

```
make uninstall
```

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
```

Asterisk 12 Specifications

The following specifications are provided for Asterisk 12:

- Asterisk Manager Interface (AMI) version 2.0
- Asterisk 12 Call Detail Records (CDR)
- Asterisk 12 Channel Event Logging (CEL)

These specifications are only valid for Asterisk 12 and should not be used as a basis for any other version of Asterisk.

AMI 1.4 Specification

> ⚠ This Specification is still in draft form.

1. Introduction

This Asterisk Manager Interface (AMI) specification describes the relationship between Asterisk and an external entity wishing to communicate with Asterisk over the AMI protocol. It describes:

- An overview of the AMI protocol
- The operations AMI provides external entities wishing to control Asterisk
- Basic formatting of AMI message structures
- Guaranteed operations, configuration control, and other information provided by Asterisk in AMI 1.4

1.1. Scope

This specification describes AMI version 1.4 for Asterisk 12. This specification provides details on the functional, operational and design requirements for AMI 1.4. Note that this does not include a comprehensive listing of the AMI configuration file parameters or messages that a system interfacing over AMI in Asterisk 12 will send/receive; however, it does provide a baseline of the supported features and messages provided in AMI 1.4. This specification should be used in conjunction with the documented AMI actions and events in Asterisk 12 to encompass the full range of functionality provided by AMI in Asterisk 12.

In addition, this specification provides interface requirements levied on AMI by Stasis Core. It conveys sufficient detail to understand how AMI attaches to Stasis Core and interacts with other entities on the Stasis Core message bus.

This specification is intended for all parties requiring such information, including software developers, system designers and testers responsible for implementing the interface.

2. Terminology

Term	Definition
Action	A command issued to Asterisk from an external entity via AMI

Client	An external entity communicating with Asterisk via AMI over some transport mechanism
Event	A message sent from Asterisk to an external entity via AMI
Field	A key/value pair that exists in either an action or event
Stasis-Core	The internal framework that AMI is built on top of

3. Protocol Overview

Asterisk provides a number of interfaces that serve different purposes. Say, for example, we wanted to manipulate a call between Alice and Bob via some external mechanism. Depending on what we wanted to do with the call, we may use one or more interfaces to manipulate the channels that make up the call between Alice and Bob.

Alice calls Bob and...	Interface
... we want to use a local script to execute some logic on Alice's channel	AGI
... we want to execute a script on a remote machine on Bob's channel	FastAGI
... we want to put Alice into an IVR with fine grained media control, where the IVR is written outside of `extensions.conf`	ExternalIVR
... we want to control Alice and Bob's underlying channel objects	AMI (with, possibly, AsyncAGI as well)

In general, AMI is used to manage Asterisk and its channels. It does not determine what actions are executed on a particular channel - the dialplan and some AGI interface does that - but it does allow a client to control call generation, aspects of call flow, and other internals of Asterisk.

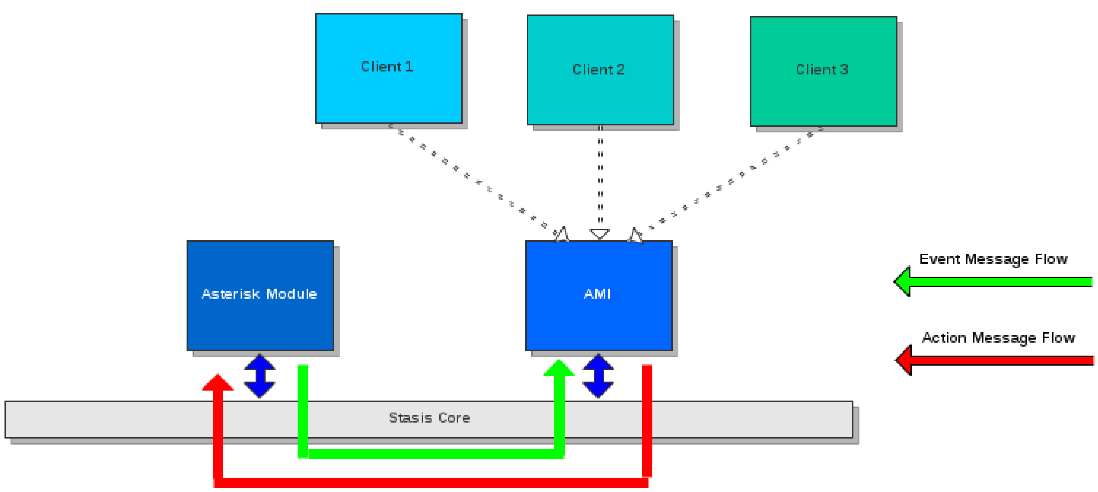
At its heart, AMI is an asynchronous message bus: it spills **events** that contain information about the Asterisk system over some transport. In response, clients may request that Asterisk takes some **action**. These two concepts - actions and events - make up the core of what is AMI. As AMI is asynchronous, as events occur in Asterisk they are immediately sent to the clients. This means that actions issued by entities happen without any synchronization with the events being received, even if those events occur in response to an action. It is the responsibility of entities to associate event responses back to actions.

Clients wishing to use AMI act as clients and connect to Asterisk's AMI server over a supported transport mechanism. Authentication may or may not be enabled, depending on the configuration. Once connected, events can be automatically spilled to the connected clients, or limited in a variety of fashions. A connected client can send an action to the AMI server at any time. Depending on the allowed authorizations, the action may be allowed or disallowed.

More information on the various ways a client can be configured can be seen in AMI Configuration.

> Sometimes, the term **command** may be used instead of the term **action**. With respect to AMI actions, command is synonymous with action, and the two can be treated the same. For the sake of consistency, we've attempted to use the term **action** where possible.

Historically, AMI has existed in Asterisk as its own core component `manager`. AMI events were raised throughout Asterisk encoded in an AMI specific format, and AMI actions were processed and passed to the functions that implemented the logic. In Asterisk 12, AMI has been refactored to sit on top of Stasis Core, which is a generic, protocol independent message bus internal to Asterisk. From the perspective of clients wishing to communicate with Asterisk over AMI very little has changed; internally, the Stasis representation affords a much higher degree of flexibility with how messages move through Asterisk.



Asterisk's Stasis-Core provides a generic publish/subscribe message bus inside of Asterisk. While AMI often directly interacts with constructs in Asterisk through actions, it receives its events through messages published over Stasis-Core. It translates the generic Stasis messages into an AMI event, and sends those to the appropriate AMI clients.

4. Semantics and Syntax

4.1. Message Sending and Receiving

By default, AMI is an asynchronous protocol that sends events immediately to clients when those events are available. Likewise, clients are free to send actions to AMI at any time, which may or may not trigger additional events. The exception to this is when the connection is over HTTP/HTTPS; in that scenario, events are only transmitted as part of the response to an HTTP POST.

Various options for configuration of clients can control which events are sent to a client. Events can be whitelisted/blacklisted explicitly via event filters, or implicitly by class authorizations.

4.2. Message Layout

AMI is an ASCII protocol that provides bidirectional communication with clients. Each field in an AMI message - action or event - is delineated by the '\r\n' characters. An action or event is terminated by an additional '\r\n' character. Within a message, each field is a key value pair delineated by a ':'. A single space MUST follow the ':' and precede the value.

```
Event: Newchannel
Privilege: call,all
Channel: SIP/misspiggy-00000001
Uniqueid: 1368479157.3
ChannelState: 3
ChannelStateDesc: Up
CallerIDNum: 657-5309
CallerIDName: Miss Piggy
ConnectedLineName:
ConnectedLineNum:
AccountCode: Pork
Priority: 1
Exten: 31337
Context: inbound
```

This is syntantically equivalent to the following ASCII string:

```
Event: Newchannel\r\nPrivilege:
call,all\r\nChannel:
SIP/misspiggy-00000001\r\n
Uniqueid:
1368479157.3\r\nChannelState:
3\r\n
ChannelStateDesc:
Up\r\nCallerIDNum: 657-5309\r\n
CallerIDName: Miss
Piggy\r\nConnectedLineName:\r\nC
onnectedLineNum:\r\nAccountCode:
Pork\r\n
Priority:\r\nExten:
31337\r\nContext:
inbound\r\n\r\n
```

Actions are specified in a similar manner. Note that depending on the message, some keys can be repeated.

```
Action: Originate
ActionId: SDY4-12837-123878782
Channel: SIP/kermit
Context: outbound
Exten: s
Priority: 1
CallerID: "Kermit the Frog" <123-4567>
Account: FrogLegs
Variable: MY_VAR=frogs
Variable: HIDE_FROM_CHEF=true
```

In addition, no ordering is implied on message specific keys. Hence, the following two messages are semantically the same.

```
Action: Originate
ActionId: SDY4-12837-123878782
Channel: SIP/kermit
Context: outbound
Exten: s
Priority: 1
CallerID: "Kermit the Frog"
<123-4567>
Account: FrogLegs
Variable: MY_VAR=frogs
Variable: HIDE_FROM_CHEF=true
```

```
ActionId: SDY4-12837-123878782
Action: Originate
Variable: HIDE_FROM_CHEF=true
Variable: MY_VAR=frogs
Channel: SIP/kermit
Account: FrogLegs
Context: outbound
Exten: s
CallerID: "Kermit the Frog"
<123-4567>
Priority: 1
```

This is also true for events, although by convention, the `Event` key is the first key in the event. If an action or event contains duplicate keys, such as `Variable`, the order in which Asterisk processes said keys is the order in which they occur within the action or event.

Keys are case insensitive. Hence, the following keys are equivalent:

```
Action:
Originate
```

```
ACTION:
Originate
```

```
action:
Originate
```

The case sensitivity for values is left up to the context in which they are interpreted.

4.2.1. Common Fields

4.2.1.1. Actions

4.2.1.1.1. General Fields

This section lists fields that apply generally to all actions.

4.2.1.1.1.1. Action

Action specifies the action to execute within Asterisk. Each value corresponds to a unique action to execute within Asterisk. The value of the **Action** field determines the allowed fields within the rest of the message. By convention, the first field in any action is the **Action** field.

4.2.1.1.1.2. ActionId

ActionId is a universal unique identifier that can optionally be provided with an action. If provided in an action, events that are related to that action will contain the same **ActionId** value, allowing a client to associate actions with events that were caused by that action.

It is recommended that clients always provide an **ActionId** for each action they submit.

4.2.1.1.2. Channels

This section lists fields that apply generally to all actions that interact upon an Asterisk channel. Note that an Action that interacts with a channel must typically supply the **Channel** field.

> **Upgrade Tip**
> In the past, AMI clients would have to contend with channel rename events. As Asterisk will now not change the name of a channel during its lifetime, this is no longer necessary.

4.2.1.1.2.1. Channel

The Asterisk channel name. A channel name is provided by AMI to clients during a **Newchannel** event. A channel name can be viewed as the handle to a channel.

4.2.1.1.2.2. Uniqueid

A universal unique identifier for the channel. In systems with multiple Asterisk instances, this field can be used to construct a globally unique identifier for a channel, as a channel name may occur multiple times across Asterisk instances.

4.2.1.2. Events

4.2.1.2.1. General Fields

This section lists fields that apply generally to all events.
4.2.1.2.1.1. Event

The unique name of the event being raised. The value of the **Event** field determines the rest of the contents of the message. By convention, the **Event** field is the first field in an AMI message.
4.2.1.2.1.2. ActionId

If present, the Action's corresponding ActionId that caused this event to be created. If an Action contained an **ActionId**, any event relating the success or failure of that action MUST contain an **ActionId** field with the same value.
4.2.1.2.1.3. Privilege

The class authorizations associated with this particular event. The class authorizations for a particular event are in a comma-delineated list. For more information, see class authorizations.

4.2.1.2.2. Channels

This section lists fields that apply generally to all events that occur due to interactions upon an Asterisk channel.

Events that relate multiple channels will prefix these fields with an event specific role specifier. For example, a **DialBegin** or a **DialEnd** event will prefix the outbound channel's fields with **Dest**. So, the **Channel** field is the **DestChannel** field; the **Uniqueid** field is the **DestUniqueid** field, etc.

4.2.1.2.2.1. Channel

The current Asterisk channel name. This corresponds to the Channel field in actions.

4.2.1.2.2.2. Uniqueid

A universal unique identifier for the channel. This corresponds to the Uniqueid field in actions.

4.2.1.2.2.3. ChannelState

The current state of the channel, represented as an integer value. The valid values are:

Value	State	Description
0	Down	Channel is down and available
1	Rsrvd	Channel is down, but reserved
2	OffHook	Channel is off hook
3	Dialing	The channel is in the midst of a dialing operation
4	Ring	The channel is ringing
5	Ringing	The remote endpoint is ringing. Note that for many channel technologies, this is the same as Ring.
6	Up	A communication path is established between the endpoint and Asterisk
7	Busy	A busy indication has occurred on the channel
8	Dialing Offhook	Digits (or equivalent) have been dialed while offhook

9	Pre-ring	The channel technology has detected an incoming call and is waiting for a ringing indication
10	Unknown	The channel is an unknown state

Depending on the underlying channel technology, not all states will be used. Channels typically begin in either the Down or Up states.

4.2.1.2.2.4. ChannelStateDesc

The text description of the channel state. This will be one of the State descriptions in the table in ChannelState.

4.2.1.2.2.5. CallerIDNum

The current caller ID number. If the caller ID number is not known, the string "<unknown>" is returned instead.

4.2.1.2.2.6. CallerIDName

The current caller ID name. If the caller ID name is not known, the string "<unknown>" is returned instead.

4.2.1.2.2.7. ConnectedLineNum

The current connected line number. If the connected line number is not known, the string "<unknown>" is returned instead.

4.2.1.2.2.8. ConnectedLineName

The current connected line name. If the connected line name is not known, the string "<unknown>" is returned instead.

4.2.1.2.2.9. AccountCode

The channel's accountcode.

4.2.1.2.2.10. Context

The current context in the dialplan that the channel is executing in.

4.2.1.2.2.11. Exten

The current extension in the dialplan that the channel is executing in.

4.2.1.2.2.12. Priority

The current priority of the current context, extension in the dialplan that the channel is executing in.

4.2.1.2.2.13. ChanVariable

Channel variables specific to a channel can be conveyed in each AMI event related to that channel. When this occurs, each variable is referenced in a **ChanVariable** field. The value of a **ChanVariable** field will always be of the form `key=value`, where `key` is the name of the channel variable and `value` is its value.

4.2.1.2.3. Bridges

4.2.1.2.3.1. BridgeUniqueid

A unique identifier for the bridge, which provides a handle to actions that manipulate bridges.

4.2.1.2.3.2. Bridgetype

The type of the bridge. Bridge types can vary based on the registered bridge technologies; in general, however, this will indicate whether the bridge is a holding bridge, a simple two-party bridge, or a multi-party bridge. The type of a bridge can change during the lifetime of the bridge as channels enter and leave a bridge.

4.2.1.2.4. Action Responses

When an Action is submitted to AMI, the success or failure of the action is communicated in subsequent events.

4.2.1.2.4.1. Response

Contains whether or not the action succeeded or failed. Valid values are "Success" or "Error". Events that are in response to an action MUST include this field.

4.2.1.2.4.2. EventList

Some actions will cause a chain of events to be created. Events that are a response to an action that causes such a sequence will contain the EventList field with a value of "start". When all generated events have been sent, a final event will be sent containing the EventList field with the value "complete".

If, for some reason, an error occurs and the events cannot be sent, an event will be sent with an EventList field that contains the value "cancelled".

Note that the events that mark the completion or cancellation of an event list are not technically action responses, and have their own specific event types.

4.2.1.2.4.3. Message

An optional text message that provides additional contextual information regarding the success or failure of the action.

4.3. Actions

The supported actions for Asterisk 12 are listed here:

Asterisk 12 AMI Actions

While new AMI Actions may be added over the lifetime of Asterisk 12, existing AMI Actions will **n**

ot be removed.

4.4. Events

The supported events for Asterisk 12 are listed here:

Asterisk 12 AMI Events

While new AMI Events may be added over the lifetime of Asterisk 12, existing AMI Events will **no t** be removed.

4.5. Channel Interaction/Lifetime

While channels are independent of AMI, they have a large implication on the events sent out over AMI. Many of the events in AMI correspond to changes in channel state. While AMI is an asynchronous protocol, there is some ordering with respect to the events that are relayed for a particular channel. This section provides the basic event relationships that are guaranteed through AMI.

4.5.1. Basic Channel Lifetime

All channels begin with a **Newchannel** event. A **Newchannel** will always contain the following fields:

- The current Channel name that acts as a handle to the channel for that channel's lifetime for a single Asterisk system
- The Uniqueid for the channel, that allows systems to have a globally unique identifier for the channel

Changes in the state of the channel, i.e., the ChannelState field, are conveyed via **Newstate** eve nts.
Notification of a Channel being disposed of occurs via a **Hangup** event. A **Hangup** signals the termination of the channel associated with the Uniqueid. After the **Hangup** event, no further events will be raised in relation to the channel with that Uniqueid, and the communication between the endpoint and Asterisk via that channel is terminated.

Example	
`` Event: Newchannel Privilege: dialplan,all Channel: SIP/kermit-00000001 Uniqueid: asterisk-1368479157.1 ChannelState: 0 ChannelStateDesc: Down ... ```	• Kermit the Frog's SIP channel is created. The is the first event for `SIP/kermit-00000001` and indicates a path of communication being opened up between Asterisk and Kermit's SIP device.
`` Event: Newstate Privilege: dialplan,all Channel: SIP/kermit-00000001 Uniqueid: asterisk-1368479157.1 ChannelState: 6 ChannelStateDesc: Up ... ```	• Kermit the Frog's SIP channel's state changes from Down to Up
`` Event: Hangup Privilege: dialplan,all Channel: SIP/kermit-00000001 Uniqueid: asterisk-1368479157.1 ChannelState: 6 ChannelStateDesc: Up Cause: 16 Cause-txt: Normal Clearing ... ```	• Kermit the Frog's SIP channel is hung up. At this point, no further events for `SIP/kermit-00000001` will be sent.

4.5.2. Channel Variables

For each channel variable that is changed, a **VarSet** event is sent to the client. The **VarSet** event contains the new value of the appropriate channel variable. Note that channel variables can also be conveyed in ChanVariable fields.

4.5.3. DTMF

DTMF is indicated via a **DTMFBegin**/**DTMFEnd** events. A **DTMFEnd** event MUST convey the duration of the DTMF tone in milliseconds.

⚠

Behavioral Change Warning
The combination of **DTMFBegin**/**DTMFEnd** events replaces the removed **DTMF** event.

4.5.4. Dialplan Execution

As a channel executes operations in the dialplan, those operations are conveyed via a **NewExten** event. Each transition to a new combination of context, extension, and priority will trigger a **NewExten** event.

Example	
`Event: Newexten` `Privilege: dialplan,all` `Channel: SIP/kermit-00000001` `Uniqueid: asterisk-1368479157.1` `...` `Context: default` `Extension: h` `Priority: 1` `Application: NoOp` `AppData: Ah Snap`	Kermit the Frog's SIP channel has been hung up, and he's been tossed rudely into the *h* extension. This event informs the clients that Kermit's channel is in context *default*, extension *h*, priority *1*, and is about to execute the *NoOp* application with application data *"Ah Snap"*.

4.5.5. Dialing and Origination

Dial operations always result in two events: a **DialBegin** event that signals the beginning of the dial to a particular destination, and a **DialEnd** event that signals the end of the dialing. In parallel dialing situations, **DialBegin**/**DialEnd** events MUST be sent for each channel dialed. For each **DialBegin** event sent, there MUST be a corresponding **DialEnd** event.

In dialing situations with a caller and a called party, the **DialBegin** and **DialEnd** events convey information about both channels. The calling channel uses the standard channel field names, while the called party's field names are prefixed with "Dest". In dialing situations where there is no caller, such as when Asterisk originates an outbound call via a call file, only the called channel is represented in the events. The channel field names are still prefixed with "Dest" in this case.

A **DialEnd** occurs whenever Asterisk knows the final state of the channel that it was attempting to establish. This is communicated in the DialStatus field.

Behavioral Change Warning
The pair of **DialBegin**/**DialEnd** events replaces the deprecated **Dial** event. Note that the **Dial** event signalling the end of dialing would not normally be sent until after bridging was complete; this operation should now occur when the dial operation has determined the status of a particular called channel.

Simple Successful Dial	
<pre>Event: Newchannel Channel: SIP/animal-00000002 Uniqueid: asterisk-1368479160.5 ... Event: DialBegin Channel: SIP/kermit-00000001 Uniqueid: asterisk-1368479155.1 DestChannel: SIP/animal-00000002 DestUniqueid: asterisk-1368479160.5 ... Event: DialEnd Channel: SIP/kermit-00000001 Uniqueid: asterisk-1368479155.1 DestChannel: SIP/animal-00000002 DestUniqueid: asterisk-1368479160.5 DialStatus: ANSWER ...</pre>	In this example, Kermit decides to dial Animal. A new channel between Asterisk and Animal's SIP device is created and conveyed via a **Newchannel** event, and then a dial operation is begun. Note that in the **DialBegin** event, Kermit's SIP device is the caller as he initiated the dial operation, while Animal's SIP device is the destination. As such, the fields referring to Animal's SIP channel are prefixed with "Dest". When Animal eats his handset (causing the device to think he merely took it off the hook), the SIP device answers and the dial operation completes. This indicated by a **DialEnd** event. At this point, the channel is ready for something - it can execute in the dialplan, or be immediately bridged with the calling channel.

Simple Failed Dial	
<pre>Event: Newchannel Channel: SIP/animal-00000003 Uniqueid: asterisk-1368479199.1 ... Event: DialBegin Channel: SIP/kermit-00000001 Uniqueid: asterisk-1368479150.1 DestChannel: SIP/animal-00000003 DestUniqueid: asterisk-1368479199.1 ... Event: DialEnd Channel: SIP/kermit-00000001 Uniqueid: asterisk-1368479150.1 DestChannel: SIP/animal-00000003 DestUniqueid: asterisk-1368479199.1 DialStatus: TIMEDOUT ... Event: Hangup Channel: SIP/animal-00000003 Uniqueid: asterisk-1368479199.1 ...</pre>	In this example, we decide to dial Animal again. Unfortunately, Animal ate his handset, so we inevitably time out. When we do, a **DialEnd** event indicates the failure condition in the DialStatus field.

Parallel Dial	
<pre>Event: Newchannel Channel: SIP/animal-00000003 Uniqueid: asterisk-1368479150.3 ... Event: Newchannel Channel: SIP/drteeth-00000004 Uniqueid: asterisk-1368479150.4 ...</pre>	

	Event: DialBegin Channel: SIP/kermit-00000001 Uniqueid: asterisk-1368479150.0 DestChannel: SIP/animal-00000003 DestUniqueid: asterisk-1368479150.3 ... Event: DialBegin Channel: SIP/kermit-00000001 Uniqueid: asterisk-1368479150.0 DestChannel: SIP/drteeth-00000004 DestUniqueid: asterisk-1368479150.4 ... Event: DialEnd Channel: SIP/kermit-00000001 Uniqueid: asterisk-1368479150.0 DestChannel: SIP/drteeth-00000004 DestUniqueid: 5asterisk-1368479150.4 DialStatus: ANSWER ... Event: DialEnd Channel: SIP/kermit-00000001 Uniqueid: asterisk-1368479150.0 DestChannel: SIP/animal-00000003 DestUniqueid: asterisk-1368479150.3	In this example, Kermit decides to dial Animal and Dr. Teeth. Dr. Teeth immediately answers, and so we cancel the dial to Animal. We can now do something with Dr. Teeth's channel (such as bridge him with his dentist) - however, Animal's channel is implicitly destroyed, as his device never answered.

```
DialStatus: CANCEL
...
```

4.5.6. Bridging

A bridge contains 0 or more channels. When a channel is in a bridge, it has the potential to communicate with other channels within the bridge. Before channels enter a bridge, a **BridgeCreate** event is sent, indicating that a bridge has been created. When a bridge is destroyed, a **BridgeDestroy** event is sent. All channels within a bridge must leave a bridge prior to the **BridgeDestroy** event being sent.

When a channel enters a bridge, a **BridgeEnter** event is raised. When a channel is put into a bridge, it is implied that the channel can pass media between other channels in the bridge. This is not guaranteed, as other properties on the channel or bridge may restrict media flow. For example, bridges of the "holding type" implicitly restrict the media flow between channels. Likewise, media may be restricted in multi-party conference bridges based on user role permissions, such as when a conference leader mutes all participants in a conference. The **BridgeEnter** event does indicate, however, that a potential relationship between channels in a bridge exists.

When a channel leaves a bridge, a corresponding **BridgeLeave** event is raised. A **BridgeLeave** event MUST mean that the channel that left the bridge can no longer pass media to other channels still in the bridge. This does not necessarily mean that the channel is being hung up; rather, that it is no longer in a communication path with some other set of channels.

In all cases, if a channel has a **BridgeEnter** event, it MUST have a corresponding **BridgeLeave** event. If a channel is hung up and it is in a bridge, a **BridgeLeave** event MUST precede the **Hangup** event.

If a transfer operation begins, a **Transfer** event MUST be raised for the channels involved in the transfer prior to those channels leaving the bridge or entering another bridge. Similarly, if a channel enters a parking lot, a **ParkedCall** event MUST be raised for the channel prior to it entering the bridge that represents the parking lot.

If a property of a bridge is changed, such as the type going from a simple two-party bridge to a multi-party bridge, then the **BridgeUpdate** event is sent with the updated parameters.

4.5.6.1. Two Party Bridging

Parties are bridged by virtue of them entering a bridge, as indicated by a **BridgeEnter**. When parties are no longer talking, a **BridgeLeave** event is sent for each channel that leaves the bridge.

Example - Two Party Bridge

Events	Description
`Event: BridgeCreate` `Bridgetype: core` `BridgeUniqueid: 1234` `...` `Event: BridgeEnter` `Bridgetype: core` `BridgeUniqueid: 1234` `Channel: SIP/kermit-00000001` `Uniqueid: asterisk-1368479150.0` `...`	Kermit the Frog's SIP channel enters into Bridge 1234. As a result, the bridge is first created (denoted by the **BridgeCreate** event), and then Kermit's channel enters the bridge (the **BridgeEnter** event)
`Event: BridgeEnter` `Bridgetype: core` `BridgeUniqueid: 1234` `Channel: SIP/gonzo-00000002` `Uniqueid: asterisk-1368479150.1` `...`	Gonzo the Great enters the bridge and talks with Kermit. Note that the bridge Gonzo entered is Bridge 1234; by virtue of this being the same bridge Kermit entered, we know that the two can talk.
`Event: BridgeLeave` `Bridgetype: core` `BridgeUniqueid: 1234` `Channel: SIP/kermit-00000001` `Uniqueid: asterisk-1368479150.0` `...` `Event: Hangup` `Channel: SIP/kermit-00000001` `Uniqueid: asterisk-1368479150.0` `...`	Kermit decided he was tired of Gonzo's pontificating, and hung up on Gonzo. We are first alerted that Kermit has left the bridge; quickly thereafter, we receive the **Hangup** event indicating that Kermit's channel is dead.

Event: BridgeLeave Bridgetype: core BridgeUniqueid: 1234 Channel: SIP/gonzo-00000002 Uniqueid: asterisk-1368479150.1 ... Event: Hangup Channel: SIP/gonzo-00000002 Uniqueid: asterisk-1368479150.1 ... Event: BridgeDestroy Bridgetype: core BridgeUniqueid: 1234 ...	Asterisk is configured to not let Gonzo continue on in the dialplan once his bridge is broken. As such, Gonzo is forcibly ejected from the bridge, and is hung up on after. Because no channels are left in the bridge, the bridge is destroyed.

In this scenario, it was perfectly acceptable for either Kermit or Gonzo's channels to continue after the bridge was broken. Since this represents the most basic two-party call scenario, once one party decided to hang up, the other party was also hung up on.

4.5.6.2. Transfers

Transfers begin with a **Transfer** event, which indicates information about the transfer that is about to take place. Subsequently, **BridgeLeave**/**BridgeEnter** events are used to indicate which channels are talking at different stages during the transfer.

Example - Blind Transfer	
Event: BridgeCreate Bridgetype: core BridgeUniqueid: 1234 ... Event: BridgeEnter Bridgetype: core BridgeUniqueid: 1234 Channel: SIP/kermit-00000001 Uniqueid: asterisk-1368479150.0 ...	Kermit the Frog's SIP channel enters into Bridge 1234

`` Event: BridgeEnter Bridgetype: core BridgeUniqueid: 1234 Channel: SIP/fozzie-00000002 Uniqueid: asterisk-1368479150.2 ... ```	Fozzie Bear's SIP channel enters into Bridge 1234. At this point, Fozzie and Kermit can talk to each other.
`` Event: Transfer Privilege: call,all TransferMethod: Core TransferType: Blind Channel: SIP/fozzie-00000002 Uniqueid: 550e8400-e29b-41d4-a716-446655440001 TargetChannel: SIP/kermit-00000001 TargetUniqueid: 550e8400-e29b-41d4-a716-446655440000 TransferExten: call_miss_piggy TransferContext: default ```	Fozzie decides he's tired of telling Kermit jokes and blind transfers him off to Miss Piggy via the dialplan extension `call_miss_piggy`. The fact that a transfer is occurring starts off with the **Transfer** event. This tells the user who initiates the transfer, who is being transferred, and where the transfer target is going.
`` Event: BridgeLeave Bridgetype: core BridgeUniqueid: 1234 Channel: SIP/kermit-00000001 ... ```	Kermit leaves the bridge with Fozzie.

At this point, the order of the events that affect Fozzie versus Kermit are not defined. Fozzie and Kermit are no longer bridged together, and their respective events may arrive in any order.

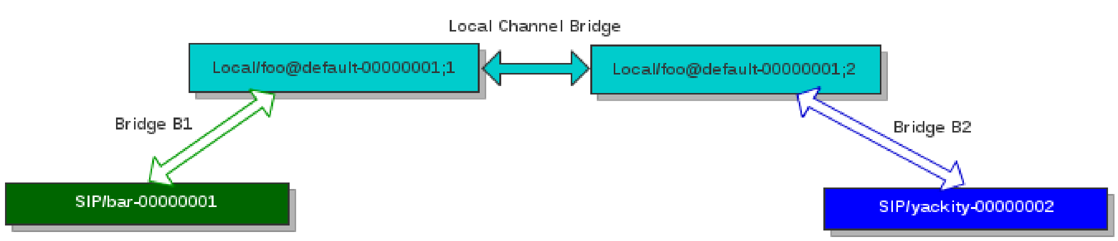
Event: BridgeLeave Bridgetype: core BridgeUniqueid: 1234 Channel: SIP/fozzie-00000002 ... Event: BridgeDestroy Bridgetype: core BridgeUniqueid: 1234 ...	Because Fozzie isn't talking to anyone anymore, he leaves the bridge as well. At this point Asterisk could hang up Fozzie's channel, or, if configured, he could continue on in the dialplan (say, perhaps, to talk to his rubber chicken).
Event: Newchannel Channel: SIP/miss_piggy-00000003 Uniqueid: 550e8400-e29b-41d4-a716-446 655440002 ... Event: DialBegin Channel: SIP/miss_piggy-00000003 Uniqueid: 550e8400-e29b-41d4-a716-446 655440002 ... Event: DialEnd Channel: SIP/miss_piggy-00000003 Uniqueid: 550e8400-e29b-41d4-a716-446 655440002 DialStatus: ANSWER ...	The act of Kermit entering into extension `call_miss_piggy` ca uses a new channel to be created between Asterisk and Miss Piggy. Asterisk immediately dials Miss Piggy, who answers.

<pre>Event: BridgeCreate Bridgetype: core BridgeUniqueid: 1235 ... Event: BridgeEnter Bridgetype: core BridgeUniqueid: 1235 Channel: SIP/kermit-00000001 Uniqueid: 550e8400-e29b-41d4-a716-446 655440000 ...</pre>	Kermit enters into a bridge. Note that the **BridgeUniqueid** is different than the previous bridge's **BridgeUniqueid**.
<pre>Event: BridgeEnter Bridgetype: core BridgeUniqueid: 1235 Channel: SIP/miss_piggy-00000003 Uniqueid: 550e8400-e29b-41d4-a716-446 655440002 ...</pre>	Miss Piggy enters the bridge. At this point, Kermit and Miss Piggy can converse, and the blind transfer is complete.

4.5.7. Local Channels

Local channels are different than 'normal' channels in Asterisk, in that they have several properties that make them unique:

- A Local channel never has a physical device that it directly communicates with - it exists as a virtual channel within Asterisk.
- A Local channel can communicate with one or more channels, or directly with an Asterisk application.
- Local channels can be optimized out of a communication path, causing the state of bridges and channels to change.
- A Local channel is actually always two separate channels with a special bridge between them.



While the Local channel exists as a single concept from the perspective of the dialplan, from the perspective of an AMI client, it is always two channels that are tied together by a special bridge. When the bridge between the Local channel halves is formed, a **LocalBridge** event is raised denoting that the two are connected.

4.5.7.1. Local Channel Creation

When a Local channel is created, a **Newchannel** event is created for each half of the Local channel being created. Each Local channel half has its own Channel field name and its own Uniqueid - for the purposes of an AMI client, they are two separate channels that can be manipulated separately but whose lifetime is tied together. When either Local channel half is hungup, denoted by a **Hangup** event, the other Local channel half will also be hungup automatically.

4.5.7.2. Local Channel Bridging

Local channels receive bridge events in the same fashion as other channels. Thus, when they join a bridge, they receive a **BridgeEnter** event and when they leave a bridge, a corresponding **BridgeLeave** event. The exception to this is a Local channel bridge, denoted by a **LocalBridge** event. Once a **LocalBridge** event occurs, the two Local channel halves can pass frames back and forth between each other. Because the lifetime of the Local channel halves is the same, there is no event representing a Local channel bridge being broken.

Example - Local Channel between two SIP Channels

```
Event: Newchannel
Channel:
SIP/miss_piggy-00000003
Uniqueid:
550e8400-e29b-41d4-a716-446
655440002
...
```

Miss Piggy decides to call Kermit the Frog, and a channel is created between Asterisk and Miss Piggy's SIP device.

```
Event: Newchannel
Channel:
Local/kermit@default-000000
01;1
Uniqueid:
550e8400-e29b-41d4-a716-446
655440003
...
Event: Newchannel
Channel:
Local/kermit@default-000000
01;2
Uniqueid:
550e8400-e29b-41d4-a716-446
655440004
...
Event: LocalBridge
Channel1:
Local/kermit@default-000000
01;1
Channel2:
Local/kermit@default-000000
01;2
Uniqueid1:
550e8400-e29b-41d4-a716-446
655440003
Uniqueid2:
550e8400-e29b-41d4-a716-446
655440004
LocalOptimization: No
...Event: DialBegin
Channel:
Local/kermit@default-000000
01;1
Uniqueid:
550e8400-e29b-41d4-a716-446
655440003
...
```

Instead of dialing Kermit the Frog directly, a Local channel is created inside Asterisk and Asterisk dials one half of the Local channel. Note that both halves of the Local channel are created first; the Local channel bridges its two halves together; then a **Dial Begin** event indicates that Asterisk is attempting to connect to one half of the Local channel, `Local/kermit@default-00000001;1`.

```
Event: DialEnd
Channel:
Local/kermit@default-000000
01;1
Uniqueid:
550e8400-e29b-41d4-a716-446
655440003
DialStatus: ANSWER
...
Event: BridgeCreate
Bridgetype: core
BridgeUniqueid: 1234
...
Event: BridgeEnter
Bridgetype: core
BridgeUniqueid: 1234
Channel:
SIP/miss_piggy-00000003
Uniqueid:
550e8400-e29b-41d4-a716-446
655440002
...
Event: BridgeEnter
Bridgetype: core
BridgeUniqueid: 1234
Channel:
Local/kermit@default-000000
01;1
Uniqueid:
550e8400-e29b-41d4-a716-446
655440003
...
```

The dial operation succeeds and Miss Piggy is bridged with one half of the Local channel that will eventually connect her to Kermit. Note that Miss Piggy doesn't realize that any changes have occurred - from her perspective, nothing has really happened, as the Local channel is a virtual channel and we haven't yet dialed Kermit.

```
Event: Newchannel
Channel:
SIP/kermit-00000002
Uniqueid:
550e8400-e29b-41d4-a716-446
655440005
...
Event: DialBegin
Channel:
SIP/kermit-00000002
Uniqueid:
550e8400-e29b-41d4-a716-446
655440005
...
Event: DialEnd
Channel:
SIP/kermit-00000002
Uniqueid:
550e8400-e29b-41d4-a716-446
655440005
DialStatus: ANSWER
...
Event: BridgeCreate
Bridgetype: core
BridgeUniqueid: 5678
...
Event: BridgeEnter
Bridgetype: core
BridgeUniqueid: 5678
Channel:
Local/kermit@default-000000
01;2
Uniqueid:
550e8400-e29b-41d4-a716-446
655440004
...
Event: BridgeEnter
Bridgetype: core
BridgeUniqueid: 5678
Channel:
SIP/kermit-00000002
Uniqueid:
550e8400-e29b-41d4-a716-446
655440005
...
```

We dial Kermit, and he foolishly answers. He is now bridged with the second half of the Local channel, `Local/kermit@default-00000001;2`, as both that channel and the SIP channel communicating with his SIP device have corresponding **BridgeEnter** events.

> In those situations where both a Local channel half and a SIP channel are bridged, there is no ordering guaranteed on which channel enters the bridge first - however, both a Local channel half and the SIP channel will be bridged in the same bridge.

```
Event: BridgeLeave
Bridgetype: core
BridgeUniqueid: 1234
Channel:
```

```
SIP/miss_piggy-00000003
Uniqueid:
550e8400-e29b-41d4-a716-446
655440002
...
Event: Hangup
Channel:
SIP/miss_piggy-00000003
Uniqueid:
550e8400-e29b-41d4-a716-446
655440002
...
Event: BridgeLeave
Bridgetype: core
BridgeUniqueid: 1234
Channel:
Local/kermit@default-000000
01;1
Uniqueid:
550e8400-e29b-41d4-a716-446
655440003
...
Event: Hangup
Channel:
Local/kermit@default-000000
01;1
Uniqueid:
550e8400-e29b-41d4-a716-446
655440003
...
Event: BridgeDestroy
Bridgetype: core
BridgeUniqueid: 1234
...
Event: BridgeLeave
Bridgetype: core
BridgeUniqueid: 5678
Channel:
Local/kermit@default-000000
01;2
Uniqueid:
550e8400-e29b-41d4-a716-446
655440004
...
Event: Hangup
Channel:
Local/kermit@default-000000
01;2
Uniqueid:
550e8400-e29b-41d4-a716-446
655440004
...
Event: BridgeLeave
```

	`` Bridgetype: core BridgeUniqueid: 5678 Channel: SIP/kermit-00000002 Uniqueid: 550e8400-e29b-41d4-a716-446655440005 ... Event: Hangup Channel: SIP/kermit-00000002 Uniqueid: 550e8400-e29b-41d4-a716-446655440005 ... Event: BridgeDestroy ```	Miss Piggy, in a fit of frustrated porkine anger, karate chops her SIP phone, causing it to hangup. She leaves the bridge with the Local channel half, `Local/kermit@default-00000001;1`. This causes it to be hungup, as no one else is in the bridge. Because `Local/kermit@default-00000001;1` is hungup, its corresponding other Local channel half, `Local/kermit@default-00000001;2`, leaves the bridge with Kermit's SIP phone and is also hung up. Kermit's SIP channel realizes it's all by itself (it's not easy being green), leaves the bridge it had with its Local channel half, and hangs up as well.

```
Bridgetype: core
BridgeUniqueid: 5678
```

4.5.7.3. Local Channel Optimization

Local channels have an option wherein they can be optimized away if there are two other channels in the Answered state on either end of the Local channel half. This option is performed on Local channel creation, and is communicated back to the AMI clients in the **LocalBridge** even t in the LocalOptimization field. When a Local channel optimization occurs, a **LocalOptimization** event is sent that indicates the channels involved in the optimization and any bridges they happen to be in. If, after the Local channel optimization, either bridges contains only a single channel, then a single channel in that bridge is moved to the bridge that had the other channel half.

> It is not defined which channel is moved first - in this situation, that is an implementation detail left up to Asterisk. Suffice to say, if a Local channel is optimized away, Asterisk attempts to rebridge the channels left over.

Example - Optimizing Local Channel between two SIP Channels

```
Event: Newchannel
Channel: SIP/gonzo-00000001
Uniqueid:
550e8400-e29b-41d4-a716-446
655440000
...
```

Gonzo decides to call Kermit the Frog, and a channel is created between Asterisk and Gonzo's SIP device.

```
Event: Newchannel
Channel:
Local/kermit@default-000000
01;1
Uniqueid:
550e8400-e29b-41d4-a716-446
655440001
...
Event: Newchannel
Channel:
Local/kermit@default-000000
01;2
Uniqueid:
550e8400-e29b-41d4-a716-446
655440002
...
Event: DialBegin
Channel:
Local/kermit@default-000000
01;1
Uniqueid:
550e8400-e29b-41d4-a716-446
655440001
...
```

Instead of dialing Kermit the Frog directly, a Local channel is created inside Asterisk and Asterisk dials it.

```
Event: DialEnd
Channel:
Local/kermit@default-000000
01;1
Uniqueid:
550e8400-e29b-41d4-a716-446
655440001
DialStatus: ANSWER
...
Event: BridgeCreate
Bridgetype: core
BridgeUniqueid: 1234
...
Event: BridgeEnter
Bridgetype: core
BridgeUniqueid: 1234
Channel: SIP/gonzo-00000001
Uniqueid:
550e8400-e29b-41d4-a716-446
655440000
...
Event: BridgeEnter
Bridgetype: core
BridgeUniqueid: 1234
Channel:
Local/kermit@default-000000
01;1
Uniqueid:
550e8400-e29b-41d4-a716-446
655440001
...
Event: LocalBridge
Channel1:
Local/kermit@default-000000
01;1
Channel2:
Local/kermit@default-000000
01;2
Uniqueid1:
550e8400-e29b-41d4-a716-446
655440001
Uniqueid2:
550e8400-e29b-41d4-a716-446
655440002
LocalOptimization: Yes
...
```

The dial operation succeeds and Gonzo is bridged with one half of the Local channel that will eventually connect him (it?) to Kermit. The Local channel itself has also decided to go ahead and bridge the two local halves together, since we have one half of the full chain of channels established. Note that we are told at this point that the Local channel will attempt to optimize itself away, if Kermit answers.

```
Event: DialBegin
Channel:
SIP/kermit-00000002
Uniqueid:
550e8400-e29b-41d4-a716-446
655440003
...
Event: DialEnd
Channel:
SIP/kermit-00000002
Uniqueid:
550e8400-e29b-41d4-a716-446
655440003
DialStatus: ANSWER
...
Event: BridgeCreate
Bridgetype: core
BridgeUniqueid: 5678
...
Event: BridgeEnter
Bridgetype: core
BridgeUniqueid: 5678
Channel:
Local/kermit@default-000000
01;2
Uniqueid:
550e8400-e29b-41d4-a716-446
655440002
...
Event: BridgeEnter
Bridgetype: core
BridgeUniqueid: 5678
Channel:
SIP/kermit-00000002
Uniqueid:
550e8400-e29b-41d4-a716-446
655440003
...
```

We dial Kermit and he answers. He is now bridged with the second half of the Local channel, `Local/kermit@default-00000001;2`.

> Note that even in the optimizing case, both the Local channel halves and the other channels involved first enter their respective bridges.

```
Event: LocalOptimization
Channel1:
Local/kermit@default-000000
01;1
Uniqueid1:
550e8400-e29b-41d4-a716-446
655440001
BridgeUniqueid1: 1234
Channel2:
Local/kermit@default-000000
01;2
Uniqueid2:
550e8400-e29b-41d4-a716-446
655440002
BridgeUniqueid2: 5678
...
Event: BridgeLeave
Bridgetype: core
BridgeUniqueid: 5678
Channel:
SIP/kermit-00000002
Uniqueid:
550e8400-e29b-41d4-a716-446
655440003
...Event: BridgeEnter
Bridgetype: core
BridgeUniqueid: 1234
Channel:
SIP/kermit-00000002
Uniqueid:
550e8400-e29b-41d4-a716-446
655440003
...
```

Asterisk determines that it can optimize away the Local channel. It notifies the AMI client that this is about to occur, which Local channels are involved, and what bridges they are currently in. It starts first by moving Kermit into Gonzo's bridge. From their perspective, nothing has happened - they just now happen to be in the same bridge, instead of having a Local channel pass frames for them.

Note that before this happens, a **LocalOptimization** event is sent indicating that some shenanigans are about to ensue with the Local channels and by extension, the bridges they are in.

<pre>Event: BridgeLeave Bridgetype: core BridgeUniqueid: 5678 Channel1: Local/kermit@default-000000 01;2 Uniqueid1: 550e8400-e29b-41d4-a716-446 655440002 ... Event: BridgeLeave Bridgetype: core BridgeUniqueid: 1234 Channel1: Local/kermit@default-000000 01;1 Uniqueid1: 550e8400-e29b-41d4-a716-446 655440001 ... Event: Hangup Channel1: Local/kermit@default-000000 01;1 Uniqueid1: 550e8400-e29b-41d4-a716-446 655440001 ... Event: Hangup Channel1: Local/kermit@default-000000 01;2 Uniqueid1: 550e8400-e29b-41d4-a716-446 655440002 ...Event: BridgeDestroy Bridgetype: core BridgeUniqueid: 5678 ...</pre>	The Local channels now leave their bridges and are hung up. As a result, the bridge with no channels left in it - Bridge 5678 - is destroyed.

4.5.8. Masquerades

Masquerades have changed

In the past, masquerades occurred rather frequently - most often in any scenario where a transfer occurred or where a `pbx_thread` needed to be associated with a channel. This has now changed. Masquerades now rarely occur, and are never communicated to AMI clients. From the perspective of AMI clients, nothing changes - you still use your handle to a channel to communicate with it, regardless of the presence (or lack thereof) of a masquerade operation.

This note only exists to explicitly call out that fact.

4.6. Transports

AMI supports the following transport mechanisms:

- TCP/TLS
- HTTP/HTTPS

When clients connect over HTTP/HTTPS, AMI events are queued up for retrieval. Events queued up for a client are automatically retrieved and sent in the response to any POST operation. The **WaitEvent** action can be used to wait for and retrieve AMI events.

5. Security Considerations

AMI supports security at the transport level via TLS using OpenSSL.

5.1. Class Authorizations

Do not rely on class authorizations for security. While they provide a means to restrict a client's access to sets of functionality, there are often ways of achieving similar functionality through multiple mechanisms. Do **NOT** assume that because a class authorization has not been granted to a client, that they can't find a way around it. In general, view class authorizations as a coarse grained way of providing sets of filters.

Events and actions are automatically classified with particular class authorizations. Clients can be configured to support some set of class authorizations, filtering the actions that they can perform and events that they receive. The supported class authorizations are listed below.

Class Type	Description
system	The item is associated with something that reports on the status of the system or manipulates the system in some fashion
call	The item is associated with calls, i.e., state changes in a call, etc.
log	The item is associated with the logging subsystem

verbose	The item is associated with verbose messages
command	The item is associated with execution of CLI commands through AMI
agent	The item is associated with Queue Agent manipulation
user	The item is associated with user defined events
config	The item is associated with manipulating the configuration of Asterisk
dtmf	The item is associated with DTMF manipulation
reporting	The item is associated with querying information about the state of the Asterisk system
cdr	The item is associated with CDR manipulation
dialplan	The item is associated with dialplan execution
originate	The item is associated with originating a channel
agi	The item is associated with AGI execution
cc	The item is associated with call completion
aoc	The item is associated with Advice of Charge
test	The item is associated with some test action
message	The item is associated with out of call messaging
all	The item has all class authorizations associated with it
none	The item has no class authorization associated with it

5.2. Access Control Lists

Access Control Lists can be used to filter connections based on address. If an attempt to connect from an unauthorized address is detected, the connection attempt will be rejected.

5.3. Authorization

Authorization can be provided via the **Login** action. If a client fails to provide a valid username/password, the connection attempt and any subsequent actions will be rejected. Events will not be sent until the client provides authorized credentials.

The actions that are excluded from successful login are:

- **Login**
- **Logout**
- **Challenge**

> ⚠ The action **Logoff** has been replaced by a corresponding action **Logout**. **Logoff** has been deprecated in favor of **Logout**.

6. AMI Configuration

AMI supports the following configuration options. Note that additional configurations MAY be specified; however, these configuration options are valid for Asterisk 12.

6.1. General Settings

Option	Type	Description	Default
enabled	Boolean	Enable AMI	no
webenabled	Boolean	Enable AMI over HTTP/HTTPS	no
port	Integer	The port AMI's TCP server will bind to	5038
bindaddr	IP Address	The address AMI's TCP server will bind to	0.0.0.0
tlsenable	Boolean	Enable TLS over TCP	no
tlsbindaddr	IP Address	The address AMI's TCP/TLS server will bind to	0.0.0.0:5039
tlscertfile	String	The full path to the TLS certificate to use	/tmp/asterisk.pem

tlsprivatekey	String	The full path to the private key. If no path is specified, *tlscertfile* will be used for the private key.	/tmp/private.pem
tlscipher	String	The string specifying which SSL ciphers to use. Valid SSL ciphers can be found at http://www.openssl.org/docs/apps/ciphers.html#CIPHER_STRINGS	
allowmultiplelogin	Boolean	Allow multiple logins for the same user. If set to no, multiple logins from the same user will be rejected.	Yes
timestampevents	Boolean	Add a Unix epoch **Timestamp** field to all AMI events	No
authlimit	Integer	The number of unauthenticated clients that can be connected at any time	

6.2. Client Settings

Note that the name of the client settings context is the username for the client connection.

Option	Type	Description	Default
secret	String	The password that must be provided by the client via the **Login** action	

deny	ACL	An address/mask to deny in an ACL. This option may be present multiple times.	
permit	ACL	An address/mask to allow in an ACL. This option may be present multiple times.	
acl	String	A Named ACL to apply to the client.	
setvar	String	A channel variable key/value pair (using the nomenclature VARIABLE=value) that will be set on all channels originated from this client	

eventfilter	Regular Expression	This option may be present multiple times. This options allows clients to whitelist or blacklist events. A filter is assumed to be a whitelist unless preceeded by a '!'. Evaluation of the filters is as follows: • If no filters are configured all events are reported as normal. • If there are white filters only: implied black all filter processed first, then white filters. • If there are black filters only: implied white all filter processed first, then black filters. • If there are both white and black filters: implied black all filter processed first, then white filters, and lastly black filters.	
read	String	A comma delineated list of the allowed class authorizations applied to events	all
write	String	A comma delineated list of the allowed class authorizations applied to actions	all

tag	String	If present, the client will only receive events that are related to channels created in relation to endpoints that are configured with the same tag. See Publish/Subscribe for more information.	

Asterisk 12 CDR Specification

Asterisk 12 has not yet been released - this specification is still in draft form.

Introduction

Call Detail Records, or CDRs, have been around in Asterisk for a long, long time. In fact, portions of it were taken from the `Zapata` library, as noted in `cdr.c`:

```
*
 * \note Includes code and algorithms from the Zapata library.
 *
```

CDRs have always attempted to provide the billing information between two parties involved in a call. While their simplicity has been a big advantage in quickly deploying a billing system, it has also been their largest disadvantage when trying to express complex scenarios. As Asterisk evolved and adapted to more complex scenarios, it became difficult to express the full details of a call within a single CDR. Compounding this problem, the way in which bridging was performed in Asterisk made call scenarios involving more than two parties incredibly difficult to track and coalesce into one or more CDRs. Numerous attempts were made to address the various problems in CDRs with varying degrees of success.

In Asterisk 12, changes in the bridging architecture necessitated a substantial upgrade to CDR behavior. The largest changes are:

- Individual components in Asterisk no longer modify CDRs directly. CDRs are produced based on the state of the channels and bridges within Asterisk. As a result, there is a much greater degree of consistency in CDRs throughout Asterisk, regardless of the application channels happen to be executing in.
- The behavior of CDRs between multiple parties is now defined.
- Depending on how channels are dialed and bridged, multiple CDRs will be created for a given call. Post-processing of these records **will** be required to determine the overall statistics of the call.

It is important to note that in Asterisk 1.8, Channel Event Logging (CEL) was introduced as an alternative to CDRs. Unlike CDRs, where a relatively few records are produced for a call, CEL provides a sequence of events regarding the state of channels within Asterisk. CELs provide substantially more information about what is occurring to the channels involved in a call, thus allowing an Asterisk user to construct their own billing system by handling the events as they choose. While CEL will never supplant CDRs, they are an option if CDRs do not provide the billing information you need or in the format you require. While improvements have been made and some complex scenarios defined in this specification, the fact that CEL can provide a more robust billing system is still true as of Asterisk 12.

Asterisk Users moving to Asterisk 12 are highly encouraged to read this specification carefully.

Scope

This CDR specification applies to Asterisk 12. It does not cover CDRs in prior versions of Asterisk.

Note that this does not include a comprehensive analysis as to how CDRs can be produced in all call scenarios in Asterisk. It defines the behavior for common scenarios, but certain scenarios are deliberately left unspecified. The behavior of CDRs in said scenarios is **undefined and is not a bug.**

It is known that the behavior of CDRs will not allow all applications to capture the billing requirements for their systems. If CDRs cannot meet the requirements of your application, Channel Event Logging (CEL) provides call information at a much finer granularity, allowing complex billing systems to be constructed. Please see the Asterisk 12 CEL Specification for more information on CEL.

Terminology

Term	Definition
CDR	Call Detail Record. The thing that this document is attempting to describe.
CEL	Channel Event Logging. An alternative way to get billing information from Asterisk, that is significantly more flexible and powerful than CDRs but requires the billing logic to be completely implemented by the user.
Party A	A CDR always involves two parties. One party is always chosen as the 'owner' of the CDR. The CDR reflects the view of the call from that party.
Party B	A CDR always involves two parties. Party B is the target of the call.

Stasis	Stasis is the internal message bus in Asterisk that conveys state to the CDR engine.

CDR Overview

A CDR is a record of communication between one or two parties. As such, a single CDR always addresses the communication between two parties: a Party A and a Party B. The CDR reflects the view of the call from the perspective of Party A, while Party B is the party that Party A is communicating with. Each CDR includes the following times:

- Start time - the time at which the CDR was created for Party A
- Answer time - the time at which Party A and Party B could begin communicating
- End time - the time at which Party A and Party B could no longer communicate

From these times, two durations are computed:

- Duration - the End time minus the Start time.
- Billsec - the End time minus the Answer time. (Whether or not you actually bill for this period of time is up to you)

A single CDR only tracks information about a single path of communication between two endpoints. In many scenarios, there will be multiple paths of communication between multiple parties, even in a single "call". Each path of communication results in a new CDR, each representing the communication between two endpoints. All of the CDRs involved are associated by virtue of a special linked identifier field, `linkedid`. The CDRs themselves, however, typically do **not** aggregate the time between records. It is up to billing systems to determine which CDRs should be used for their billing records, and add up the times/durations themselves.

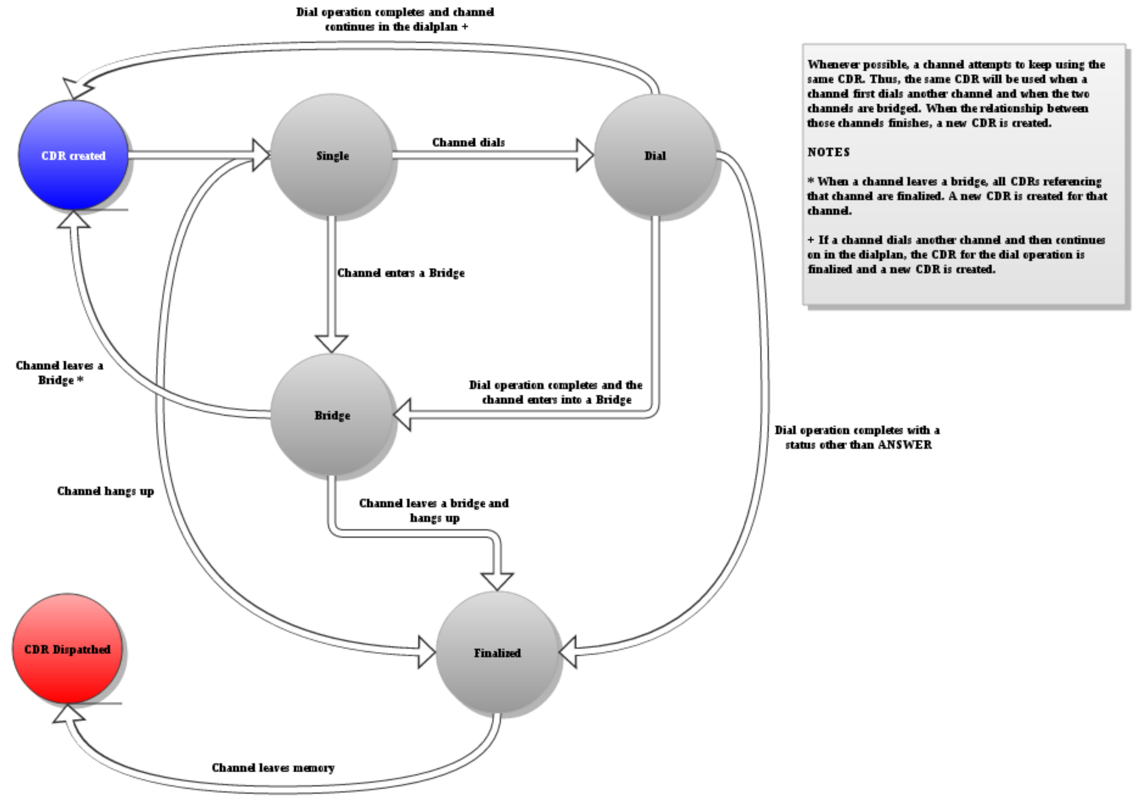
Semantics and Syntax

Basic CDR Lifetime

Fundamentally, a CDR represents a path of communication between a channel and Asterisk or between two channels communicating through Asterisk. Those relationships can be broken down into the following states:

- Single - a channel is executing dialplan in Asterisk or otherwise has no relationship with any other channel. This is the default state of a CDR.
- Dial - a channel is involved in a dial operation - either as a caller or as the callee. If they are the caller, they are automatically the Party A in the CDR. If they are the callee, they are automatically the Party B if there is a caller. If there is no caller - as is the case in channel origination - they are the Party A and there is no Party B.
- Bridge - two channels share a bridge in which they can potentially talk to each other.
- Finalized - the channel and Asterisk can no longer communicate or the relationship between the two channels has been broken. At this point, the last state of the channels involved is locked in the CDR.
- Dispatched - the CDR has been written to the backend storages.

When possible, Asterisk attempts to reuse a CDR. Thus a single CDR can transition through all of the above states. However, once a relationship is established between a Party A and a Party B, the CDR **cannot** be reused. The CDR will be finalized when the relationship between channels is broken and a new CDR created for any channel that is not hung up.



When a channel is created, a CDR is created for that channel. The channel is automatically the Party A of that CDR. The CDR is considered to be in the **Single** state while the channel executes dialplan or is waiting to be associated with another channel.

If a channel dials another channel, the CDR for that channel is transitioned to the **Dial** state. If the dial operation completes successfully and the channels are bridged together, the CDR transitions to the **Bridged** state. If the dial operation fails, the CDR transitions to the **Finalized** state. If the caller continues on in the dialplan, a new CDR is generated for them.

While the channels are bridged together, the CDR remains in the **Bridged** state. Operations that restrict media flow, such as call hold, are not reflected in CDRs. When the bridge is broken - either because one of the parties hangs up or a party is transferred - the CDR transitions to the **Finalized** state and any non-hungup channels have a new CDR created for them.

When a channel hangs up, all CDRs associated with it are implicitly finalized. When all CDRs for a channel are finalized, the CDRs are **Dispatched** to the backends for storage.

Detailed Semantics

CDR Lifetime

Creation

A CDR record is created in any one of the following situations:

- Whenever a channel is created.
- Whenever a channel leaves a bridge and is not hung up.
- When a CDR is forked from a prior record.
- When a channel enters a multi-party bridge.
- When a channel dials more than one channel.

When it is created, a CDR inherits the `uniqueid` and `linkedid` from its Party A channel, and a new `sequence` number is generated. When created as a result of a dial operation, the channel acting as the caller is always the Party A.

Dialing Parties

When a channel is known to dial other channels, a CDR is created for each dial attempt. The dial status is recorded for each dial attempt as a CDR Disposition. Note that not all dial attempts may be dispatched depending on the CDR configuration. The caller is always the Party A in the created CDRs.

Bridging

If a channel is bridged with another channel, the following procedure is performed:

- The CDR is marked as having entered a bridge. If there is no other channel in the bridge, the CDR waits for another channel to join.
- For each pairing of channels, the channels are compared to determine which is the Party A channel and which is the Party B channel. (See Choosing the Party A channel).
  - If the channel entering the bridge is the Party A, the CDR has a Party B, and the channel it is bridged with is the Party B, the CDR continues.
  - If the channel entering the bridge is the Party A, the CDR has a Party B, and the channel is not already the Party B, the current CDR is finalized and a new CDR is created for the relationship between the two parties. Note that the original CDR will be re-activated if the existing Party B enters the bridge.
  - If the channel entering the bridge is the Party A, the CDR has no Party B, then the channel it is bridged with becomes the Party B.
  - If the channel entering the bridge is the Party B, the other channel has a CDR with no Party B, this channel becomes the Party B and the existing CDR is finalized.
  - If the channel entering the bridge is the Party B, the other channel has a CDR with a Party B, and this channel is that CDR's Party B, then the existing CDR is finalized and the other channel's CDR activated.
  - If the channel entering the bridge is the Party B, the other channel has a CDR with a Party B, and this channel is not that CDR's Party B, then the existing CDR is finalized and a new CDR is created for that other channel with this channel as the Party B.
- If a third party joins the bridge with Party A and Party B, the process Choosing the Party A channel is repeated for each pairing of channels. Thus, in a three-way call there will be three CDR records; in a four-way call there will be six records, etc.

> This feels complex, but there's really two rules going on here:
>
> 1. Keep using the existing CDR for a channel as long as possible
> 2. Make CDRs for all pairings of channels in a bridge

Finalization

A CDR is finalized in one of the following scenarios:

- If in a dial, the dial operation completes with a status other than ANSWER
- If in a bridge, either party A or party B leaves the bridge
- Either channel in a CDR hangs up
- The CDR is forked and the forking operation instructs that the CDR should be finalized

When a CDR is finalized, no further modifications can be made to the CDR by the user or Asterisk.

If a Party A channel in a CDR is not hung up but the CDR is finalized - such as when the channel leaves a bridge of its Party B hangs up - a new CDR is made for that channel and the process in CDR Creation is begun again. Note that if the Party B in a CDR continues on in the the dialplan and/or is bridged with a new party, it may become Party A for a new CDR.

If at any point the Party A channel for a CDR is hung up, all CDR records for that Party A are dispatched.

Dispatch

When a CDR is dispatched, all CDRs associated with the channel are committed to permanent storage. The CDRs at this point are removed from memory.

Choosing the Party A channel

Asterisk does not have the concept of "internal" versus "external" devices. As such, what constitutes the Party A channel is highly dependent on a particular system configuration which is outside the control of the CDR system. As such, choosing a Party A uses the following rules:

1. If the channel was dialed (but not originated), the channel is always Party B.
2. If one of the two channels has the `party_a` flag set, then that channel is chosen as the Party A.
3. If neither or both channels have the `party_A` flag, the channel with the oldest creation time is chosen as the Party A.

The `party_A` flag may be set using the CDR function.

LinkedID Propagation

When two channels are bridged, the `linkedid` property for the channels is updated. The channel with the oldest `linkedid` "wins", and the other channel's `linkedid` is replaced. This creates an association between the channels that lasts even if the bridge is broken at a latter time.

Note that dialed channels automatically receive the `linkedid` of the calling channel.

Fields

Standard Fields

Field	Type	Description	Access
accountcode	String (20)	An account code associated with the Party A channel	r/w

src	String (80)	The Caller ID Number	r
dst	String (80)	The destination extension	r
dcontext	String (80)	The destination context	r
clid	String (80)	The Caller ID with text	r
channel	String (80)	The name of the Party A channel	r
dstchannel	String (80)	The name of the Party B channel	r
lastapp	String (80)	The last application the Party A channel executed	r
lastdata	String (80)	The application data for the last application the Party A channel executed	r
start	Date/time	The time the CDR was created	r
answer	Date/time	The time when Party A was answered, or when the bridge between Party A and Party B was created	r
end	Date/time	The time when the CDR was finished. This occurs when either party hangs up, or when the bridge between the parties is broken	r
duration	Integer	The time in seconds from start until end	r

billsec	Integer	The time in seconds from answer until end	r
disposition	Enum	The final known disposition of the CDR record. See CDR dispositions for possible values	r
amaflags	Enum	A flag specified on the Party A channel. See AMA records for possible values	r/w
userfield	String (255)	A user defined field set on the channels. If set on both the Party A and Party B channel, the userfields of both are concatenated and separated by a `;`	r/w
uniqueid	String (32)	A unique identifier for the Party A channel	r
linkedid	String (32)	A unique identifier that unites multiple CDR records. See linkedid propagation for more details	r
peeraccount	String (80)	The account code of the Party B channel	r/w
sequence	Integer	A numeric value that, combined with uniqueid and linkedid, can be used to uniquely identify a single CDR record	r

Any of the values may be accessed using the CDR function. Any value that is read/write may be modified using this same function. CDR field values cannot be modified once the CDR is finalized.

Dispositions

Dispositions represent the final state of the call from the perspective of Party A.

Value	Description	Hangup Cause Mapping	Dial Status Mapping
NO ANSWER	The channel was never answered. This is the default disposition for an unanswered channel.	Any not explicitly listed	• CANCEL • NOANSWER
CONGESTION	The channel dialed something that was congested.	• `AST_CAUSE_CONGESTION`	CONGESTION
FAILED	The channel attempted to dial but the call failed. ⓘ The `congestion` setting in `cdr.conf` can result in the `AST_CAUSE_CONGESTION` hangup cause or the CONGESTION dial status to map to this disposition.	• `AST_CAUSE_CONGESTION` • `AST_CAUSE_NO_ROUTE_DESTINATION` • `AST_CAUSE_UNREGISTERED`	• CONGESTION • FAILED
BUSY	The channel attempted to dial but the remote party was busy.	• `AST_CAUSE_BUSY`	• BUSY

ANSWERED	The channel was answered. When the channel is answered, the hangup cause no longer changes the disposition.	Any not explicitly listed	• ANSWER

AMA Flags

AMA Flags are set on a channel and are conveyed in the CDR. They inform billing systems how to treat the particular CDR. Asterisk provides no additional semantics regarding these flags - they are present simply to help external systems classify CDRs.

- OMIT
- BILLING
- DOCUMENTATION

User Defined Fields

Any CDR record may have user defined fields associated with it. Fields can be added to either the Party A or Party B channel. Note that not all CDR backends support user defined fields; in those cases the user field is simply dropped when the CDR is dispatched to the backend.

If a Party A channel and a Party B channel both contain a field with the same key, only the Party A channel's field will be written to the CDR.

User defined CDR fields are created using the CDR function, and read using the same function.

Scenarios

> The following scenarios show examples of CDRs created in common use cases. If a particular scenario is not shown below, the CDRs created during the scenario should still match the behavior described previously. Some applications, however, may have undefined behavior as their use is not common or the mechanism by which they manipulate channels does not allow for the capturing of the channel state.
>
> Undefined behavior means that the behavior of CDRs in those cases is unsupported and will not be addressed as a bug.

Unanswered "Inbound" Call

> Unanswered calls may not always be logged to CDR backends if the configuration has explicitly disabled unanswered calls.

Alice calls into Asterisk at extension 500 using a SIP phone and, during dialplan execution of a NoOp(), hangs up.

clid	src	dst	dcontext	channel	dstchannel	lastapp	lastdata	start	answer	end	duration	billsec	disposition	amaflags	accountcode	peeraccount	uniqueid	userfield	sequence	linkedid
"Alice <100>"	100	500	default	SIP/alice-00000000		NoOp		2013-03-04 13:11:18		2013-03-04 13:11:20	2	0	NO ANSWER	DOCUMENTATION	100		Asterisk-01-1362424276.2		34	Asterisk-01-1362424276.2

Unanswered "Outbound" Call

Asterisk creates a call file to dial Alice and playback tt-monkeys to her. Alice, anticipating the screeching of howler monkeys, never picks up the phone.

clid	src	dst	dcontext	channel	dstchannel	lastapp	lastdata	start	answer	end	duration	billsec	disposition	amaflags	accountcode	peeraccount	uniqueid	userfield	sequence	linkedid

		s	default	SIP/alice-00000000		AppDial	SIP/Alice	2013-03-04 13:11:18		2013-03-04 13:11:20	2	0	NOANSWER	DOCUMENTATION			Asterisk-01-1362424276.2		4	Asterisk-01-1362424276.2

Single Party

Alice calls into Asterisk's VoiceMailMain application. This implicitly Answers the channel. She checks her voicemail for awhile, then hangs up.

clid	src	dst	dcontext	channel	dstchannel	lastapp	lastdata	start	answer	end	duration	billsec	disposition	amaflags	accountcode	peeraccount	uniqueid	userfield	sequence	linkedid

"Alice <100>"	100	8500	default	SIP/alice-00000000		Hangup		2013-03-04 13:11:18	2013-03-04 13:11:20	2013-03-04 13:12:18	60	58	ANSWERED	DOCUMENTATION	100		Asterisk-01-1362424276.2		112	Asterisk-01-1362424276.2

Basic Two Party Calls

Two party calls can be initiated in a variety of ways. Several of the more common ways are illustrated here.

Basic Call

Alice calls into Asterisk, which dials Bob. Bob Answers, and a bridge is formed between Alice and Bob. Alice and Bob talk for awhile, then Bob hangs up. This breaks the bridge between Alice and Bob, and Alice is hung up on as well.

clid	src	dst	dcontext	channel	dstchannel	lastapp	lastdata	start	answer	end	duration	billsec	disposition	amaflags	accountcode	peeraccount	uniqueid	userfield	sequence	linkedid

"Alice <100>"	100	200	default	SIP/alice-00000000	SIP/bob-00000001	Dial	SIP/bob,,Tt	2013-03-04 13:11:18	2013-03-04 13:11:26	2013-03-04 13:13:18	120	112	ANSWERED	DOCUMENTATION			asterisk-01-1362424276.2		12	asterisk-01-1362424276.2

Unanswered Dial

Alice calls into Asterisk, which dials Bob. Bob refuses to pick up his phone, and the call eventually times out.

clid	src	dst	dcontext	channel	dstchannel	lastapp	lastdata	start	answer	end	duration	billsec	disposition	amaflags	accountcode	peeraccount	uniqueid	userfield	sequence	linkedid

"Alice <100>"	100	200	default	SIP/alice-00000000	SIP/bob-00000001	Dial	SIP/bob,10,Tt	2013-03-04 13:11:18		2013-03-04 13:11:28	10	0	NOANSWER	DOCUMENTATION			Asterisk-01-1362424276.2		1	Asterisk-01-1362424276.2

Parallel Dial

Alice calls into Asterisk, which dials Bob's SIP desk phone as well as his IAX2 soft phone. Both ring for awhile, and Bob eventually presses the Answer button on his IAX2 soft phone.

clid	src	dst	dcontext	channel	dstchannel	lastapp	lastdata	start	answer	end	duration	billsec	disposition	amaflags	accountcode	peeraccount	uniqueid	userfield	sequence	linkedid

"Alice <1000>"	100	200	default	SIP/alice-00000000	SIP/bob-00000001	Dial	SIP/bob&IAX2/bob,,Tt	2013-03-04 13:11:18		2013-03-04 13:11:28	10	0	NOANSWER	DOCUMENTATION			Asterisk-01-1362424276.2		12	Asterisk-01-1362424276.2
"Alice <1000>"	100	200	default	SIP/alice-00000000	IAX2/bob-00000000	Dial	SIP/bob&IAX2/bob,,Tt	2013-03-04 13:11:18	2013-03-04 13:11:28	2013-03-04 13:12:28	70	60	ANSWERED	DOCUMENTATION			Asterisk-01-1362424276.2		13	Asterisk-01-1362424276.2

Call Forward

Alice calls into Asterisk, which dials Bob's SIP desk phone. Bob is on vacation, and the SIP phone returns a "302" and redirects Asterisk to dial his SIP mobile.

Transfers

Transfers create multiple CDRs. In general, a CDR is created for each path of communication between two endpoints. Note that Asterisk does **not** attempt to compute the total duration or billing time of any of the various channels involved - it is up to the businesses consuming CDRs to know whether or not the amount of time they want to bill a party includes the transfer, the time spent dialing another party, consultation time, etc.

Blind Transfer

Alice calls into Asterisk, which dials Bob's SIP phone. Bob answers, and Alice and Bob talk for awhile. Eventually, Bob decides to send Alice off to Charlie, and he blind transfers Alice to Charlie's extension. Asterisk dials Charlie's SIP phone, and Charlie answers. Alice and Charlie talk for awhile until Alice decides to hang up.

clid	src	dst	dcontext	channel	dstchannel	lastapp	lastdata	start	answer	end	duration	billsec	disposition	amaflags	accountcode	peeraccount	uniqueid	userfield	sequence	linkedid
"Alice <1000>"	1000	200	default	SIP/alice-00000000	SIP/bob-00000001	Dial	SIP/bob,,Tt	2013-03-04 13:11:18	2013-03-04 13:11:28	2013-03-04 13:11:48	30	20	ANSWERED	DOCUMENTATION			Asterisk-01-1362424276.2		101	Asterisk-01-1362424276.2

"Alice <100>"	100	300	default	SIP/alice-00000000	SIP/charlie-00000002	Dial	SIP/charlie,,Tt	2013-03-04 13:11:48	2013-03-04 13:11:53	2013-03-04 13:12:53	65	60	ANSWERED	DOCUMENTATION			Asterisk-01-1362424276.2		102	Asterisk-01-1362424276.2

Attended Transfer to Channel

Alice calls into Asterisk, which dials Bob's SIP phone. Bob answers, and Alice and Bob talk for awhile. Eventually, Bob decides to send Alice off to Charlie, and he initiates an attended transfer. Alice is put on hold, and Bob dials Charlie's extension. Asterisk dials Charlie's SIP phone, and Charlie answers. Bob and Charlie talk for a bit, and Charlie agrees to talk to Alice. Bob completes the attended transfer, Alice is taken off hold, and Alice and Charlie are bridged. Alice talks to Charlie for awhile, then hangs up.

clid	src	dst	dcontext	channel	dstchannel	lastapp	lastdata	start	answer	end	duration	billsec	disposition	amaflags	accountcode	peeraccount	uniqueid	userfield	sequence	linkedid

"Alice <1000>"	100	200	default	SIP/alice-00000000	SIP/bob-00000001	Dial	SIP/bob,,Tt	2013-03-04 13:11:18	2013-03-04 13:11:28	2013-03-04 13:12:18	60	50	ANSWERED	DOCUMENTATION			Asterisk-01-1362424276.2		101	Asterisk-01-1362424276.2
"Bob <200>"	200	300	default	SIP/bob-00000001	SIP/charlie-00000002	Dial	SIP/charlie,,Tt	2013-03-04 13:11:48	2013-03-04 13:11:53	2013-03-04 13:12:18	30	26	ANSWERED	DOCUMENTATION			Asterisk-01-1362424280.1		102	Asterisk-01-1362424276.2

"Alice <100>"	100	300	default	SIP/alice-00000000	SIP/charlie-00000002	Dial	SIP/bob,,Tt	2013-03-04 13:12:18	2013-03-04 13:12:18	2013-03-04 13:12:53	45	45	ANSWERED	DOCUMENTATION			Asterisk-01-1362424276.2		103	Asterisk-01-1362424276.2

⚠ In the example above, note the following:

- Hold time is not reflected in CDRs.
- When Bob dials Charlie, he becomes the Party A channel. However, the `linkedid` from Alice 'wins', and so the CDR reflects the `linkedid` from Alice's CDR.
- Alice and Charlie are bridged automatically by the attended transfer, so their start and answer times are identical.
- The billsec/duration of Alice and Charlie are reflective of their portion of the call, and do not include the times from Alice and Bob.

Attended Transfer to Application

Alice calls into Asterisk, which dials Bob's SIP phone. Bob answers, and Alice and Bob talk for awhile. Eventually, Bob decides to send Alice off into Charlie's voicemail mailbox, and he initiates an attended transfer. Alice is put on hold, and Bob dials an extension that calls into VoiceMail. Bob enters in the Charlie's voicemail mailbox number, then completes the attended transfer to put Alice into the voicemail mailbox. Alice records some voicemail, then hangs up.

clid	src	dst	dcontext	channel	dstchannel	lastapp	lastdata	start	answer	end	duration	billsec	disposition	amaflags	accountcode	peeraccount	uniqueid	userfield	sequence	linkedid

"Alice <100>"	100	200	default	SIP/alice-00000000	SIP/bob-00000001	Dial	SIP/bob,,Tt	2013-03-04 13:11:18	2013-03-04 13:11:28	2013-03-04 13:12:18	60	50	ANSWERED	DOCUMENTATION			Asterisk-01-1362424276.2		101	Asterisk-01-1362424276.2
"Bob <200>"	200	8500	default	SIP/bob-00000001		VoiceMail	300	2013-03-04 13:11:48	2013-03-04 13:11:49	2013-03-04 13:12:18	30	29	ANSWERED	DOCUMENTATION			Asterisk-01-1362424280.1		102	Asterisk-01-1362424276.2

"Alice <100>"	100	8500	default	SIP/alice-00000000		VoiceMail	300	2013-03-04 13:12:18	2013-03-04 13:12:18	2013-03-04 13:13:18	60	60	ANSWERED	DOCUMENTATION			Asterisk-01-1362424276.2		103	Asterisk-01-1362424276.2

Blonde Transfer

Alice calls into Asterisk, which dials Bob's SIP phone. Bob answers, and Alice and Bob talk for awhile. Eventually, Bob decides to send Alice off to Charlie, and he initiates an attended transfer. Alice is put on hold, and Bob dials Charlie's extension. Asterisk dials Charlie's SIP phone, but before Charlie answers Bob hangs up. Asterisk recognizes that this is a blonde transfer, takes Alice off hold, and ties Charlie's ringing phone to Alice. Charlie answers, Alice talks to Charlie for awhile, then hangs up.

clid	src	dst	dcontext	channel	dstchannel	lastapp	lastdata	start	answer	end	duration	billsec	disposition	amaflags	accountcode	peeraccount	uniqueid	userfield	sequence	linkedid

"Alice<100>"	100	200	default	SIP/alice-00000000	SIP/bob-00000001	Dial	SIP/bob,,Tt	2013-03-04 13:11:18	2013-03-04 13:11:28	2013-03-04 13:12:18	60	50	ANSWERED	DOCUMENTATION			Asterisk-01-1362424276.2		101	Asterisk-01-1362424276.2
"Bob<200>"	200	300	default	SIP/bob-00000001	SIP/charlie-00000002	Dial	SIP/charlie,,Tt	2013-03-04 13:11:48		2013-03-04 13:11:50	2	0	NO ANSWER	DOCUMENTATION			Asterisk-01-1362424280.1		102	Asterisk-01-1362424276.2

"Alice <100>"	100	300	default	SIP/alice-00000000	SIP/charlie-00000002	Dial	SIP/charlie,,Tt	2013-03-04 13:11:50	2013-03-04 13:11:55	2013-03-04 13:12:30	40	35	ANSWERED	DOCUMENTATION			Asterisk-01-1362424276.2		103	Asterisk-01-1362424276.2

Three Way Call

Alice calls into Asterisk, which dials Bob's SIP phone. Bob answers, and Alice and Bob talk for awhile. Eventually, Bob decides to bring Charlie into the mix. He puts Alice on hold and dials Charlie's extension. Asterisk dials Charlie's SIP phone, and Charlie answers. Bob and Charlie talk for awhile, and Bob then finishes the three-way call by finalizing the attempt. Alice is taken off hold, and Alice, Bob, and Charlie can all talk. Eventually, Bob hangs up, and all parties are ejected and hung up on.

clid	src	dst	dcontext	channel	dstchannel	lastapp	lastdata	start	answer	end	duration	billsec	disposition	amaflags	accountcode	peeraccount	uniqueid	userfield	sequence	linkedid

"Alice <100>"	100	200	default	SIP/alice-00000000	SIP/bob-00000001	Dial	SIP/bob,,Tt	2013-03-04 13:11:18	2013-03-04 13:11:28	2013-03-04 13:13:18	120	110	ANSWERED	DOCUMENTATION			Asterisk-01-1362424276.2		101	Asterisk-01-1362424276.2
"Bob <200>"	200	300	default	SIP/bob-00000001	SIP/charlie-00000002	Dial	SIP/charlie,,Tt	2013-03-04 13:11:48	2013-03-04 13:11:53	2013-03-04 13:12:08	20	15	ANSWERED	DOCUMENTATION			Asterisk-01-1362424280.1		102	Asterisk-01-1362424276.2

"Alice <100>"	100	300	default	SIP/alice-00000000	SIP/charlie-00000002	Dial	SIP/bob,,Tt	2013-03-04 13:12:08	2013-03-04 13:12:08	2013-03-04 13:13:18	70	70	ANSWERED	DOCUMENTATION			Asterisk-01-1362424276.2		103	Asterisk-01-1362424276.2
"Bob <200>"	200	300	default	SIP/bob-00000001	SIP/charlie-00000002	Dial	SIP/charlie,,Tt	2013-03-04 13:12:08	2013-03-04 13:12:08	2013-03-04 13:13:18	70	70	ANSWERED	DOCUMENTATION			Asterisk-01-1362424280.1		104	Asterisk-01-1362424276.2

> ⚠ In the example above, note the following:
>
> - The consultation between Bob and Charlie is treated as a separate conversation from the conversation between all three parties. Thus, there are two CDRs between Bob and Charlie.
> - Because the path of communication never was broken between Alice and Bob (despite Alice being put on hold), there is only one CDR for Alice to Bob.

SIP Attended Transfer

Alice calls into Asterisk, which dials Bob's SIP phone. Bob answers, and Alice and Bob talk for awhile. Eventually, Alice decides to transfer Bob to Charlie, and performs an attended transfer using her SIP phone. Bob is put on hold. Alice and Charlie talk for awhile, and then Alice finishes the attended transfer. Bob is taken off hold, and bridged with Charlie. Alice is hung up.

clid	src	dst	dcontext	channel	dstchannel	lastapp	lastdata	start	answer	end	duration	billsec	disposition	amaflags	accountcode	peeraccount	uniqueid	userfield	sequence	linkedid
"Alice <1000>"	1000	2000	default	SIP/alice-00000000	SIP/bob-00000001	Dial	SIP/bob	2013-03-04 13:11:18	2013-03-04 13:11:28	2013-03-04 13:12:18	60	50	ANSWERED	DOCUMENTATION			Asterisk-01-1362424276.2		101	Asterisk-01-1362424276.2
"Bob <2000>"	2000	3000	default	SIP/bob-00000002	SIP/charlie-00000003	Dial	SIP/charlie	2013-03-04 13:11:48	2013-03-04 13:11:53	2013-03-04 13:12:08	20	15	ANSWERED	DOCUMENTATION			Asterisk-01-1362428280.1		102	Asterisk-01-1362424276.2

"Alice<100>"	100	200	default	SIP/alice-00000000	SIP/charlie-00000003	Dial	SIP/bob	2013-03-04 13:12:18	2013-03-04 13:12:18	2013-03-04 13:12:28	10	10	ANSWERED	DOCUMENTATION			Asterisk-01-1362424276.2		103	Asterisk-01-1362424276.2

> ⚠ The important point to note here is that a SIP attended transfer uses two channels to communicate with Bob - `SIP/bob-00000001` and `SIP/bob-00000002`. The CDR records are associated by virtue of the `linkedid` field.

Local Channels

Local channels are a special Asterisk construct that create a path of communication between two bridges or a bridge and an application. A Local channel always consists of two channels with the same name prefix - the first of the Local channel pair is delineated with a `;1`; the second is delineated with a `;2`. Local channels have two different modes in Asterisk:

- They can exist permanently. In that case, the Local channel pair appears as two separate channels. From the perspective of CDRs, they are treated as such with an implicit bridge between the channels. CDRs with a permanent Local channel pair will share the same `linkedid`.
- They can optimize. A Local channel optimization occurs when a pair of Local channels exist between two bridges and the Local channel has been configured to optimize. If that situation occurs, the Local channel will cause the two bridges to merge into a single merge, and the Local channel will disappear.

Each situation and how it appears in CDRs is explored further below.

Non-optimizing Local Channels

Local channel to an application

An external application Originates a Local channel. The first half of the Local channel Dials Alice over a SIP channel. The second half of the Local channel is placed into her VoiceMail account. Alice listens to her VoiceMail through the Local channel, then hangs up.

clid	src	dst	dcontext	channel	dstchannel	lastapp	lastdata	start	answer	end	duration	billsec	disposition	amaflags	accountcode	peeraccount	uniqueid	userfield	sequence	linkedid
""	dial_alice	100	default	Local/dial_alice@default-00000001;1	SIP/alice-00000002	Dial	SIP/alice,,tT	2013-03-04 13:11:18	2013-03-04 13:11:28	2013-03-04 13:12:18	60	50	ANSWERED	DOCUMENTATION			Asterisk-01-1362424290.1		101	Asterisk-01-1362424290.1

""	voicemail		default	Local/dial_alice@default-00000001;2		VoiceMailMain	100	2013-03-04 13:11:28	2013-03-04 13:11:28	2013-03-04 13:12:18	50	50	ANSWERED	DOCUMENTATION			Asterisk-01-1362424290.1		102	Asterisk-01-1362424290.1

Local channel between bridges

An external application Originates a Local channel. The first half of the Local channel Dials Alice over a SIP channel; Alice answers. This triggers the second half of the Local channel, which Dials Bob. Bob Answers, and Alice and Bob talk. Alice hangs up, the Local channels are hung up, and Bob is hung up on.

clid	src	dst	dcontext	channel	dstchannel	lastapp	lastdata	start	answer	end	duration	billsec	disposition	amaflags	accountcode	peeraccount	uniqueid	userfield	sequence	linkedid

""	dial_alice	100	default	Local/dial_alice@default-00000001;1	SIP/alice-00000002	Dial	SIP/alice,,tT	2013-03-04 13:11:18	2013-03-04 13:11:28	2013-03-04 13:12:18	60	50	ANSWERED	DOCUMENTATION			Asterisk-01-1362424290.1		101	Asterisk-01-1362424290.1

""	dial_bob	200	default	Local/dial_alice@default-00000001;2	SIP/bob-00000003	Dial	SIP/bob,,tT	2013-03-04 13:11:28	2013-03-04 13:11:38	2013-03-04 13:12:18	50	40	ANSWERED	DOCUMENTATION			Asterisk-01-1362424290.1		102	Asterisk-01-1362424290.1

Local Channel Optimization

When a Local channel optimization occurs, the CDR records associated with the Local channel are discarded. New CDR records are generated for the channels in the merged bridge, per the rules in CDR Consolidation.

> CDR properties set on optimized Local channels are **not** propagated to other channels. Setting CDR information on optimizing Local channels will cause that information to be lost.
>
> In prior versions of Asterisk it was sometimes necessary to set CDR information on Local channels - with the addition of Pre-Dial handlers, it is always possible to set CDR information on the appropriate channel at the time of creation.

Call Hold

Call Hold is a state of the media between two or more channels, and not a change in the actual bridging of those channels. As such, Call Hold is **not** reflected in CDRs. Channels may be put on hold, taken off hold, put on hold again, forgotten about, found again, taken off hold, etc. without affecting CDRs at all.

Call Park

Call Park is different from Call Hold. Whereas Call Hold is a change of media state, Call Park implies that the channel has been moved into a state where it can be retrieved by any other channel. As such, calls in Park receive their own CDR.

Alice calls into Asterisk and Bob answers. Alice says she wants to talk to Charlie, but Bob isn't sure Charlie wants to talk to Alice so he blind transfers her into Park. Alice sits, waiting in her parking slot, listening to serenading robots while Bob asks if Charlie wants to talk to Alice. Charlie says sure, so he picks Alice up out of Park and they talk for awhile before Alice hangs up.

clid	src	dst	dcontext	channel	dstchannel	lastapp	lastdata	start	answer	end	duration	billsec	disposition	amaflags	accountcode	peeraccount	uniqueid	userfield	sequence	linkedid
"Alice <1000>"	100	200	default	SIP/alice-00000000	SIP/bob-00000001	Dial	SIP/bob,,Tt	2013-03-04 13:11:18	2013-03-04 13:11:28	2013-03-04 13:11:38	20	10	ANSWERED	DOCUMENTATION			Asterisk-01-1362424276.2		101	Asterisk-01-1362424276.2

"Alice <100>"	100	700	default	SIP/alice-00000000		Park	600,default,200,1	2013-03-04 13:11:38	2013-03-04 13:11:38	2013-03-04 13:12:38	60	60	ANSWERED	DOCUMENTATION			asterisk-01-1362424276.2		102	asterisk-01-1362424276.2
"Alice <100>"	100	300	default	SIP/alice-00000000	SIP/charlie-00000002	Park	600,default,200,1	2013-03-04 13:12:38	2013-03-04 13:12:38	2013-03-04 13:13:38	60	60	ANSWERED	DOCUMENTATION			asterisk-01-1362424276.2		103	asterisk-01-1362424276.2

Call Queues

Call Queue - Example 1

Alice calls into Asterisk and enters Queue without being Answered. She waits in the queue for a period of time. At some point in time, she enters into the head of the queue and the queue performs a ring-all on the members of the queue. There are two queue members - Bob and Charlie. Bob is out to lunch, so his queue member is paused and it returns a Busy indication. Charlie, on the other hand, is not busy and answers. Alice and Charlie talk for awhile, and eventually Alice hangs up.

clid	src	dst	dcontext	channel	dstchannel	lastapp	lastdata	start	answer	end	duration	billsec	disposition	amaflags	accountcode	peeraccount	uniqueid	userfield	sequence	linkedid
"Alice <1000>"	1000	800	default	SIP/alice-00000000	SIP/bob-00000001	Queue	complaints	2013-03-04 13:11:18		2013-03-04 13:14:18	180	0	BUSY	DOCUMENTATION			Asterisk-01-1362424276.2		101	Asterisk-01-1362424276.2
"Alice <1000>"	1000	800	default	SIP/alice-00000000	SIP/charlie-00000002	Queue	complaints	2013-03-04 13:11:18	2013-03-04 13:12:18	2013-03-04 13:14:18	180	120	ANSWERED	DOCUMENTATION			Asterisk-01-1362424276.2		102	Asterisk-01-1362424276.2

Call Queue - Example 2

Alice calls into Asterisk and enters Queue without being Answered. She waits in the queue for a period of time. At some point in time, she enters into the head of the queue and the queue performs a round-robin strategy ring on the members of the queue. Bob is rung first, but being lazy, he ignores his phone. After some time of ringing, it times out and goes on to Charlie. When Charlie's SIP phone rings, he immediately answers. Alice and Charlie talk for some period of time, and eventually Alice hangs up.

clid	src	dst	dcontext	channel	dstchannel	lastapp	lastdata	start	answer	end	duration	billsec	disposition	amaflags	accountcode	peeraccount	uniqueid	userfield	sequence	linkedid
"Alice <1000>"	100	800	default	SIP/alice-00000000	SIP/bob-00000001	Queue	complaints	2013-03-04 13:11:18		2013-03-04 13:14:18	180	0	NOANSWER	DOCUMENTATION			Asterisk-01-1362424276.2		101	Asterisk-01-1362424276.2

"Alice <100>"	100	800	default	SIP/alice-00000000	SIP/charlie-00000002	Queue	complaints	2013-03-04 13:11:18	2013-03-04 13:12:38	2013-03-04 13:14:18	180	100	ANSWERED	DOCUMENTATION			Asterisk-01-1362424276.2		102	Asterisk-01-1362424276.2

Conference Call

Alice calls into Asterisk and joins a ConfBridge conference. Bob does a bit later as well. Finally, Charlie joins the Conference. After talking for awhile, Bob realizes he's late for lunch and hangs up. Alice and Charlie talk for a bit longer, then finally Alice hangs up. Charlie stays in the conference for another second before he hangs up as well.

clid	src	dst	dcontext	channel	dstchannel	lastapp	lastdata	start	answer	end	duration	billsec	disposition	amaflags	accountcode	peeraccount	uniqueid	userfield	sequence	linkedid

"Alice <100>"	100	1000	default	SIP/alice-00000000	SIP/bob-00000001	ConfBridge	1000,public_bridge,public_user,public_menu	2013-03-04 13:11:18	2013-03-04 13:11:28	2013-03-04 13:14:18	180	170	ANSWERED	DOCUMENTATION			Asterisk-01-1362424276.2		1	Asterisk-01-1362424276.2

"Bob<2000>"	200	1000	default	SIP/bob-00000001	SIP/charlie-00000002	ConfBridge	1000,public_bridge,public_user,public_menu	2013-03-04 13:11:28	2013-03-04 13:12:18	2013-03-04 13:14:18	170	120	ANSWERED	DOCUMENTATION			Asterisk-01-1362424280.1		2	Asterisk-01-1362424276.2

"Alice<100>"	100	1000	default	SIP/alice-00000000	SIP/charlie-00000002	ConfBridge	1000,public_bridge,public_user,public_menu	2013-03-04 13:11:18	2013-03-04 13:12:18	2013-03-04 13:15:18	240	180	ANSWERED	DOCUMENTATION			Asterisk-01-1362424276.2		3	Asterisk-01-1362424276.2

A Complex Example

Alice calls into Asterisk and Dials Bob. Bob answers, and he and Alice talk for awhile. Bob realizes that what Alice really needs is to talk to Sales, so he blind transfers Alice off to the Sales Queue. As Alice is heading off to the Sales Queue, Bob realizes that he should talk with Charlie about Alice, so he Dials Charlies and he and Charlie talk for awhile. Alice enters into the Sales Queue, where she waits for a bit while agents David and Frank are dialed using Local channels to SIP devices. Alice is eventually Answered by David, a sales agent. David and Alice talk for a bit, but David isn't able to sell her on their new fantastic product, so he puts Alice on hold for a bit and calls Ellen from engineering. Ellen agrees to be on the call, and Alice, David, and Ellen are put into a three-way call. Around this time, Charlie decides that he should talk to Alice as well. He transfers himself to the Sales bridge, hanging up on Bob in the process. This turns the Sales bridge into a four-way call. The four parties talk for awhile, and eventually Alice is sold on the new whiz-bang product, so she hangs up. Ellen realizes she isn't need any more either, and hangs up as well. Charlie and David talk about the weather for awhile, and then Charlie hangs up, hanging up David as well.

- Alice calls into Asterisk and Dials Bob. Bob and Alice talk for awhile.

clid	src	dst	dcontext	channel	dstchannel	lastapp	lastdata	start	answer	end	duration	billsec	disposition	amaflags	accountcode	peeraccount	uniqueid	userfield	sequence	linkedid
"Alice <1000>"	1000	2000	default	SIP/alice-00000000	SIP/bob-00000001	Dial	SIP/bob,,Tt	2013-03-04 13:00:00	2013-03-04 13:00:05	2013-03-04 13:01:00	60	55	ANSWERED	DOCUMENTATION			Asterisk-01-1362424276.2		1	Asterisk-01-1362424276.2

- Bob realizes Alice wants to talk to Sales, so he blind transfers her off to the Sales Queue. Alice enters into the Sales Queue, where she waits for a bit while agents David and Frank are dialed using Local channels to SIP devices. Alice is eventually Answered by David, a sales agent.

clid	src	dst	dcontext	channel	dstchannel	lastapp	lastdata	start	answer	end	duration	billsec	disposition	amaflags	accountcode	peeraccount	uniqueid	userfield	sequence	linkedid
"Alice <1000>"	100	700	default	SIP/alice-00000000	Local/member1@default-00000001;1	Queue	sales	2013-03-04 13:01:00		2013-03-04 13:02:00	60	0	NO ANSWER	DOCUMENTATION			Asterisk-01-1362424276.2		2	Asterisk-01-1362424276.2

	700	member1	default	Local/member1@default-00000001;2	SIP/frank-00000002	Dial	SIP/frank	2013-03-04 13:01:50		2013-03-04 13:02:00	10	0	NOANSWER	DOCUMENTATION			Asterisk-01-1362424280.1		3	Asterisk-01-1362424276.2

"Alice <1000>"	100	700	default	SIP/alice-00000000	Local/member2@default-00000002;1	Queue	sales	2013-03-04 13:01:00	2013-03-04 13:02:00	2013-03-04 13:05:00	240	180	ANSWERED	DOCUMENTATION			asterisk-01-1362424276.2		4	asterisk-01-1362424276.2

	700	member2	default	Local/member2@default-00000002;2	SIP/david-00000003	Dial	SIP/david	2013-03-04 13:02:00	2013-03-04 13:02:05	2013-03-04 13:06:00	240	235	ANSWERED	DOCUMENTATION			asterisk-01-1362424280.3		5	asterisk-01-1362424276.2

Note that with non-optimizing Local channels, the duration of the Alice to the Local channel (which in turns passes media to/from David) may not reflect the length of time that the Local channel to David is in the bridge. As we'll see, additional channels joining the bridge will change that CDR's durations.

- Meanwhile, Bob calls Charlie.

clid	src	dst	dcontext	channel	dstchannel	lastapp	lastdata	start	answer	end	duration	billsec	disposition	amaflags	accountcode	peeraccount	uniqueid	userfield	sequence	linkedid
"Bob <200>"	200	300	default	SIP/bob-00000004	SIP/charlie-00000005	Dial	SIP/charlie,,Tt	2013-03-04 13:01:10	2013-03-04 13:01:20	2013-03-04 13:03:20	130	120	ANSWERED	DOCUMENTATION			Asterisk-01-1362424277.1		6	Asterisk-01-1362424277.1

- David and Alice talk for a bit, but David isn't able to sell her on their new fantastic product, so he puts Alice on hold for a bit and calls Ellen from engineering. Ellen agrees to be on the call, and Alice, David, and Ellen are put into a three-way call.

clid	src	dst	dcontext	channel	dstchannel	lastapp	lastdata	start	answer	end	duration	billsec	disposition	amaflags	accountcode	peeraccount	uniqueid	userfield	sequence	linkedid

"David <900>"	900	901	default	SIP/david-00000003	SIP/ellen-00000006	Dial	SIP/ellen,,Tt	2013-03-04 13:02:30	2013-03-04 13:02:40	2013-03-04 13:03:00	30	20	ANSWERED	DOCUMENTATION			Asterisk-01-1362424300.1		7	Asterisk-01-1362424276.2
"Alice <1000>"	100	700	default	SIP/alice-00000000	SIP/ellen-00000006	Queue	sales	2013-03-04 13:03:00	2013-03-04 13:03:00	2013-03-04 13:05:00	120	120	ANSWERED	DOCUMENTATION			Asterisk-01-1362424276.2		8	Asterisk-01-1362424276.2

	700	member2	default	Local/member2@default-00000002;2	SIP/ellen-00000006	Dial	SIP/david	2013-03-04 13:03:00	2013-03-04 13:03:00	2013-03-04 13:05:00	120	120	ANSWERED	DOCUMENTATION			Asterisk-01-1362424280.3		9	Asterisk-01-1362424276.2

> ⚠ During the consultation period, David's SIP channel directly Dials Ellen's SIP device. However, when Ellen joins the bridge with David, it is the Local channel to David that is in the bridge, not David's SIP channel. Thus, the CDR reflects the Local channel to Ellen's SIP channel.

- Around this time, Charlie decides that he should talk to Alice as well. He transfers himself to the Sales bridge, hanging up on Bob in the process. This turns the Sales bridge into a four-way call. The four parties talk for awhile, and eventually Alice is sold on the new whiz-bang product, so she hangs up. Ellen realizes she isn't need any more either, and hangs up as well. Charlie and David talk about the weather for awhile, and then Charlie hangs up, hanging up David as well.

clid	src	dst	dcontext	channel	dstchannel	lastapp	lastdata	start	answer	end	duration	billsec	disposition	amaflags	accountcode	peeraccount	uniqueid	userfield	sequence	linkedid
"Alice <1000>"	100	700	default	SIP/alice-00000000	SIP/charlie-00000005	Queue	sales	2013-03-04 13:03:20	2013-03-04 13:03:20	2013-03-04 13:05:00	100	100	ANSWERED	DOCUMENTATION			Asterisk-01-1362424276.2		10	Asterisk-01-1362424276.2

"Charlie <3000>"	300	701	default	SIP/charlie-00000005	Local/member2@default-00000002;2	Bridge	SIP/alice-00000000	2013-03-04 13:03:20	2013-03-04 13:03:20	2013-03-04 13:06:00	160	160	ANSWERED	DOCUMENTATION			Asterisk-01-1362424278.1		11	Asterisk-01-1362424276.2
"Charlie <3000>"	300	701	default	SIP/charlie-00000005	SIP/ellen-00000006	Bridge	SIP/alice-00000000	2013-03-04 13:03:20	2013-03-4 13:03:20	2013-03-04 13:05:05	105	105	ANSWERED	DOCUMENTATION			Asterisk-01-1362424278.1		12	Asterisk-01-1362424276.2

⚠ Because Charlie's channel is older then either the Local channel to David's SIP channel or

Ellen's SIP channel, Charlie is chosen as Party A for those CDRs. Alice, on the other hand, is older than Charlie, so she is Party A for that CDR. Because Alice is the oldest channel, her `linkedid` is propagated to all CDRs in the bridge. However, the CDR between Charlie and Bob is not affected, as Bob is the Party A in that CDR and the CDR would already have been dispatched by the time Charlie joined this bridge.

Asterisk CDR APIs

The following details high level APIs that Asterisk provides for manipulating CDRs.

⚠ These still describe applications/functions available in Asterisk 11. When documentation has been updated for these applications/functions for Asterisk 12, the links will be updated appropriately.

Applications

NoCDR

When this application is executed on a channel, the channel is no longer considered for CDRs. Any previous CDRs involving the channel will continue to be updated.

ForkCDR

ForkCDR now does significantly less than it used to. The application will finalize the current CDR and create a new CDR for the party A channel. The new CDR record may or may not inherit properties of the previously finalized CDR, based on parameters passed to the application.

ResetCDR

ResetCDR has two purposes:

1. It resets the start time/answer time for the current CDR. If the channel is answered, the start time and answer time will reflect when ResetCDR was called on the channel.
2. Alternatively, it simply enables CDRs on a channel that previously had NoCDR executed on it.

Functions

CDR

Retrieve or modify a field in a CDR record.

Example: Setting the called party as Party A of the CDR

```
[default]

exten => 100,1,NoOp()
 same =>     n,Dial(SIP/bob,,b(default^callee_handler^1))
 same =>     n,Hangup()

exten => callee_handler,1,NoOp()
 same =>                n,Set(CDR(party_A)=true)
 same =>                n,Return()
```

Configuration

The following parameters can be configured for the CDR engine. Additional CDR backends may have their own configuration settings that are outside the scope of this specification.

General Settings

The following settings must appear in the context {{ general }}.

Name	Type	Description
enable	Boolean	Enable/disable the CDR engine
batch	Boolean	Dispatch CDRs in batches.
unanswered	Boolean	Dispatch unanswered CDRs. See Definition of Unanswered for more information.
congestion	Boolean	Treat congestion calls as failed calls
endbeforehexten	Boolean	Finalize CDRs before the h extension or hangup handlers are executed
initiatedseconds	Boolean	Count microseconds for the purposes of the `billsec` field
size	Integer	The number of records to buffer before initiating a batch
time	Integer	The time, in seconds, before initiating a batch

scheduleronly	Boolean	Deprecated. See `usethreadpool` instead.
usethreadpool	Boolean	For any CDRs that are dispatched, use a thread pool thread to perform the dispatching. This prevents the CDR taskprocessor thread from being blocked by any CDR backends.
safeshutdown	Boolean	Block Asterisk shutdown on dispatching of CDRs

Asterisk 12 CEL Specification

Asterisk 12 has not yet been released - this specification is still in draft form.

Introduction

Channel Event Logging (CEL) provides a series of records describing the state of channels in Asterisk to any of several event recording backends. CEL records provide substantially more information than CDRs and thus allow an Asterisk User to construct their own more complex billing system.

As a result of the bridging work done for Asterisk 12, CEL behavior has changed for several events that occur in the system. The most significant changes are:

- AST_CEL_BRIDGE_ENTER and AST_CEL_BRIDGE_EXIT have been introduced to denote participant changes in bridges.
- AST_CEL_BRIDGE_START and AST_CEL_BRIDGE_END have been removed as they no longer applies to the new bridging framework.
- AST_CEL_BRIDGE_UPDATE has been removed as it no longer applies to the new bridging framework.
- AST_CEL_LOCAL_OPTIMIZE has been added to describe local channel optimizations that occur.
- All linkedid accounting and record generation is now handled within the CEL engine.
- The peer field is no longer available due to the changes in the bridging system.

Scope

This CEL specification applies to Asterisk 12. While some portions of this specification are applicable to prior versions of Asterisk, other portions are specific to Asterisk 12 and their counterparts in prior versions are not discussed.

Terminology

Term	Definition
CEL	Channel Event Logging. The focus of this documentation.
CEL record	An individual event record produced by the CEL engine.
CDR	Call Detail Record. An alternative method of extracting billing information from Asterisk. Simpler, but less flexible.
Stasis	The internal message bus in Asterisk that conveys state to the CEL engine.
Primary	The channel around which a CEL record is focused.
AMI	Asterisk Manager Interface
CSV	Comma Separated Values. A format commonly used for tabular data when stored outside of a database.

CEL Overview

A CEL record contains information about a system event including a partial dump of the Primary's state and may contain data relevant to that specific record type such as channel names, bridge unique identifiers, channel variable values, or other miscellaneous information. The CEL engine tracks changes in individual channel state and guarantees ordering of records for a given Primary, but does not guarantee ordering of records in relation to other Primaries. The exception to this record ordering occurs with meta-records which occur adjacent to the events they describe. Applicable event ordering is provided in the descriptions below. CEL output does not describe interaction with MeetMe conferences other than MeetMe as an application.

Record Types

The records produced by the CEL engine can be grouped in to three general categories:

Stand-Alone Records

These records convey a channel event on the channel that does not involve channels or bridges other than the Primary.

Channel Start

An AST_CEL_CHANNEL_START record is generated when a channel is created. This record introduces a new Primary and is the first record available for all Primaries.

Channel End

An AST_CEL_CHAN_END record is generated when a channel is destroyed. This record indicates that a Primary is going away and that there will be no further records for this Primary with the exception of AST_CEL_LINKEDID_END.

Answer

An AST_CEL_ANSWER record is generated when a channel is answered. Depending on the state transitions that occur on a Primary, this record may not be generated.

Hangup

An AST_CEL_HANGUP record is generated when a channel is hung up. This record will occur on every Primary prior to channel destruction.

Application Start

An AST_CEL_APP_START record is generated when a channel enters an application. This record will always be generated before its corresponding AST_CEL_APP_END.

Application End

An AST_CEL_APP_END record is generated when a channel exits an application. This record will be generated after its corresponding AST_CEL_APP_START, but is not guaranteed to be generated on hangup.

User Defined

An AST_CEL_USER_DEFINED record is generated when a channel enters the CELGenUserEvent application. The application sets the user defined name field and additional information in the extra field in the "extra" key.

Linked ID End

An AST_CEL_LINKEDID_END record is generated when the last channel using the given linked ID is destroyed or the last instance of a linked ID is overwritten by a different linked ID. This is the only type of record that may occur after AST_CEL_CHANNEL_END.

Interaction Records

These records convey the Primary's interactions with other channels or bridges.

Bridge Enter

An AST_CEL_BRIDGE_ENTER record is generated when a channel enters a bridge. The entering channel is the Primary for this event. Additional information is conveyed in the extra field under the "bridge_id" key.

Bridge Exit

An AST_CEL_BRIDGE_EXIT record is generated when a channel exits a bridge. The leaving channel is the Primary for this event. Additional information is conveyed in the extra field under the "bridge_id" key.

Forward

An AST_CEL_FORWARD record is generated when a dialing channel is forwarded elsewhere by a dialed channel. The dialing channel is the Primary for this event. Additional information is conveyed in the extra field under the "forward" key.

Park Start

An AST_CEL_PARK_START record is generated when a channel is parked. The parked channel is the Primary for this event. Additional information is conveyed in the extra field under the keys "parker_dial_string" and "parking_lot".

Park End

An AST_CEL_PARK_START record is generated when a channel is unparked. The unparked channel is the Primary for this event. Additional information is conveyed in the extra field under the "reason" key. This record always occurs after its corresponding AST_CEL_PARK_START.

Pickup

An AST_CEL_PICKUP record is generated when a channel is picked up. The picked up channel (also known as the target) is the Primary for this record. The name of the channel that is picking up is conveyed in the extra field under the "pickup_channel" key.

Meta-Records

These records convey additional context relating to surrounding CEL records

Blind Transfer

An AST_CEL_BLINDTRANSFER record is generated when a blind transfer feature is activated on a bridge. The initiating channel is the Primary for this record. Additional information is conveyed in the extra field under the "extension", "context", and "bridge_id" keys.

Attended Transfer

An AST_CEL_ATTENDEDTRANSFER record is generated when an attended transfer is successfully performed.

Bridge-Bridge Attended Transfers

This type of attended transfer occurs when both involved channels are bridged. The initiating channel is the Primary for this record. Additional information is conveyed in the extra field under the "bridge1_id", "channel2_name", and "bridge2_id" keys.

The records associated with this type of transfer will vary depending on the configuration of the bridges involved and the number of channels involved. Possible methods of accomplishing the transfer include (but are not limited to) channel swap, bridge merge, and bridge link via a local channel.

Bridge-App Attended Transfers

This type of attended transfer occurs when one involved channel is bridged while the other is running an application. The bridged channel is the Primary for this record. Additional information is conveyed in the extra field under the "bridge1_id", "channel2_name", and "app" keys.

App-App Attended Transfers

Attended transfers involving only channels that are running applications are not currently possible. This is not possible with internal transfers since there is no bridge involved to handle the feature codes and any externally initiated attended transfer that attempts to bridge two app-bound channels will fail.

Local Channel Optimization

An AST_CEL_LOCAL_OPTIMIZE record is generated when a local channel optimization attempt completes successfully. The semi-one (local channel ending in ';1') channel is the Primary for this event. The name of the semi-two (local channel ending in ';2') channel is conveyed in the extra field under the "local_two" key.

Removed Records

The following record types are no longer available in Asterisk 12:

- AST_CEL_BRIDGE_START
- AST_CEL_BRIDGE_END
- AST_CEL_CONF_START
- AST_CEL_CONF_END
- AST_CEL_CONF_ENTER
- AST_CEL_CONF_EXIT
- AST_CEL_HOOKFLASH
- AST_CEL_3WAY_START
- AST_CEL_3WAY_END
- AST_CEL_BRIDGE_UPDATE
- AST_CEL_TRANSFER

Record Fields

Primary Fields

These fields are populated exclusively from their corresponding fields on the Primary in a consistent manner for every CEL record.

CallerID Name

The name identifying the caller for this channel.

CallerID Number

The number identifying the caller for this channel.

CallerID ANI

Automatic Number Identification caller information provided for this channel.

CallerID RDNIS

Redirecting information for this channel.

CallerID DNID

Dialed Number Identification for this channel.

Extension

The extension in which this channel is currently executing.

Context

The context in which this channel is currently executing.

Channel Name

The name of this channel.

Application Name

The name of the application that this channel is currently executing.

Application Data

The data provided to the application being executed.

Account Code

The account code used for billing.

Peer Account Code

The peer channel's account code.

Unique ID

This channel's instance unique identifier.

Linked ID

This channel's current linked ID which is affected by bridging operations. This identifier starts as the channel's unique ID.

AMA Flags

This channel's Automated Message Accounting flags.

Record Type Specific Fields

These fields vary or may be blank depending on the CEL record type.

User Defined Name

This field is only used for AST_CEL_USER_DEFINED and conveys the user-specified event type.

Extra

This field contains a JSON blob describing additional record-type-specific information.

Logging Backends

CEL provides several methods of logging records to be processed at a later time. CEL only publishes record types to backends that are enabled in the general CEL configuration. Sample configurations are provided with the Asterisk 12 source for all of these backends.

Custom

The Custom CEL output module provides logging capability to a CSV file in a format described in the configuration file. This module is configured in cel_custom.conf.

Manager

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.

ODBC

The ODBC CEL output module provides logging capability to any ODBC-compatible database. This module is configured in cel_odbc.conf.

PGSQL

The PGSQL CEL output module provides logging capability to PostgreSQL databases when it is desirable to avoid the ODBC abstraction layer. This module is configured in cel_pgsql.conf.

RADIUS

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.

SQLite

The SQLite CEL output module provides logging capability to a SQLite3 database in a format described in its configuration file. This module is configured in cel_sqlite3_custom.conf.

TDS

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. This module is configured in cel_tds.conf.

Example Scenarios

For the following scenarios, assume the CEL engine is configured to generate the following record types:

- AST_CEL_CHANNEL_START
- AST_CEL_CHAN_END
- AST_CEL_BRIDGE_ENTER
- AST_CEL_BRIDGE_EXIT

Two-Participant Bridge

The following scenario demonstrates channel creation, channel destruction, bridge start, and bridge end:

Event	Record	Primary	Extra
Channel Alice is created	AST_CEL_CHANNEL_START	Alice	
Channel Bob is created	AST_CEL_CHANNEL_START	Bob	
Bridge Link is created			
Alice enters bridge Link	AST_CEL_BRIDGE_ENTER	Alice	{"bridge_id": "Link"}
Bob enters bridge Link	AST_CEL_BRIDGE_ENTER	Bob	{"bridge_id": "Link"}
Bob exits bridge Link	AST_CEL_BRIDGE_EXIT	Bob	{"bridge_id": "Link"}
Bob is destroyed	AST_CEL_CHAN_END	Bob	
Alice exits bridge Link	AST_CEL_BRIDGE_EXIT	Alice	{"bridge_id": "Link"}
Alice is destroyed	AST_CEL_CHAN_END	Alice	

Multi-participant Conference

The following scenario demonstrates conversion of a bridge to a multi-participant conference:

Event	Record	Primary	Extra
Channel Alice is created	AST_CEL_CHANNEL_START	Alice	
Channel Bob is created	AST_CEL_CHANNEL_START	Bob	
Channel Charlie is created	AST_CEL_CHANNEL_START	Charlie	
Channel David is created	AST_CEL_CHANNEL_START	David	
Bridge Link is created			
Alice enters bridge Link	AST_CEL_CONF_ENTER	Alice	{"bridge_id", "Link"}
Bob enters bridge Link	AST_CEL_CONF_ENTER	Bob	{"bridge_id", "Link"}
Charlie enters bridge Link	AST_CEL_CONF_ENTER	Charlie	{"bridge_id", "Link"}
David enters bridge Link	AST_CEL_CONF_ENTER	David	{"bridge_id", "Link"}
Alice exits bridge Link	AST_CEL_CONF_EXIT	Alice	{"bridge_id", "Link"}
Alice is destroyed	AST_CEL_CHAN_END	Alice	
Bob exits bridge Link	AST_CEL_CONF_EXIT	Bob	{"bridge_id", "Link"}
Bob is destroyed	AST_CEL_CHAN_END	Bob	
Charlie exits bridge Link	AST_CEL_CONF_EXIT	Charlie	{"bridge_id", "Link"}
Charlie is destroyed	AST_CEL_CHAN_END	Charlie	
David exits bridge Link	AST_CEL_CONF_EXIT	David	{"bridge_id", "Link"}

David is destroyed	AST_CEL_CHAN_END	David	

Dial Nominal

For this scenario, assume that AST_CEL_ANSWER, AST_CEL_HANGUP, AST_CEL_APP_START, and AST_CEL_APP_END are configured in addition to the aforementioned record types and that "Dial" is configured to be watched.

The following scenario demonstrates a Dial that results in an answer followed by bridging and hangup:

Event	Record	Primary	Extra
Channel Alice is created	AST_CEL_CHANNEL_START	Alice	
Alice executes Dial(SIP/Bob)	AST_CEL_APP_START	Alice	
Channel Bob is created	AST_CEL_CHANNEL_START	Bob	
Bob answers	AST_CEL_ANSWER	Bob	
Alice answers	AST_CEL_ANSWER	Alice	
Bridge Link is created			
Alice enters bridge Link	AST_CEL_BRIDGE_ENTER	Alice	{"bridge_id": "Link"}
Bob enters bridge Link	AST_CEL_BRIDGE_ENTER	Bob	{"bridge_id": "Link"}
Bob initiates hangup, exits bridge Link	AST_CEL_BRIDGE_EXIT	Bob	{"bridge_id": "Link"}
Bob completes hang up	AST_CEL_HANGUP	Bob	{"hangupcause":16,"dialstatus":"","hangupsource":"Bob"}
Bob is destroyed	AST_CEL_CHAN_END	Bob	
Alice exits bridge Link	AST_CEL_BRIDGE_EXIT	Alice	{"bridge_id": "Link"}
Alice is hung up	AST_CEL_HANGUP	Alice	{"hangupcause":16,"dialstatus":"ANSWER","hangupsource":""}
Alice is destroyed	AST_CEL_CHAN_END	Alice	

Dial Busy

For this scenario, assume that AST_CEL_ANSWER, AST_CEL_HANGUP, AST_CEL_APP_START, and AST_CEL_APP_END are configured in addition to the aforementioned record types and that "Dial" is configured to be watched. The following scenario demonstrates a Dial that results in a busy:

Event	Record	Primary	Extra
Channel Alice is created	AST_CEL_CHANNEL_START	Alice	
Alice executes Dial(SIP/Bob)	AST_CEL_APP_START	Alice	

Channel Bob is created	AST_CEL_CHANNEL_START	Bob	
Bob responds BUSY	AST_CEL_HANGUP	Bob	{"hangupcause":21,"dialstatus":"","hangupsource":""}
Bob is destroyed	AST_CEL_CHAN_END	Bob	
Alice is hung up	AST_CEL_HANGUP	Alice	{"hangupcause":17,"dialstatus":"BUSY","hangupsource":""}
Alice is destroyed	AST_CEL_CHAN_END	Alice	

Blind Transfer

For this scenario, assume that AST_CEL_HANGUP is configured in addition to the aforementioned record types. The following scenario demonstrates a blind transfer:

Event	Record	Primary	Extra
Channel Alice is created	AST_CEL_CHANNEL_START	Alice	
Channel Bob is created	AST_CEL_CHANNEL_START	Bob	
Alice answers	AST_CEL_ANSWER	Alice	
Bob answers	AST_CEL_ANSWER	Bob	
Bridge Link is created			
Bob enters bridge Link	AST_CEL_BRIDGE_ENTER	Bob	{"bridge_id":"Link"}
Alice enters bridge Link	AST_CEL_BRIDGE_ENTER	Alice	{"bridge_id":"Link"}
Alice initiates a blind transfer to exten@context	AST_CEL_BLINDTRANSFER	Alice	{"bridge_id":"Link","extension":"exten","context":"context"}
Alice exits bridge Link	AST_CEL_BRIDGE_EXIT	Alice	{"bridge_id":"Link"}
Alice is hung up	AST_CEL_HANGUP	Alice	{"hangupcause":16,"dialstatus":"","hangupsource":""}
Alice is destroyed	AST_CEL_CHANNEL_END	Alice	
A local channel pair is created to handle dialplan	AST_CEL_CHANNEL_START	Local1	
	AST_CEL_CHANNEL_START	Local2	
Local1 enters bridge Link	AST_CEL_BRIDGE_ENTER	Local1	{"bridge_id":"Link"}
Local2 executes dialplan at exten@context			
Local2 is eventually hung up by the dialplan	AST_CEL_HANGUP	Local2	{"hangupcause":16,"dialstatus":"","hangupsource":""}
Hangup is initiated on Local1, exiting bridge Link	AST_CEL_BRIDGE_EXIT	Local1	{"bridge_id":"Link"}
Local1 is hung up	AST_CEL_HANGUP	Local1	{"hangupcause":16,"dialstatus":"","hangupsource":""}

Local1 is destroyed	AST_CEL_CHANNEL_END	Local1	
Local2 is destroyed	AST_CEL_CHANNEL_END	Local2	
Bob is the last channel and so is hung up	AST_CEL_HANGUP	Bob	{"hangupcause":16,"dialstatus":"","hangupsource":""}
Bob is destroyed	AST_CEL_CHANNEL_END	Bob	

Attended Transfer

For this scenario, assume that AST_CEL_ANSWER and AST_CEL_HANGUP are configured in addition to the aforementioned record types. The following scenario demonstrates a channel-swapping attended transfer:

Event	Record	Primary	Extra
Channel Alice is created	AST_CEL_CHANNEL_START	Alice	
Channel Bob is created	AST_CEL_CHANNEL_START	Bob	
Alice answers	AST_CEL_ANSWER	Alice	
Bob answers	AST_CEL_ANSWER	Bob	
Bridge Link1 is created			
Bob enters bridge Link1	AST_CEL_BRIDGE_ENTER	Bob	{"bridge_id":"Link1"}
Alice enters bridge Link1	AST_CEL_BRIDGE_ENTER	Alice	{"bridge_id":"Link1"}
Channel Charlie is created	AST_CEL_CHANNEL_START	Charlie	
Channel David is created	AST_CEL_CHANNEL_START	David	
Charlie answers	AST_CEL_ANSWER	Charlie	
David answers	AST_CEL_ANSWER	David	
Bridge Link2 is created			
David enters bridge Link2	AST_CEL_BRIDGE_ENTER	Bob	{"bridge_id":"Link2"}
Charlie enters bridge Link2	AST_CEL_BRIDGE_ENTER	Alice	{"bridge_id":"Link2"}
An attended transfer between Alice and David begins			
Bob exits bridge Link1	AST_CEL_BRIDGE_EXIT	Bob	{"bridge_id":"Link1"}
Bob enters bridge Link2	AST_CEL_BRIDGE_ENTER	Bob	{"bridge_id":"Link2"}
David exits bridge Link2	AST_CEL_BRIDGE_EXIT	David	{"bridge_id":"Link2"}
David is hung up	AST_CEL_HANGUP	David	{"hangupcause":16,"dialstatus":"","hangupsource":""}
David is destroyed	AST_CEL_CHANNEL_END	David	
Alice and David execute an attended transfer	AST_CEL_ATTENDEDTRANSFER	Alice	{"bridge1_id":"Link1","channel2_name":"David","bridge2_id":"Link2"}
Alice exits bridge Link1	AST_CEL_BRIDGE_EXIT	Alice	{"bridge_id":"Link1"}

Alice is hung up	AST_CEL_HANGUP	Alice	{"hangupcause":16,"dialstatus":"" ,"hangupsource":""}
Alice is destroyed	AST_CEL_CHANNEL_END	Alice	
Bob exits bridge Link2	AST_CEL_BRIDGE_EXIT	Bob	{"bridge_id":"Link2"}
Charlie exits bridge Link2	AST_CEL_BRIDGE_EXIT	Charlie	{"bridge_id":"Link2"}
Bob is hung up	AST_CEL_HANGUP	Bob	{"hangupcause":16,"dialstatus":"" ,"hangupsource":""}
Bob is destroyed	AST_CEL_CHANNEL_END	Bob	
Charlie is hung up	AST_CEL_HANGUP	Charlie	{"hangupcause":16,"dialstatus":"" ,"hangupsource":""}
Charlie is destroyed	AST_CEL_CHANNEL_END	Charlie	

Note that the ATTENDEDTRANSFER event does not necessarily occur before or after the records it is related to.

Getting Started

A Beginners Guide to Asterisk. Herein, you will find content related to installing Asterisk and basic usage concepts.

Precursors, Background and Business

Discovering Asterisk

This section of the documentation attempts to explain at a high level what Asterisk is and does. It also attempts to provide primers on the key technical disciplines that are required to successfully create and manage Asterisk solutions. Much of the material in this section is optional and may be redundant for those with a background in communications application development. For the other 99.9875% of the population, this is good stuff. Read on...

Asterisk Concepts

Asterisk is a very large application that does many things. It can be somewhat difficult to understand, especially if you are new to communications technologies. In the next few chapters we will do our best to explain what Asterisk is, what it is not, and how it came to be this way. This section doesn't cover the technology so much as the concept. If you're already familiar with the function of a telephony engine, feel free to jump ahead to the next section.

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.

A Brief History of the Asterisk Project

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.

Rather than give in and take out a small business loan, Mark made a fateful 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.

Beginning Asterisk

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 an Asterisk-based Linux distribution such as AsteriskNOW. This takes care of installing Linux, Asterisk, and some web-based interfaces all at the same time, and 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 AsteriskNOW

Thank you for downloading AsteriskNOW. This Linux distribution has been carefully customized and tested with Asterisk, and installs all of the packages needed for its use. It is the officially recommended development and runtime platform for Asterisk and Digium hardware, including Digium phones.

This guide provides a brief overview of installation, configuration, and maintenance of your system.

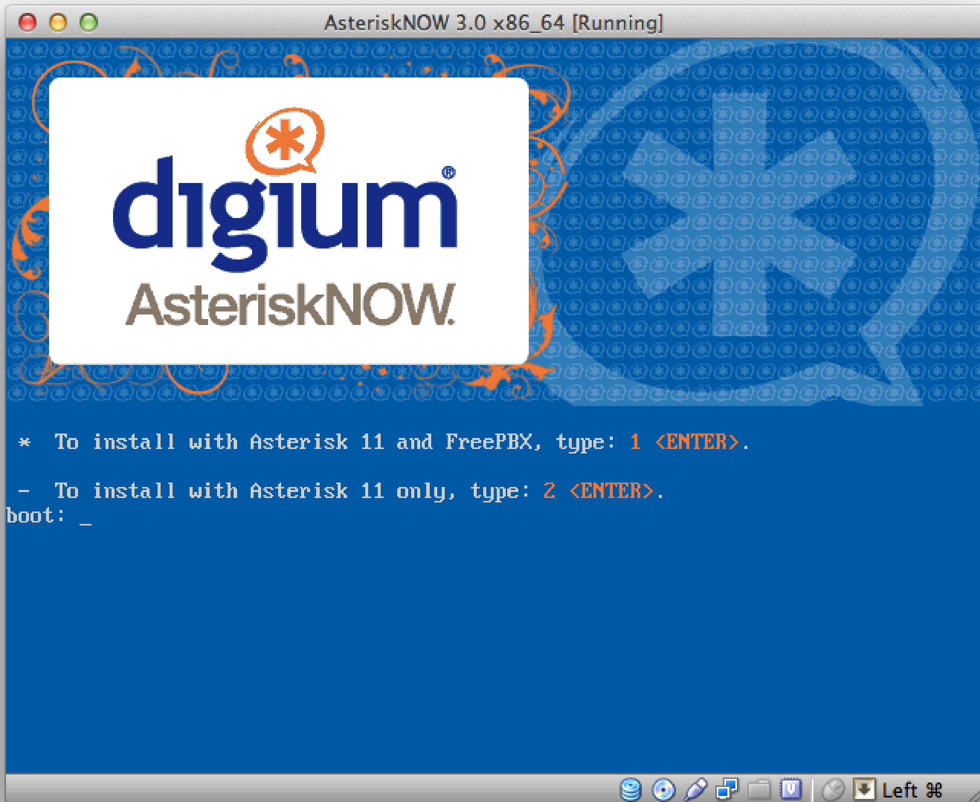
More information is available at http://wiki.centos.org/.

Please report any bugs at https://issues.asterisk.org/jira

Installation

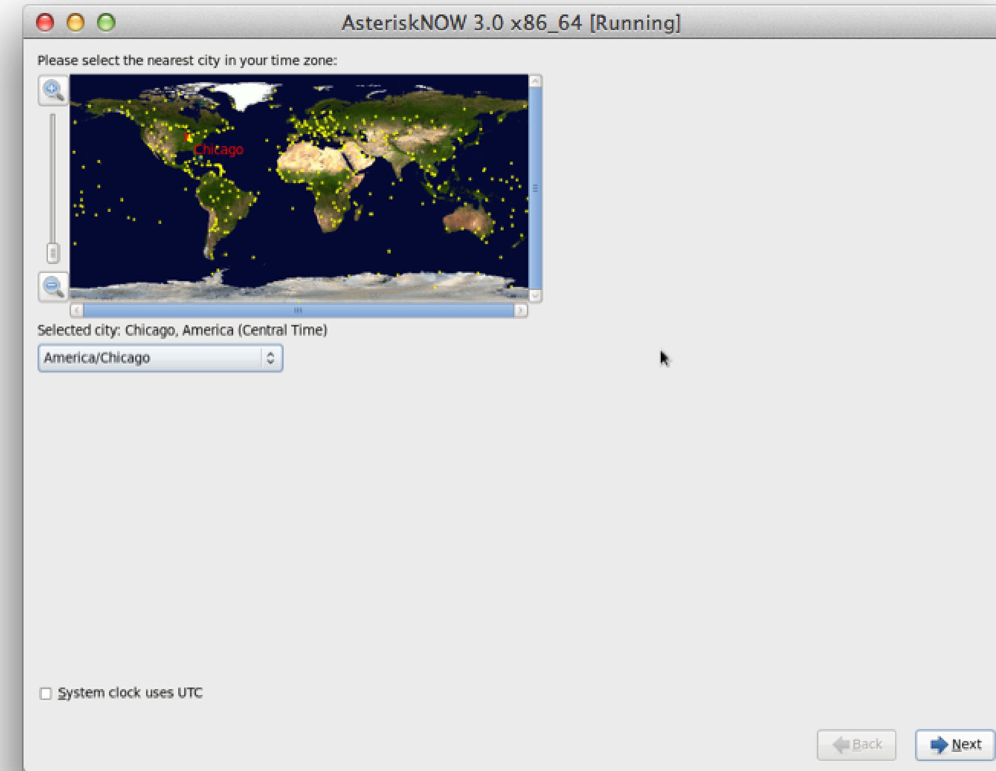
- Burn the AsteriskNOW DVD image to DVD disc and then boot from the DVD to begin the installation process.
  - If you are unfamiliar with burning disc images, the Ubuntu community has a great Burning ISO Howto available at https://help.ubuntu.com/community/BurningIsoHowto.
  - If you are unfamiliar with booting to DVD, the Ubuntu community has a wonderful Boot From DVD HOWTO available at https://help.ubuntu.com/community/BootFromCD.

- After booting from the AsteriskNOW DVD, you will be presented with the following screen and options for an installation with, or without the FreePBX web interface. This QuickStart assumes that the FreePBX web interface has been installed. To do this, selection option 1 and press <ENTER>:



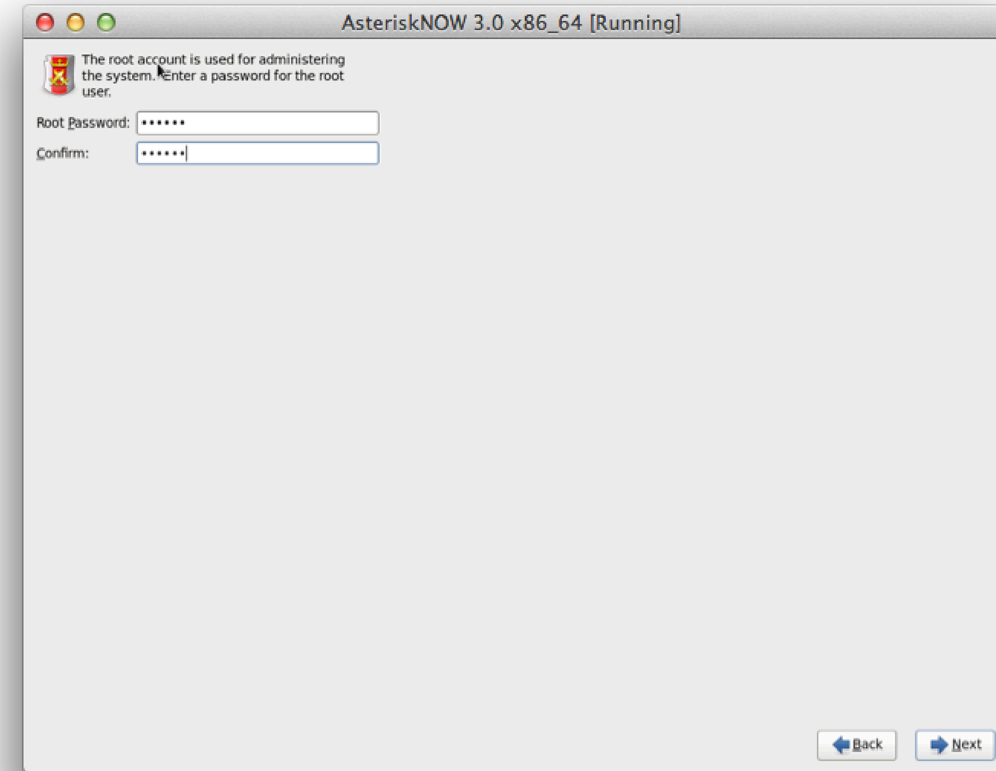
ⓘ This will begin the automated graphical installation process.

- During the installation, you are first presented with an option for setting the system Time Zone:



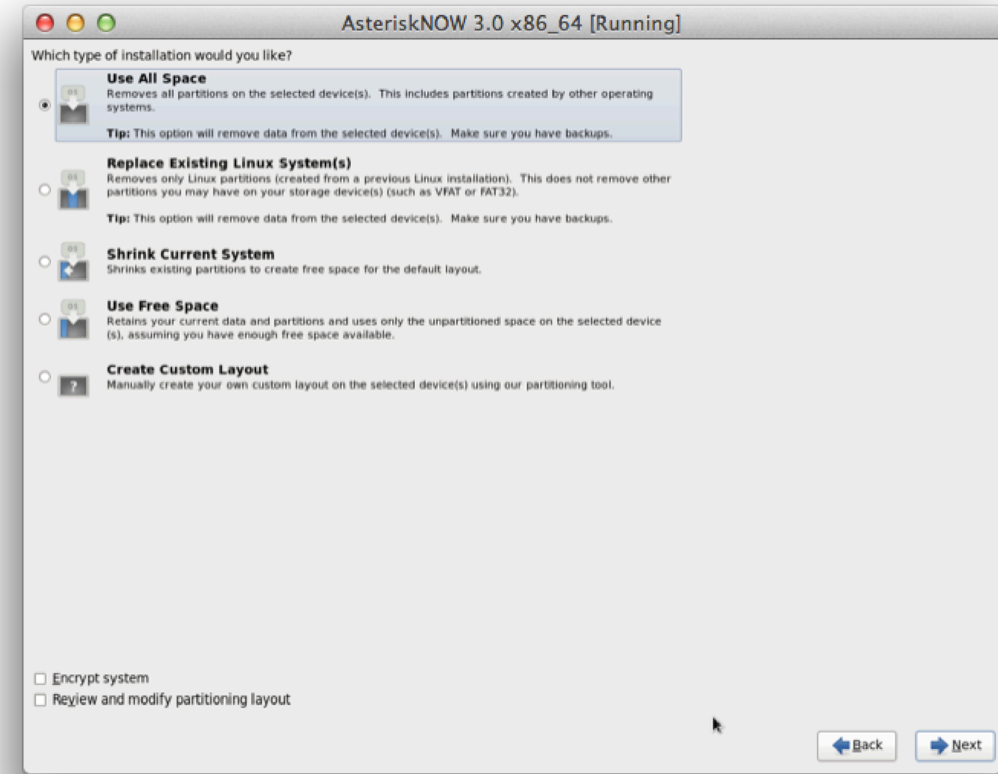
** Choose the location that is nearest to you and move to the next screen.

- Next, you will be prompted to set a root password:



> The 'root' user is the administrative account for Linux systems. Most system configuration requires 'root' access. If this password is lost, it is impossible to recover. It is recommend that your password contain a mix of lowercase and UPPERCASE letters, numbers, and/or symbols. Or, if you're into entropy, try a pass phrase.

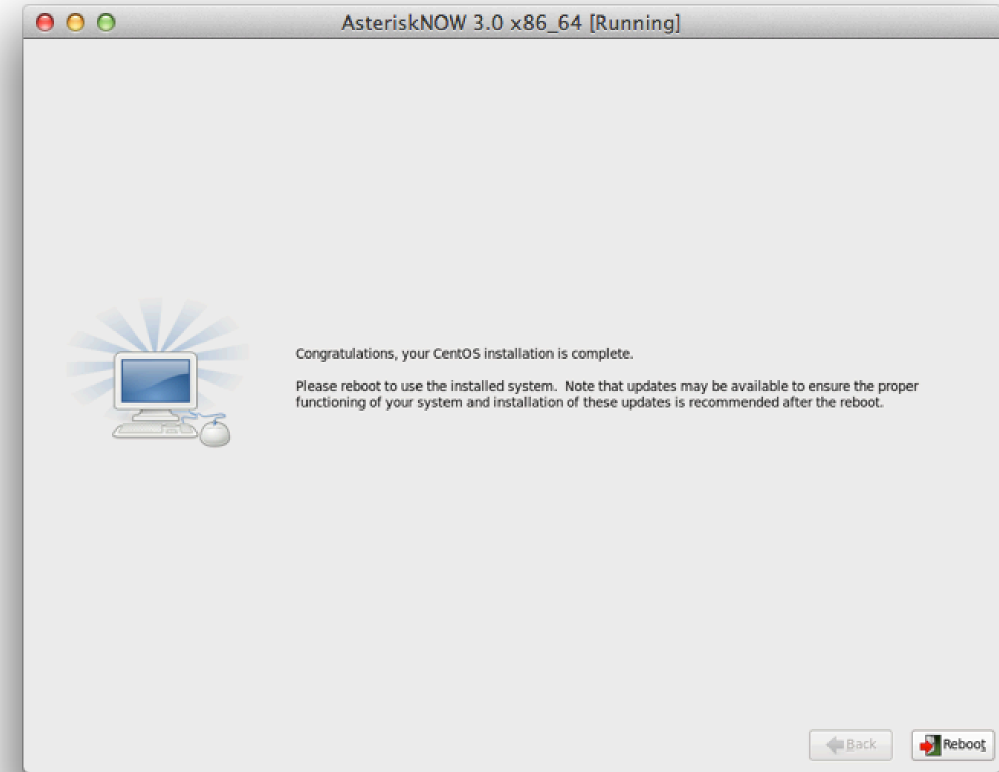
- Then, you will choose your Hard Disk Layout:



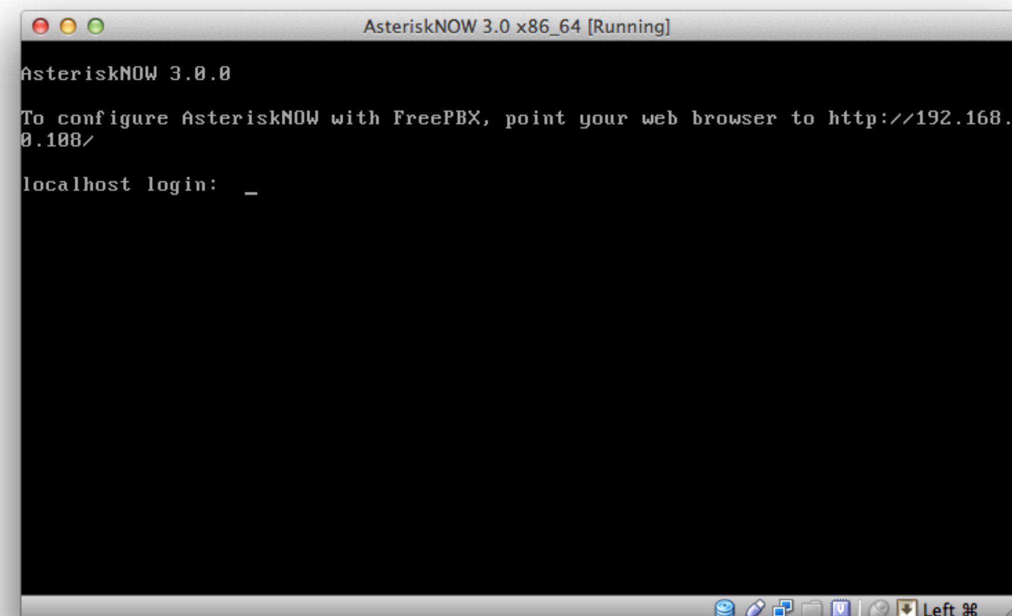
It is recommended to select "Use All Space" and move to the next screen.

- Now, sit back, relax, have a cup of coffee and wait while the system is installed. This will take approximately 15-30 minutes. You will see a progress bar indicating the installation status.

Once installation has completed, you will be prompted to reboot into your installation:



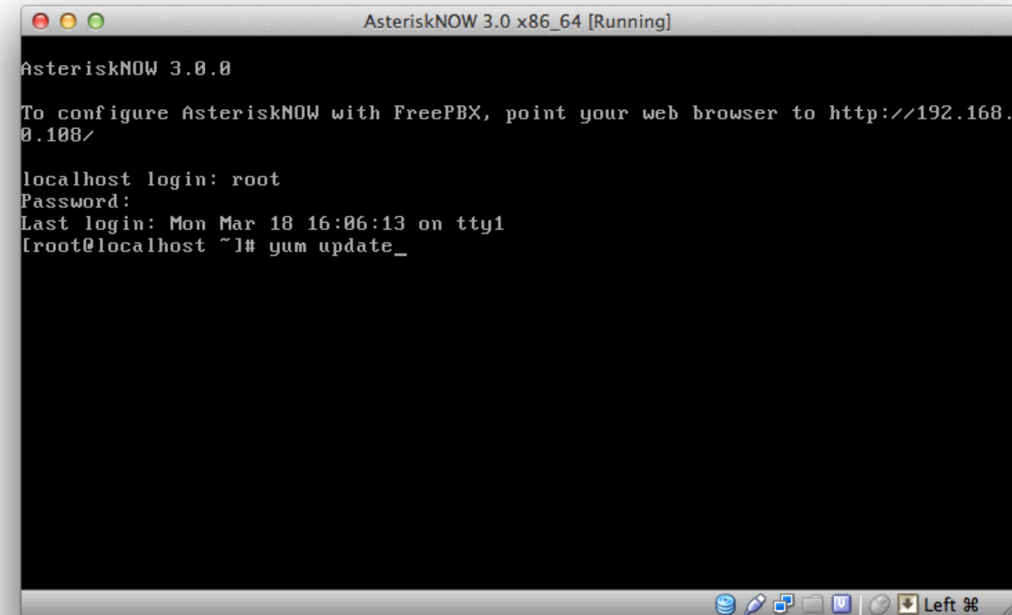
- After the system reboots you will see this screen:



Congratulations! You have successfully installed AsteriskNOW.

Notice the text that says "To configure AsteriskNOW with FreePBX, point your web browser to http://xx.xx.xx.xx/." Write this down, you will need it in the next section.

- Now, before you move on, it is important to update your AsteriskNOW system to the latest Linux packages. To do this, use the yum utility "yum." Perform a "yum update"

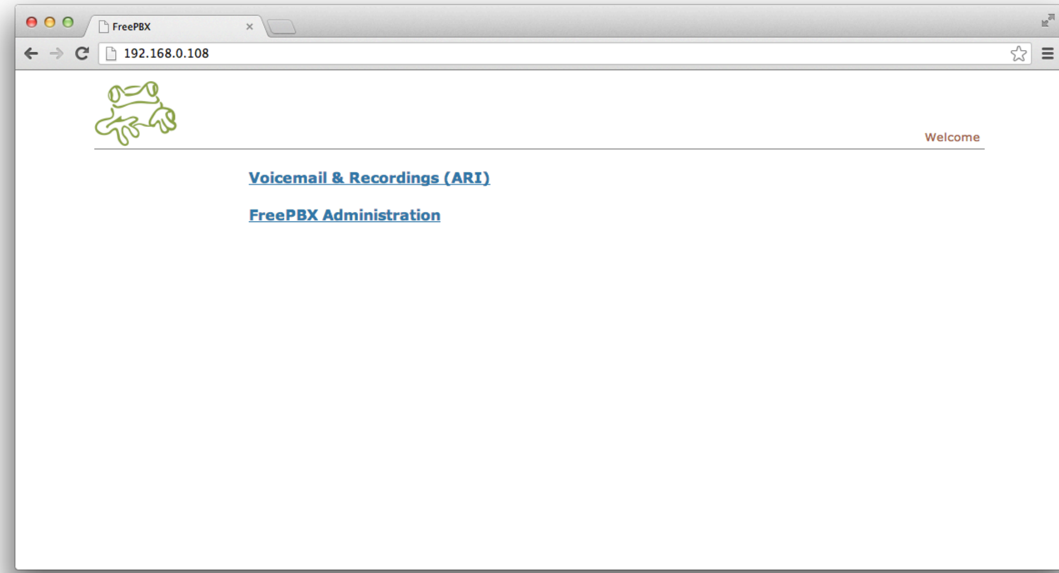


** If new packages are available for installation, the utility will ask permission to install them. And, if the utility has not been run before, it may ask permission to accept a yum key. You should accept both to stay up to date.

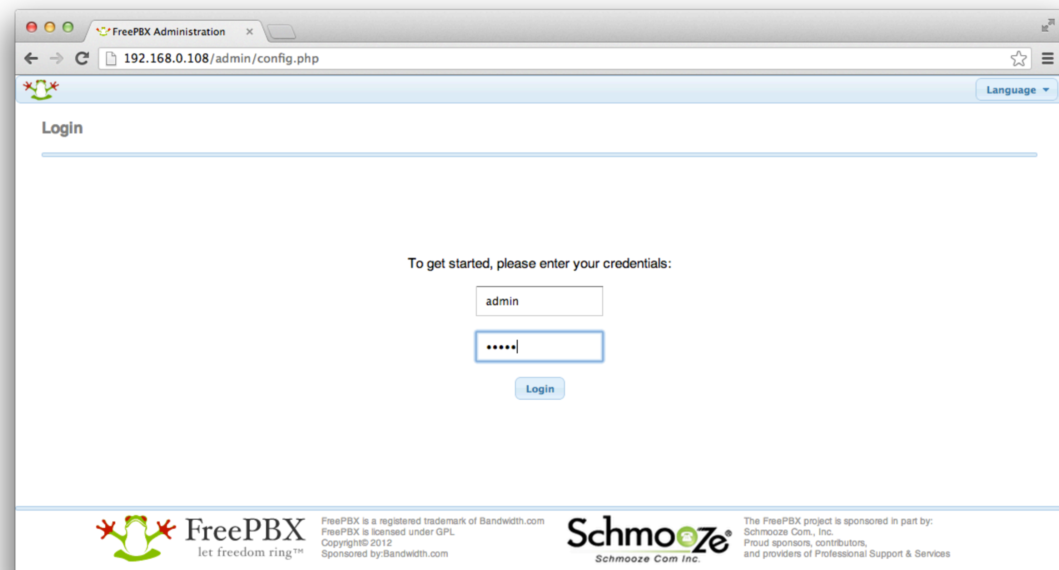
- You are now ready to move on to configuration of AsteriskNOW from the FreePBX web interface.

FreePBX Configuration

- To configure your system using FreePBX, open a web browser on another PC to the address specified during boot, e.g. "To configure AsteriskNOW with FreePBX, point your web browser to http://xx.xx.xx.xx/. If successful, you will be presented with the FreePBX main screen:

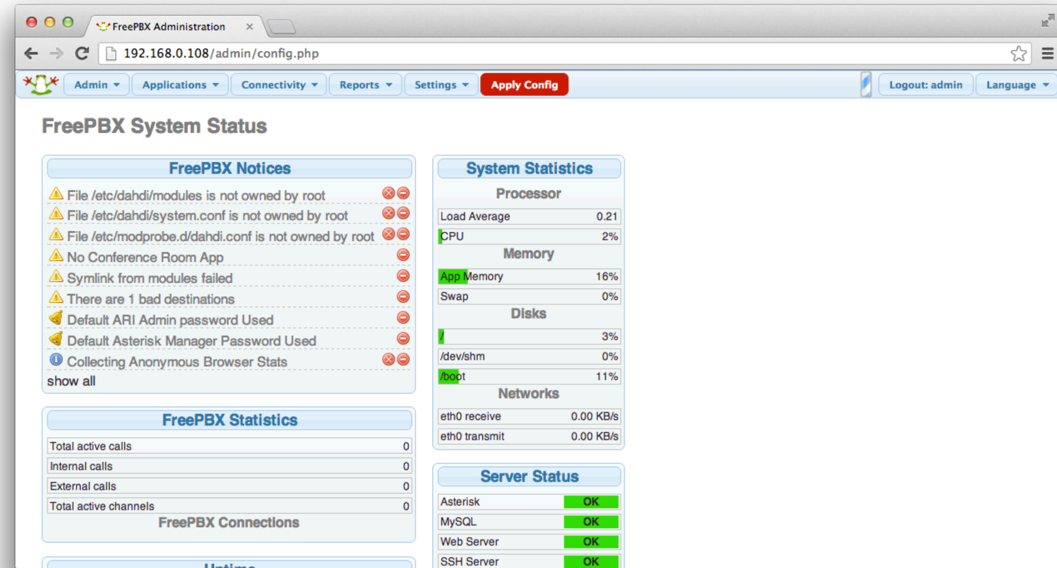


- From here, we want the "FreePBX Administration" link. Click it, and you will see the FreePBX login screen:



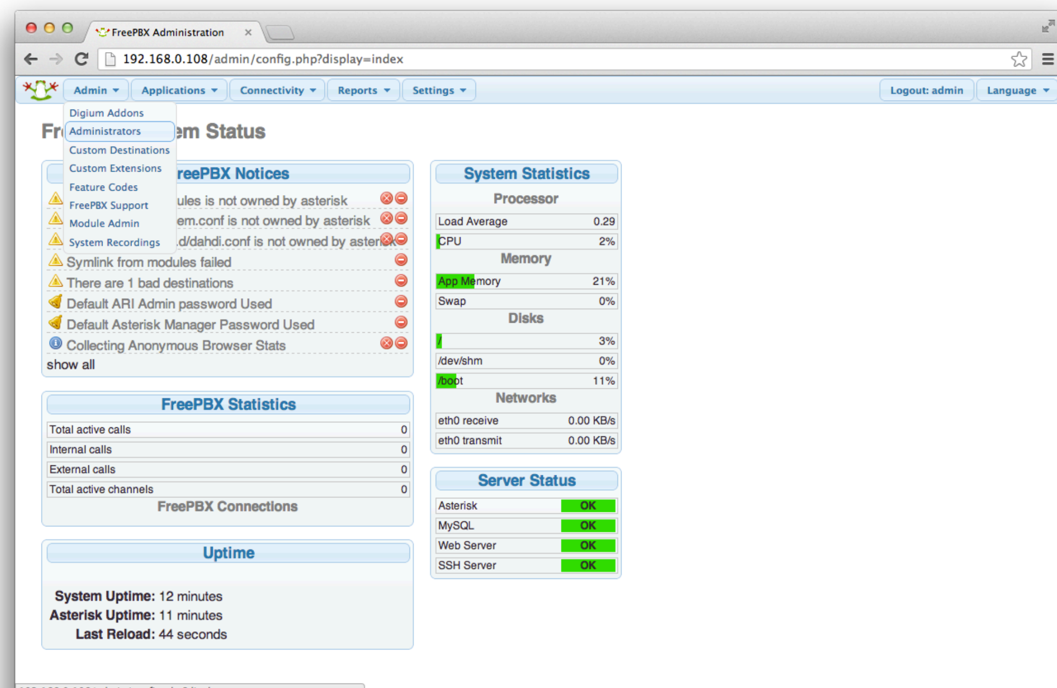
ⓘ The default username is **admin**
The default password is **admin**

- Having successfully logged into FreePBX, you will see the FreePBX dashboard:

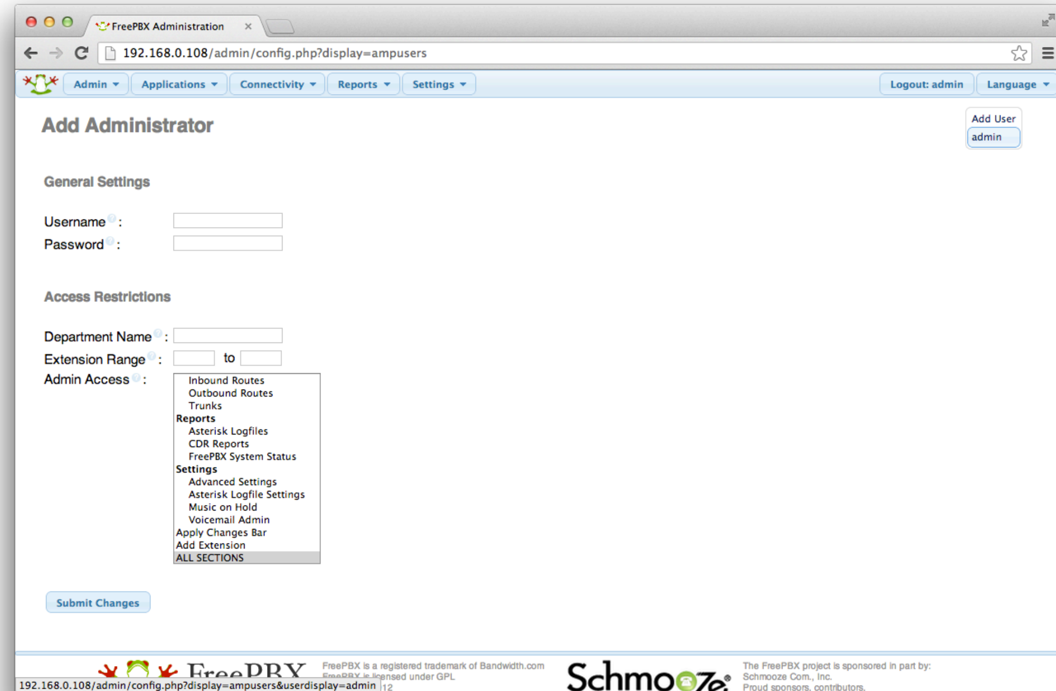


Notice the **Red** reload button. It will appear after changes are made to any page. If you see it, it should be clicked, it will affect any changes on the system that FreePBX needs to make. This guide assumes that whenever you see it, you will click it.

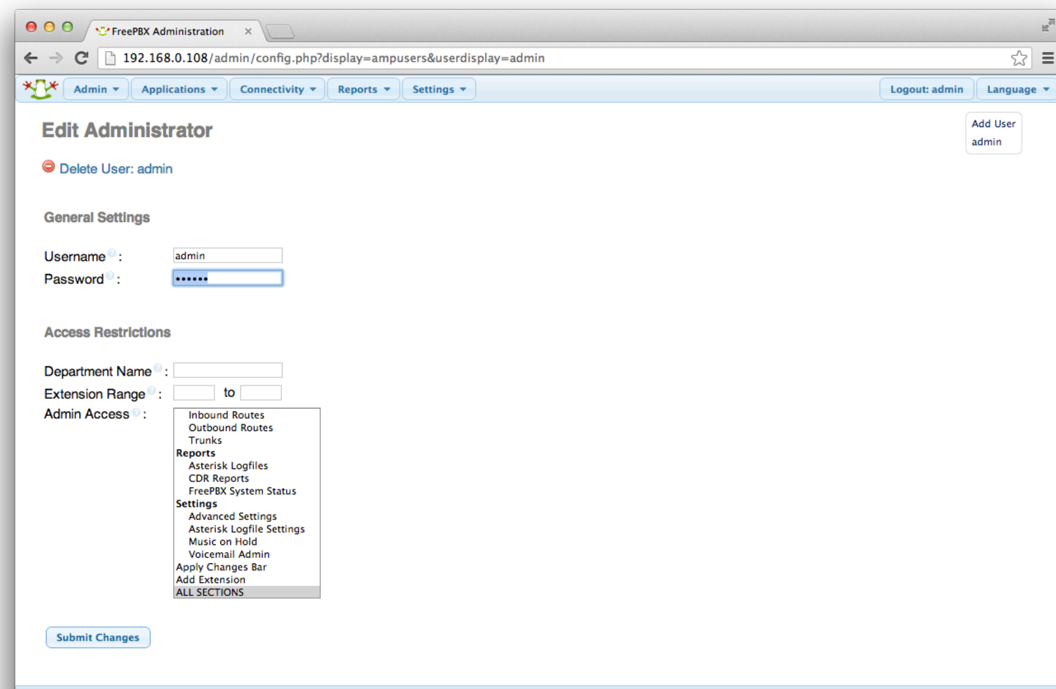
- Next, we will change the default admin password. This is imperative! Failure to do this is inviting disaster. The importance of doing this **C ANNOT be understated.**
  - First, visit the Admin tool



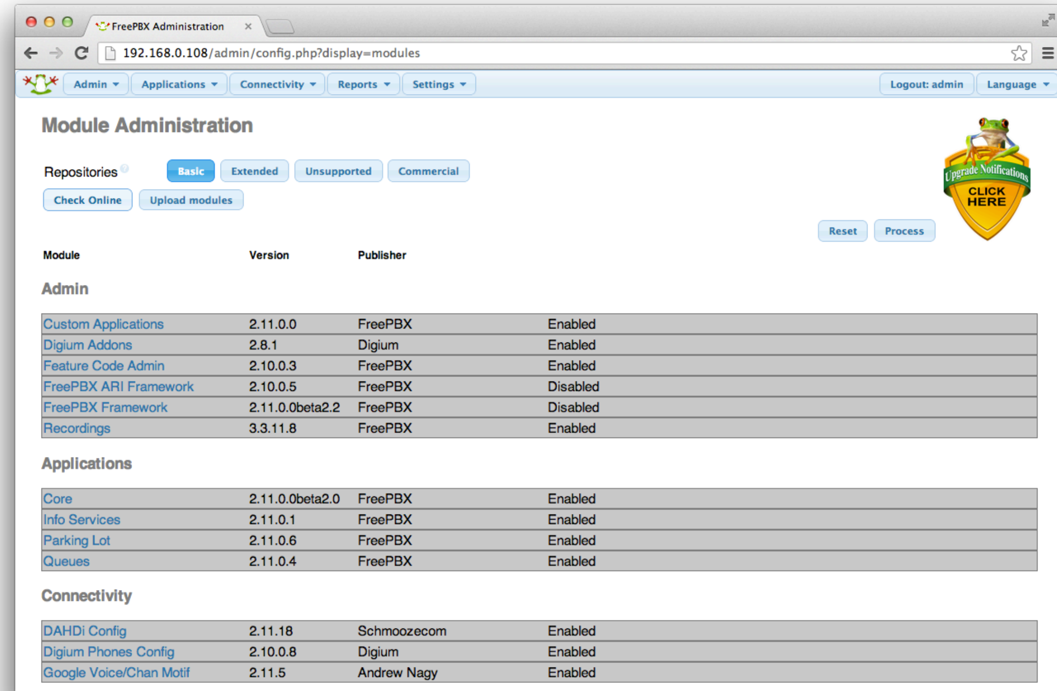
  - Next, select admin from the right column



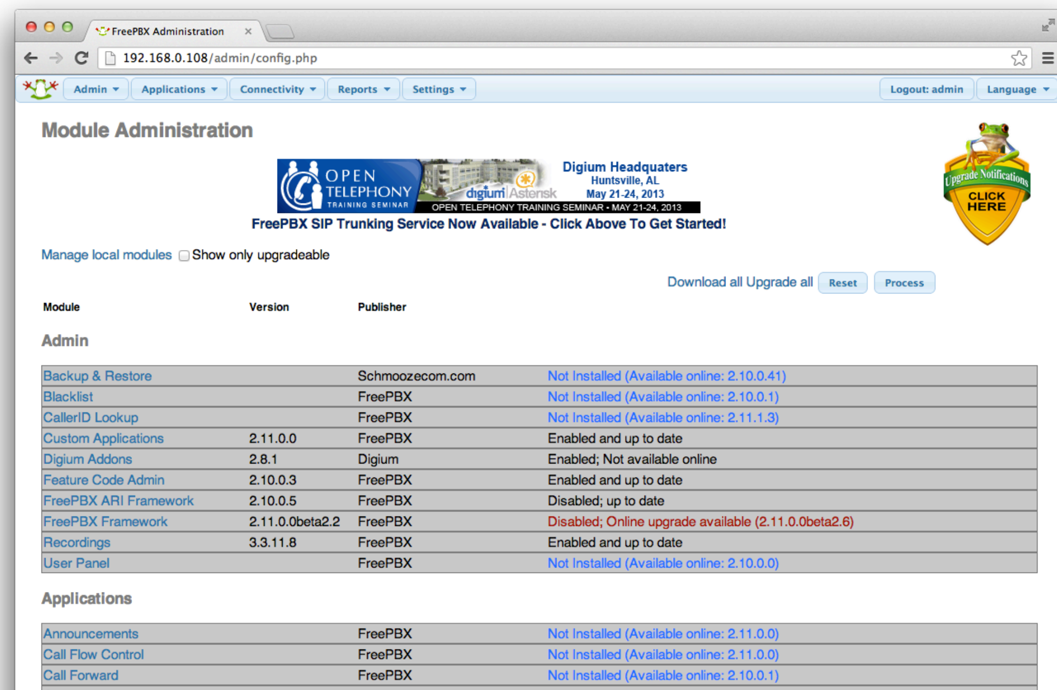
- Then, change the admin password



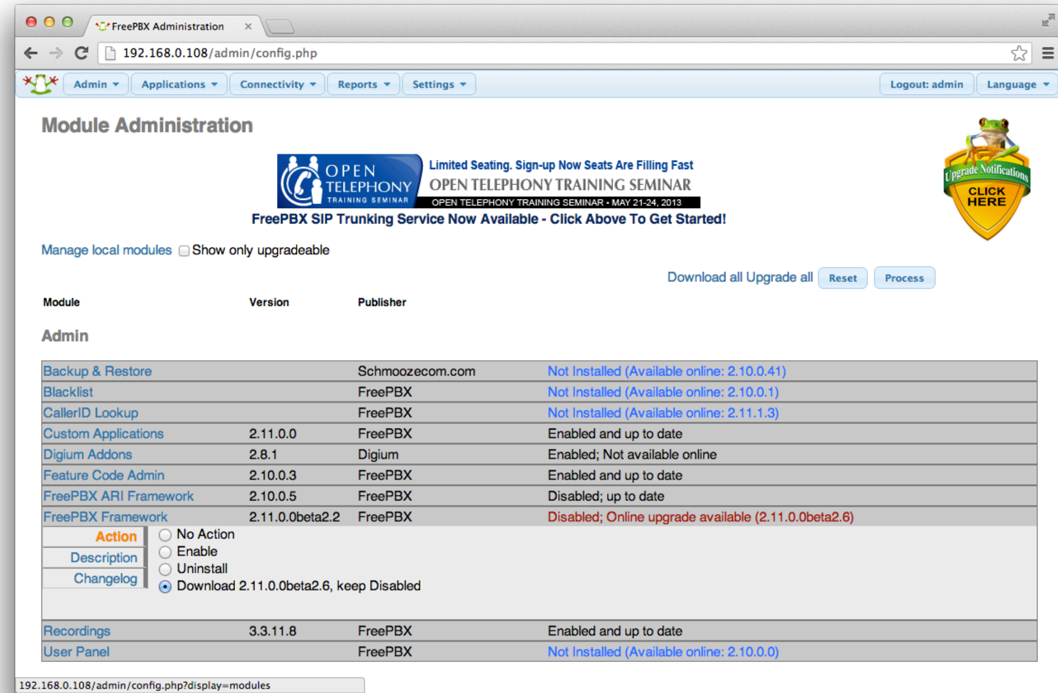
- Finally, one should update any out of date modules on the system. To do this, we will visit the Module Admin tool:



- Click the Check Online button and you will see any out of date modules



- To update a module, click it, and then select the Download option



- Finally, press the Process button and follow the instructions to complete the module update.

Updating, Querying, Removing Packages

After completing installation of AsteriskNOW, all of the packages for running Asterisk are installed. However system updates are often available.

AsteriskNOW contains several yum repositories in addition to the ones provided by CentOS. These are asterisk-current/asterisk-tested and digium-current/digium-tested. The asterisk- repositories contain packages for Digium-provided Open Source software (such as Asterisk, libpri, and DAHDI). The digium- repositories contain non-free or commercial software (such as the Digium Phone module for Asterisk, G.729 for Asterisk, Fax For Asterisk, and the HPEC echo cancellation module). This allows you to install additional software and to stay up to date with the latest changes.

Packages can be installed or removed by using `yum install <package>` or `yum remove <package>`. Updates should be regularly installed by using `yum update`. For a very full list of available and installed packages, you can use `yum list | less`. For more information about Yum, visit http://yum.baseurl.org/wiki/YumCommands or type `man yum`.

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. Packages for Red Hat Enterprise Linux and CentOS are also available at http://packages.asterisk.org/ (see Asterisk Packages for instructions on use).

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. After logging in as the **root** user 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-1.6.X.Y]# lsmod | grep dahdi
```

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

```
[root@server asterisk-1.6.X.Y]# service dadhi 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 ~]# lsmod | grep dahdi

dahdi_dummy          4288   0

dahdi_transcode      7928   1 wctc4xxp

dahdi_voicebus       40464  2 wctdm24xxp,wcte12xp

dahdi                196544 12
dahdi_dummy,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 ~]# service asterisk status
asterisk is stopped
```

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

```
[root@server ~]# service 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 by using the command below. (The process identifier, or pid, will obviously be different on your system.)

```
[root@server ~]# service asterisk status
asterisk (pid 32117) is running...
```

And there you have it... you have an Asterisk system up and running! You should now continue on in Section 202. Getting Started with Asterisk.

Asterisk Configuration Files

Intro to Asterisk Configuration Files

In this section, we'll introduce you to the Asterisk configuration files, and show you how to use some advanced features.

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. 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
```

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 and exec Constructs

There are two other constructs we can use within 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 **#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.

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.

Let's look at example of both constructs in action.

```
[section-name]
setting=true
#include otherconfig.conf      ; include another configuration file
#exec otherprogram             ; include output of otherprogram
```

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.

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, place an exclamation mark in parentheses after the section heading, as shown in the example below.

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

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

[section-name](template-name,template-2)
setting3=value3
```

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
```

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

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 access to two 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. Soft phones 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.

Creating SIP Accounts

In order for our two 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 both the phones are using the SIP protocol, we'll configure accounts in the SIP channel driver configuration file, called **sip.conf**. (This file resides in the Asterisk configuration directory, which is typically **/etc/asterisk**.) Let's name your phones **Alice** and **Bob**, so that we can easily differentiate between them.

Open **sip.conf** with your favorite text editor, and spend a minute or two looking at the 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. Scroll to the bottom of the file, and add a section for Alice and Bob. You'll need to choose your own unique password for each account, and change the **permit** line to match the settings for your local network.

```
[demo-alice]
type=friend
host=dynamic
secret=verysecretpassword ; put a strong, unique password here instead
context=users
deny=0.0.0.0/0
permit=192.168.5.0/255.255.255.0 ; replace with your network settings

[demo-bob]
type=friend
host=dynamic
secret=othersecretpassword ; put a strong, unique password here instead
context=users
deny=0.0.0.0/0
permit=192.168.5.0/255.255.255.0 ; replace with your network settings
```

Be Serious About Account Security

We can't stress enough how important it is for you to pick a strong password for all accounts on Asterisk, and to only allow access from trusted networks. Unfortunately, we've found many instances of people exposing their Asterisk to the internet at large with easily-guessable passwords, or no passwords at all. You could be at risk of toll fraud, scams, and other malicious behavior.

For more information on Asterisk security and how you can protect yourself, check out http://www.asterisk.org/security/webinar/.

After adding the two sections above to your **sip.conf** file, go to the Asterisk command-line interface and run the **sip reload** command to tell Asterisk to re-read the **sip.conf** configuration file.

```
server*CLI> sip reload

Reloading SIP

server*CLI>
```

Reloading Configuration Files

Don't forget to reload the appropriate Asterisk configuration files after you have made changes to them.

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.

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.

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" 
<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'll cover contexts more in Dialplan Fundamentals, 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.

Go to the bottom of your **extensions.conf** file, and add a new context named [users].
Naming Your Dialplan Contexts

There's nothing special about the name **users** 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 for the users of our PBX system, we'll call our context **users**.

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.

```
[users]
exten=>6001,1,Dial(SIP/demo-alice,20)
exten=>6002,1,Dial(SIP/demo-bob,20)
```

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 users** at the CLI.

```
server*CLI> dialplan show users
[ Context 'users' 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!

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@users: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 (users, 6002, 1) exited non-zero on
'SIP/demo-alice-00000000'
```

As you can see, Alice called extension **6002** in the [users] 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** at the Asterisk CLI.

To see which context your SIP phones will send calls to, type **sip show users**.

To ensure that you've created the extensions correctly in the **[users]** context in the dialplan, type **dialplan show users**.

To see which extension will be executed when you dial extension **6002**, type **dialplan show 6002@users**.

Sound Prompt Searching based on Channel Language

Dialplan Fundamentals

The dialplan is essential to the operation of any successful Asterisk system. In this module, we'll help you learn the fundamental components of the Asterisk dialplan, and how to combine them to begin scripting your own dialplan. We'll also add voice mail and a dial-by-name directory features to your dialplan.

Contexts, Extensions, and Priorities

The dialplan is organized into various 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. We'll use contexts to enforce security boundaries between the various parts of our dialplan, as well as to provide different classes of service to groups of users.

The syntax for a context is exactly the same as any other section heading in the configuration files, as explained in Section 206.2.1. Sections and Settings. Simply place the context name in square brackets. For example, here is the context we defined in the previous module:

```
[users]
```

Within each context, we can define one or more **extensions**. As explained in the previous module, 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...]])
```

As an example, let's review extension **6001** from the previous module. It looks like:

```
exten => 6001,1,Dial(SIP/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 **SIP/demo-alice** and **20**.
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 priorites one and two, but would then terminate the call, because it couldn't find priority number three.

Priority number can also be simplied 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,do something
exten => 6123,n,do something else
exten => 6123,n,do something different
```

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,do something
exten => 6123,n(repeat),do something else
exten => 6123,n,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 => _1NXXNXXXXXX,1,do something
same => n(repeat),do something else
same => n,do something different
```

Applications

Each priority in the dialplan calls an application. An application does 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. For a complete list of the dialplan applications available to your installation of Asterisk, type **core show applications** at the Asterisk CLI.

Most applications take one or more parameters, which provide additional information to the application or change its behavior. Parameters should be separated by commas.

Syntax for Parameters

You'll often find examples of Asterisk dialplan code online and in print which use the pipe character or vertical bar character (|) between parameters, as shown in this example:

```
exten => 6123,1,application(one|two|three)
```

This is a deprecated syntax, and will no longer work in newer versions of Asterisk. Simply replace the pipe character with a comma, like this:

```
exten => 6123,1,application(one,two,three)
```

Answer, Playback, and Hangup Applications

As its name suggests, the **Answer()** application answers an incoming call. The **Answer()** applica tion 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)
```

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, an 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

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**.

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.

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. Place this extension in your **[docs:users]** context:

```
exten => 6000,1,Answer(500)
exten => 6000,n,Playback(hello-world)
exten => 6000,n,Hangup()
```

Dial Application

Now that you've learned the basics of using dialplan applications, let's take a closer look at the **Dial()** application that we used earlier in extensions **6001** and **6002**. **Dial()** attempts to ring an external device, and if the call is answered it bridges the two channels together and does any necessary protocol or CODEC conversion. It also handles call progress responses (busy, no-answer, ringing).

> **Dial() and the Dialplan**
> Please note that if the Dial() application successfully bridges two channels together, that the call does not progress in the dialplan. The call will only continue on to the next priority if the Dial() application is unable to bridge the calling channel to the dialed device.

The **Dial()** application takes four parameters:

1. **Devices**
    - A list of the device(s) you want to call. Devices are specified as technology or channel driver, a forward slash, and the device or account name. For example, **SIP/demo-alice** would use the SIP channel driver to call the device specified in the **demo-alice** section of **sip.conf**. Devices using the IAX2 channel driver take the form of **IAX2/demo-george**, and DAHDI channels take the form of **DAHDI/1**.
    - When calling through a device (such as a gateway) or service provider to reach another number, the syntax is **technology/device/number** such as **SIP/my_provider/5551212** or **DAHDI/4/5551212**.
    - To dial multiple devices at once, simply concatenate the devices together whith the ampersand character (**&**). The first device to answer will get bridged with the caller, and the other endpoints will stop ringing.

    ```
    exten => 6003,1,Dial(SIP/demo-alice&SIP/demo-bob,30)
    ```

2. **Timeout**
    - The number of seconds to allow the device(s) to ring before giving up and moving on to the next priority in the extension.
3. **Options**
    - There are dozens of options that you can set on the outbound call, including call screening, distinctive ringing and more. Type **core show application dial** at the Asterisk CLI for a complete list of all available options. If you want to specify multiple options, simply concatenate them together. For example, if you want to use both the *m*and **H** options, you would set **mH** as the options parameter.
4. **URL**
    - The fourth parameter is a URL that will be sent to the endpoint. Few endpoints do anything with the URL, but there are a few (softphones mostly) that do act on the URL.

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.

VoiceMail Application

The **VoiceMail()** applications 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.

VoiceMailMain Application

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 Voice mail**
> 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.

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** t o 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?

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]** cont ext 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.

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. For more information on DTMF tones, see Section 440.3. DTMF Dialing.

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 **Got**

o() 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]** con text 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. The most common special extensions include:

i: the 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.

t: the reponse timeout extension

When the caller waits too long before entering a response to the **Background()** or **WaitExten()** a pplications, and there are no more priorities in the current extension, the call is sent to the t extension.

s: the 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.

h: the 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.

o: the 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.

a: the 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.

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.

Dialplan Architecture

In this section, we'll begin adding structure to our dialplan. We'll begin by talking about variables and how to use them, as well as how to manipulate them. Then we'll cover more advanced topics, such as pattern matching and using include statements to build classes of functionality.

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 Basics

Channel variables are variables that are set for the current channel (one leg of a bridged phone call). They exist for the lifetime of the channel, and then go away when that channel is hung up.

Channel variables on one particular channel are completely independent of channel variables on any other channels; in other words, two channels could each have variables called COUNT with different values.

To assign a value to a channel variable, we use the **Set()** application. Here's an example of setting a variable called **COUNT** to a value of **3**.

```
exten=>6123,1,Set(COUNT=3)
```

To retrieve the value of a variable, we use a special syntax. We put a dollar sign and curly braces around the variable name, like **${COUNT}**

When Asterisk sees the dollar sign and curly braces around a variable name, it substitutes in the value of the variable. Let's look at an example with the **SayNumber()** application.

```
exten=>6123,1,Set(COUNT=3)
exten=>6123,n,SayNumber(${COUNT})
```

In the second line of this example, Asterisk replaces the **${COUNT}** text with the value of the **CO UNT** variable, so that it ends up calling **SayNumber(3)**.

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[docs::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**.

Variable Inheritance Basics

When building your Asterisk dialplan, it may be useful to have one channel inherit variables from another channel. For example, imagine that Alice's call has a channel variable containing an account code, and you'd like to pass that variable on to Bob's channel when Alice's call gets bridged to Bob. We call this variable inheritance. There are two levels of variable inheritance in Asterisk: *single inheritance* and *multiple inheritance*.

Multiple Inheritance

Multiple inheritance means that a channel variable will be inherited by created (spawned) channels, and it will continue to be inherited by any other channels created by the spawned channels. To set multiple inheritance on a channel, preface the variable name with two underscores when giving it a value with the **Set()** application, as shown below.

```
exten=>6123,1,Set(__ACCOUNT=5551212)
```

Single Inheritance

Single inheritance means that a channel variable will be inherited by created (spawned) channels, but not propogate from there to any other swawned channels. To follow our example above, if Alice sets a channel variable with single inheritance and calls Bob, Bob's channel will inherit that channel variable, but the channel variable won't get inherited by any channels that might get spawned by Bob's channel (if the call gets transferred, for example). To set single inheritance on a channel, preface the variable name with an underscore when giving it a value with the **Set()** application, as shown below.

```
exten=>6123,1,Set(_ACCOUNT=5551212)
```

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 **[docs: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** varia ble. A SIP call, for example, might have a channel name that looks like **SIP/george-0000003b**, for example.

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.

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.

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.

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".

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!

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 **charac ters**. 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.

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 problem, unless you purposely want to fall through to a less specific match.

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 **[docs:features]**. We'll leave our extensions **6001** and **6002** for Alice and Bob in the **[docs:users]** context, and place extensions such as **6500** in the new **[docs: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 **[docs: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 **[docs: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. Asterisk will always try to find a matching extension in the current context first, and only follow the include statement to a new context if there isn't anything that matches in the current context.

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. For more information on configuring a SIP provider, see Section 420. The SIP Protocol. For analog connectivity information, see Section 441. Analog Telephony with DAHDI. For more information on connectivity via digital circuits, see Section 450. Basics of Digital Telephony

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 **[docs: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 **[docs:local]** context, it could only make local and internal calls. On the other hand, if we were to point it at the **[docs: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.

Many people find it instructive to look at a visual diagram at this point, so let's draw ourselves a map of the contexts we've created so far.

Insert graphic showing chain of includes from international through long-distance to local and to users and features

In this graphic, we've illustrated the various contexts and how they work together. We've also shown that Alice's phone is pointed at the **[docs:international]** context, while Bob's phone is only pointed at the **[docs:local]** context.

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.

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?

On a typical system, you'll want to download three components:

- Asterisk
- DAHDI
- libpri

The **libpri** library allows Asterisk to communicate with ISDN connections. (We'll cover more about ISDN connections in Section 450.8, "Intro to ISDN PRI and BRI Connections".) While not always necessary, we recommend you install it on new systems.

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. It should also be installed on new systems, even if you don't immediately plan on using analog or digital connections to your Asterisk system.

DAHDI

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.

> 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.

The current version of libpri, DAHDI, and Asterisk can be downloaded from http://downloads.digi um.com/pub/telephony/.

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. While any C compiler should be able to compile the Asterisk code, we strongly recommend that you use the **GCC** compiler. Not only is it the most popular free C compiler on Linux and Unix systems, but it's also the compiler that the Asterisk developers are using.

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 the C++ portion of GCC as well, as certain Asterisk modules will use it.

System Libraries

In addition to the C compiler, you'll also need a set of system libraries. These libraries are used by Asterisk and must be installed before you can compile Asterisk. 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.

A list of libraries you'll need to install include:

- OpenSSL
- ncurses
- newt
- libxml2
- Kernel headers (for building DAHDI drivers)

We recommend you use the package management system of your operating system to install these libraries before compiling and installing libpri, DAHDI, and Asterisk.

> Help Finding the Right Libraries
>
> If you're installing Asterisk 1.6.1.0 or later, it 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.

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-1.X.Y.tar.gz

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

[root@server src]# tar -zxvf asterisk-1.8.X.Y.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

- With Internet Access
- Without Internet Access

Let's install DAHDI! On Linux, we will use the DAHDI-linux-complete tarball, which contains both the DAHDI Linux drivers as well as the DAHDI tools. 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.

LibPRI 1.4.13 and later source code depends on DAHDI include files. So, as a change from older versions, one must install DAHDI before installing libPRI.

With Internet Access

```
[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
```

Without Internet Access

When installing on a system without internet access, there are a few additional steps that are required to build DAHDI.
The firmware files for the various VPM modules will need to be downloaded and extracted in the source directory. The file specific links provided below are the current versions as of this writing. Please check the link below for the full list of versions.

http://downloads.digium.com/pub/telephony/firmware/releases/

On a system with internet access, download the following files:

```
wget
http://downloads.digium.com/pub/telephony/firmware/releases/dahdi-fw-hx8-2
.06.tar.gz

wget
http://downloads.digium.com/pub/telephony/firmware/releases/dahdi-fw-oct61
14-064-1.05.01.tar.gz

wget
http://downloads.digium.com/pub/telephony/firmware/releases/dahdi-fw-oct61
14-128-1.05.01.tar.gz

wget
http://downloads.digium.com/pub/telephony/firmware/releases/dahdi-fw-vpmoc
t032-1.8.0.tar.gz

wget
http://downloads.digium.com/pub/telephony/firmware/releases/dahdi-fw-tc400
m-MR6.12.tar.gz

wget
http://downloads.digium.com/pub/telephony/firmware/releases/dahdi-fwload-v
pmadt032-1.25.0.tar.gz
```

Now send these to the Asterisk system and store them in

```
/usr/local/src/dahdi-linux-complete-2.X.Y+2.X.Y/linux/drivers/dahdi/firmwa
re/
```

Now we can continue the installation on the Asterisk system using the steps below.

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

[root@server dahdi-linux-complete-2.X.Y+2.X.Y]# cd
linux/drivers/dahdi/firmware

[root@server firmware]# for tarball in $(ls dahdi-fw-*.tar.gz); do tar -zxf
$tarball; done;

[root@server firmware]# cd -

[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

Before you can build libpri, you'll need to Building and Installing DAHDI

Having finished that, let's compile and install libpri. Again, 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

Checking Asterisk Requirements

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

```
[root@server dahdi-linux-complete-2.X.Y+2.X.Y]# cd
/usr/local/src/asterisk-1.8.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-1.8.X.Y]# ./configure
```

This will run for a couple of minutes, and warn you of any missing system libraries or other dependencies.

If you have missing dependencies then you should install them now and then run **configure** agai n to make sure they are recognized. A helpful way to install most of the dependencies you need is to use the **install_prereq** script included in the **contrib/scripts/** directory of your Asterisk source. It's quite straightforward to use, but may not work on all systems. Run the script with no arguments to see the usage help.

Upon 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: 
configure: OS type  : 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-1.8.X.Y]# make distclean
```

Using Menuselect to Select Asterisk Options

The next step in the build process is to tell Asterisk which modules[docs:1] 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-1.8.X.Y]# make menuselect
```

Terminal Window

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

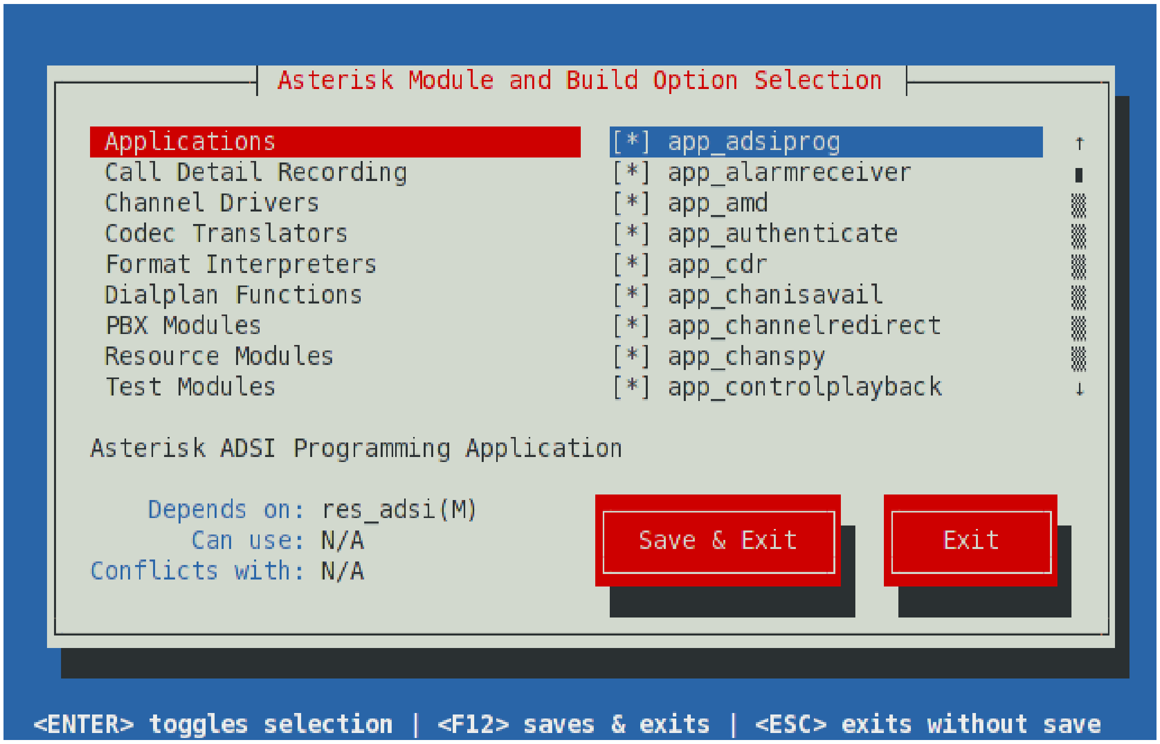
```
Terminal must be at least 80 x 21.
```

Asterisk 1.8+

```
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 [docs:**] 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.

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.



Easier Debugging of Asterisk Crashes

If you're finding that Asterisk is crashing on you, there's a setting in menuselect that will help provide additional information to the Asterisk developers. Go into menuselect, select the the Compiler Flags section (you'll need to scroll down in the left-hand list), and select the DONT_OPTIMIZE setting. Then rebuild Asterisk as shown below. While the Asterisk application will be slightly larger, it will provide additional debugging symbols in the event of a crash.

We should also inform people that the sound prompts are selected in menuselect as well

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.

Building and Installing Asterisk

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

```
[root@server asterisk-1.8.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 ---------+
```

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-1.8.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.

If you installed Asterisk from a tarball (as shown above), the security information is located in a PDF file named asterisk.pdfin the tex/ sub-directory of the source code. If that file doesn't exist, please install the rubber application on your system, and then type:

```
[root@server asterisk-1.8.X.Y]# make pdf
```

Installing Sample Files

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

```
[root@server asterisk-1.8.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, and can be used to stop or restart Asterisk as well. To install the initscript, use the **make config** command.

```
[root@server asterisk-1.8.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-1.8.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-1.8.X.Y]# lsmod | grep dahdi
```

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

```
[root@server asterisk-1.8.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-1.8.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-1.8.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-1.8.X.Y]# lsmod | grep dahdi
dahdi_dummy      4288 0
dahdi_transcode  7928 1 wctc4xxp
dahdi_voicebus  40464 2 wctdm24xxp,wcte12xp
dahdi          196544 12
dahdi_dummy,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-1.8.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-1.8.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-1.8.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.

Getting Started with Asterisk

In this section, we'll show you how to get started with Asterisk, and how to get around on the Asterisk command-line interface (commonly abbreviated as CLI). We'll also show you how to troubleshoot common problems that you might encounter when first learning Asterisk

Connecting to the CLI

First, let's show you how to connect to the Asterisk command-line interface. As you should recall from the installation, Asterisk typically runs in the background as a service or daemon. If the Asterisk service is already running, type the command below to connect to its command-line interface.

```
[root@server ~]# asterisk -r
```

The -r parameter tells the system that you want to re-connect to the Asterisk service. If the reconnection is successful, you'll see something like this:

```
[root@server ~]# asterisk -r
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.
=========================================================================
Connected to Asterisk version currently running on server (pid = 11187)
server*CLI>
```

Notice the ***CLI>** text? That's your Asterisk command-line prompt. All of the Asterisk CLI commands take the form of **module action parameters....** For example, type **core show uptime** to see how long Asterisk has been running.

```
server*CLI> core show uptime
System uptime: 1 hour, 34 minutes, 17 seconds
Last reload: 1 hour, 34 minutes, 17 seconds
```

You can use the built-in help to get more information about the various commands. Simply type **core show help** at the Asterisk prompt for a full list of commands, or **core show help command** for help on a particular command.

If you'd like to exit the Asterisk console and return to your shell, just use the **quit** command from the CLI. Such as:

```
server*CLI> quit
```

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"
```

Stopping and Restarting Asterisk

There are four 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.

Changing the Verbose and Debug Levels

Asterisk has two different classes of messages that appear in the command-line interface. The first class is called **verbose** messages. Verbose messages give information about the calls on the system, as well as notices, warnings, and errors. Verbose messages are intended for Asterisk administrators to be able to better manage their systems.

Asterisk allows you to control the verbosity level of the command-line interface. At a verbosity level of zero, you'll receive minimal information about calls on your system. As you increase the verbosity level, you'll see more and more information about the calls. For example, if you set the verbosity level to three or higher, you'll see each step a call takes as it makes its way through the dialplan. There are very few messages that only appear at verbosity levels higher than three.

To change the verbosity level, use the CLI command **core set verbose**, as shown below:

```
server*CLI> core set verbose 3
Verbosity was 0 and is now 3
```

You can also increase (but not decrease) the verbosity level when you connect to the Asterisk CLI from the Linux prompt, by using one or more **-v** parameters to the **asterisk** application. For example, this would connect to the Asterisk CLI and set the verbosity to three (if it wasn't already three or higher), because we added three -v parameters:

```
[root@server ~]# asterisk -vvvr
```

The second class of system messages is known as **debug** messages. These messages are intended for Asterisk developers, to give information about what's happening in the Asterisk program itself. They're often used by developers when trying to track down problems in the code, or to understand why Asterisk is behaving in a certain manner.

To change the debugging level, use the CLI command **core set debug**, as shown below:

```
server*CLI> core set debug 4
Core debug was 0 and is now 4
```

You can also increase (but not decrease) the debugging level when you connect to the Asterisk CLI from the Linux prompt. Simply add one or more **-d** parameters to the **asterisk** application.

```
[root@server ~]# asterisk \-ddddr
```

> **Verbose and Debug Levels**
> Please note that the verbose and debug levels are global settings, and apply to all of Asterisk, not just your command-line interface.
>
> We recommend that you set your verbosity level to three while learning Asterisk, so that you can get a feel for what is happening as calls are processed. On a busy production system, however, you'll want to set the verbosity level lower. We also recommend that you use debug messages sparingly, as they tend to be quite verbose and can affect call volume on busy systems.

Simple CLI Tricks

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
```

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.

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>
```

As you can see, there's a wealth of information available from the Asterisk command-line interface, and we've only scratched the surface. In later sections, we'll go into more details about how to use the command-line interface for other purposes.

Troubleshooting

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** a pplication. 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.

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. Each module that you loads 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 various types of modules and the capabilities they provide.

Types of Asterisk Modules

There are many different types of modules, many of which are shown in the diagram above.

- **Channel Drivers**

At the top of the diagram, we show 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, and so forth.

- **Dialplan Functions**

Functions 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.

- **Resources**

As the name suggests, resources provide resources to Asterisk. 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.

- **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.

Now let's go into more detail on each of the module types.

Channel Driver Modules

All calls from the outside come 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 SIP channel driver, 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 PBX, it's a multi-protocol PBX.

For more information on the various channel drivers, see Section 400. Channel Drivers and External Connectivity.

All channel drivers have a file name that look like **chan_xxxxx.so**, such as **chan_sip.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 Dialplan Fundamentals.

All application modules have file names that looks like **app_xxxxx.so**, such as **app_voicemail.so**.

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 application modules have file names that looks like **func_xxxxx.so**, such as **func_callerid.so**.

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 of 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**.

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.

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

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

- ADPCM, 32kbit/s
- G.711 alaw, 64kbit/s
- G.711 ulaw, 64kbit/s
- G.722, 64kbit/s
- G.726, 32kbit/s
- GSM, 13kbit/s
- LPC-10, 2.4kbit/s

If the Speex (www.speex.org) development libraries are detected on your system when Asterisk is built, a CODEC module for Speex will also be installed.

If the iLBC (www.ilbcfreeware.org) development libraries are detected on your system when Asterisk is built, a CODEC module for 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 both software and hardware 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.

For more detailed information on CODECs, see CODECs.

File Format Drivers

Add a list of the file formats that Asterisk supports, then point them at the module in section 400 that goes into more detail?
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** an d **format_jpeg.so**.

Add a list of the file formats that Asterisk supports, then point them at the module in section 400 that goes into more detail?

Call Detail Record (CDR) Drivers

CDR modules are used to store call detail records in a variety of formats. Popular storage mechanisms include comma-separated value (CSV) files, as well as relational databases such as 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.

For more information on call detail records, see Section 370. Call Detail Records.

Call detail record modules have file names that look like **cdr_xxxxx.so**, such as **cdr_csv.so** and **cdr_pgsql.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** an d **bridge_multiplexed.so**.

Call Flow and Bridging Model

Now that you know about the various modules that Asterisk uses, let's talk about the ways that calls flow through an Asterisk system. To explain this clearly, let's say that Alice wants to talk to Bob, and they both have SIP phones connected to their Asterisk system. Let's see what happens!

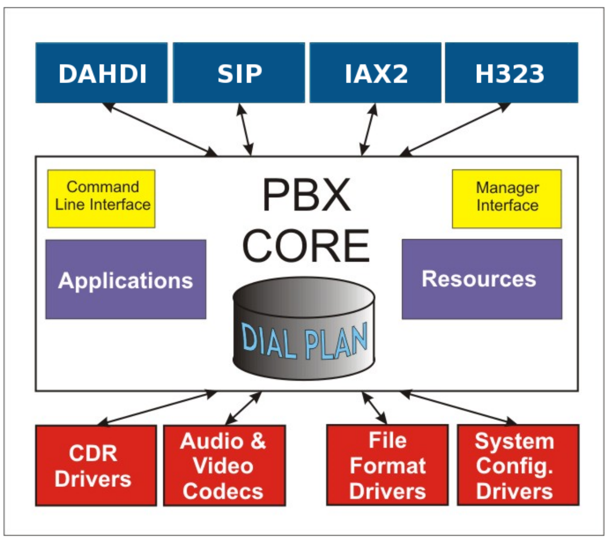
Should we add a graphic to help explain the call flow model?

1. Alice dials extension 6002, which is Bob's extension on the Asterisk system.
2. A SIP message goes from Alice's phone to the SIP channel driver in Asterisk
3. The SIP channel driver authenticates the call. If Alice's phone does not provide the proper credentials, Asterisk rejects the call.
4. At this point, we have Alice's phone communicating with Asterisk.
5. Now the call goes from the SIP channel driver into the core of Asterisk. Asterisk looks for a set of instructions to follow for extension 6002 in the dialplan.
6. Extension 6002 in the dialplan tells Asterisk to call Bob's phone
7. Asterisk makes a call out through the SIP channel driver to Bob's phone.
8. Bob answers his phone.
9. Now we have two independent calls on the Asterisk system: one from Alice, and to Bob. Asterisk now bridges the audio between these two calls (known as **channels** in Asterisk parlance).
10. When one channel hangs up, Asterisk signals the other channel to hang up.

And there we have it! We've shown how calls flow from external devices, through the channel drivers to the core of Asterisk, and back out through the channel drivers to external devices.

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 Architecture

We need to add CEL and Bridge modules to this picture, and take CLI and Manager out for now

The heart of any Asterisk system is the **core**. The PBX core is the essential component that takes care of bridging calls. The core also takes care of other items like reading the configuration files and loading the other modules. We'll talk more about the core below, but for now just remember that all the other modules connect to it.
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.

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 typical Asterisk system has over one hundred fifty different modules!

The core also contains 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.

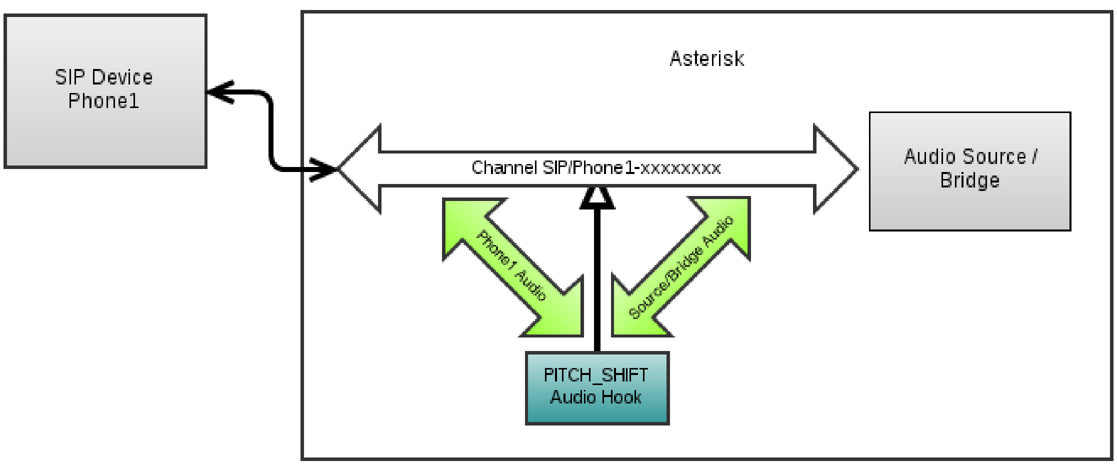
Asterisk modules which are part of the core have a file name that look like **pbx_xxxxx.so**.

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:

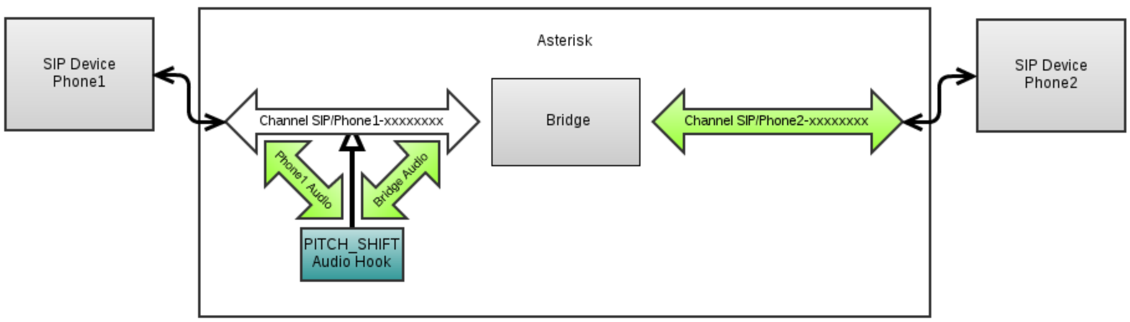
```
exten => 1,1,Answer()
exten => 1,n,Set(PITCH_SHIFT(both)=higher)
exten => 1,n,Voicemail(501)
```

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.

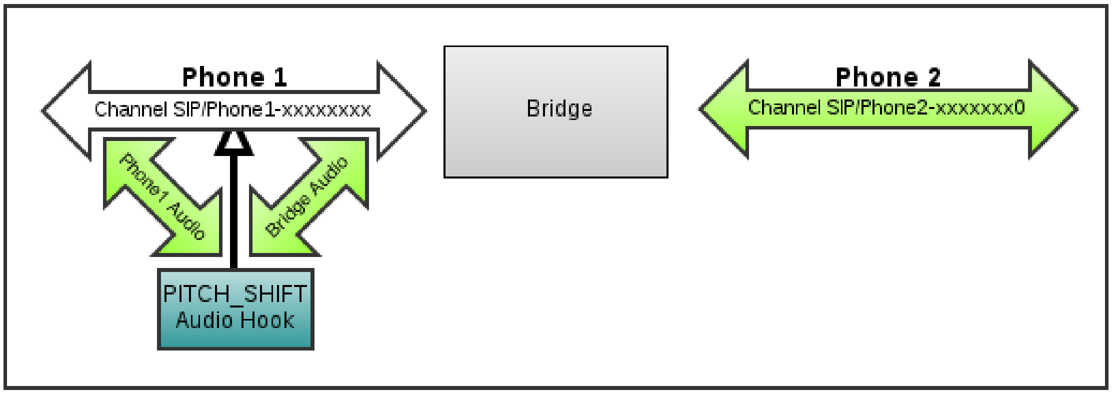
Using AUDIOHOOK_INHERT only requires that AUDIOHOOK_INHERIT(source)=yes is set where source is the name given for the source of the audiohook. For more information on the sources available, see the description of the source argument in the documentation for AUDIOHOOK_INHERIT.

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

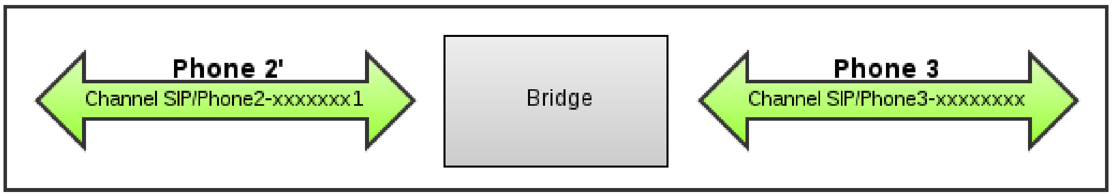
```
exten => 1,1,Answer()
exten => 1,n,MixMonitor(training_recording.wav)
exten => 1,n,Set(AUDIOHOOK_INHERIT(MixMonitor)=yes)
exten => 1,n,Queue(techsupport)
```

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

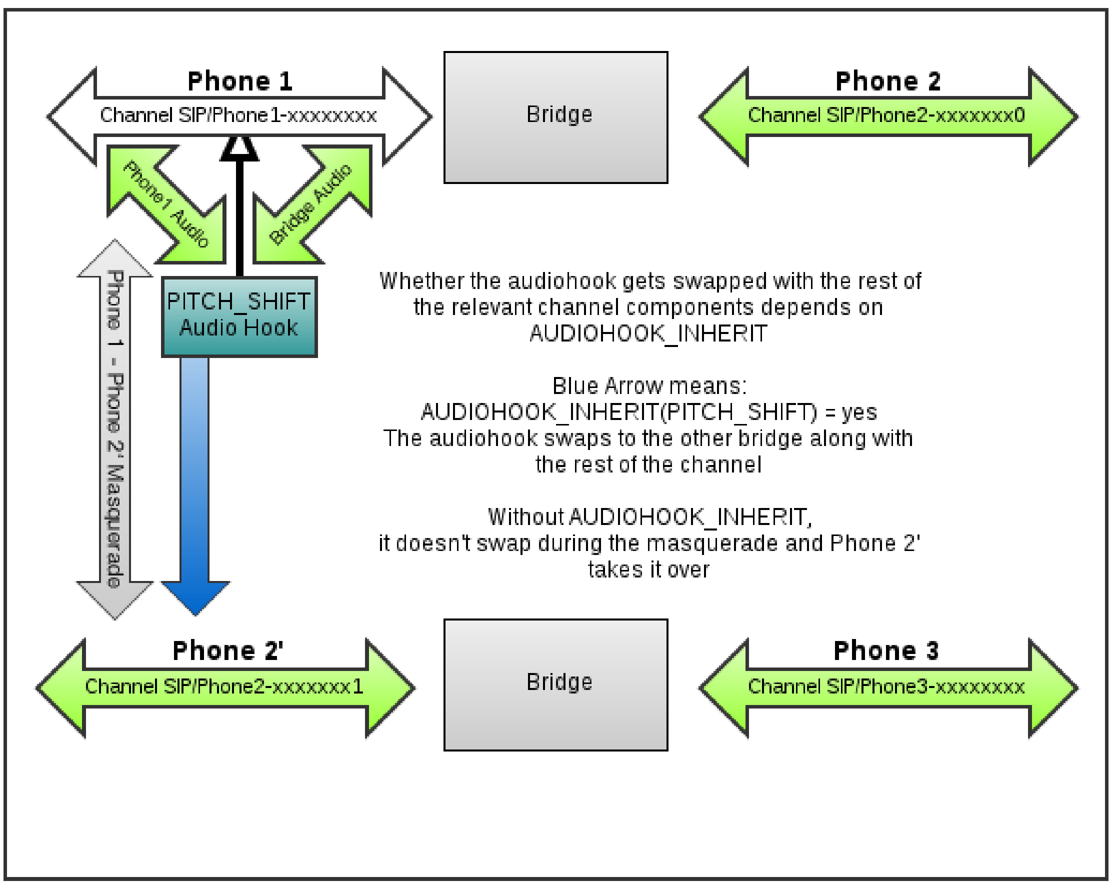
Initial Call Setup



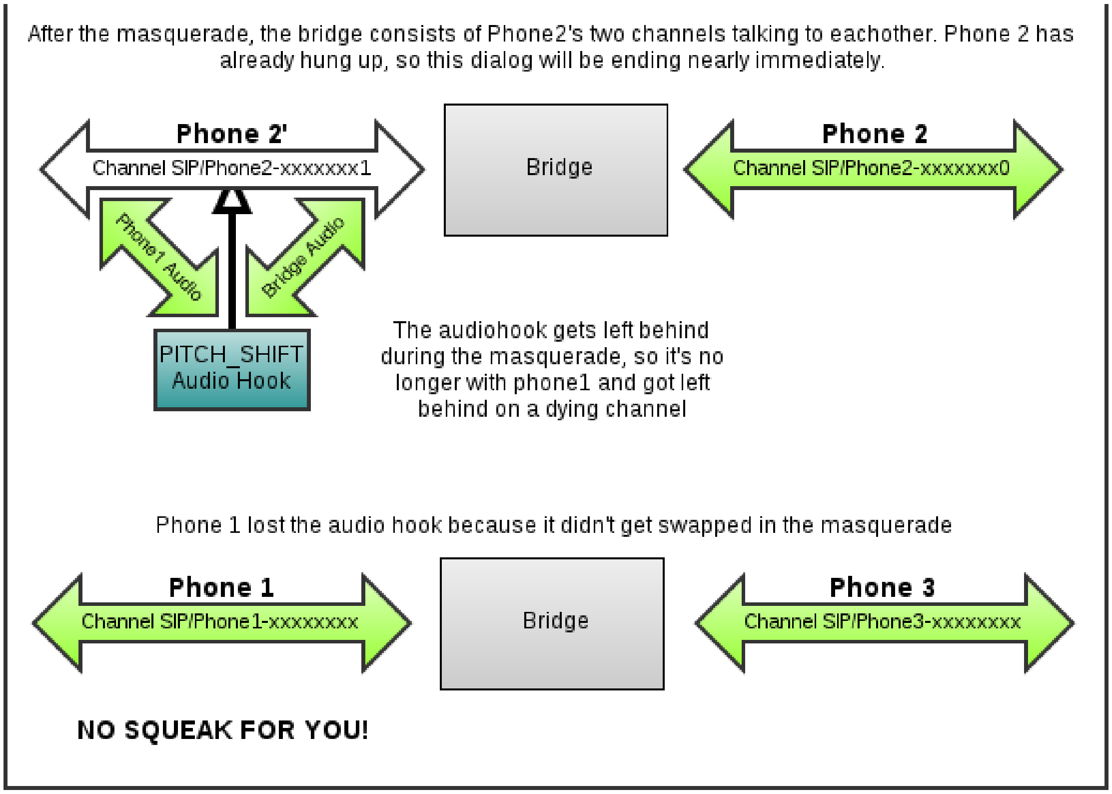
SIP/Phone2 attempts starts to attended transfer SIP/Phone1 to SIP/Phone3



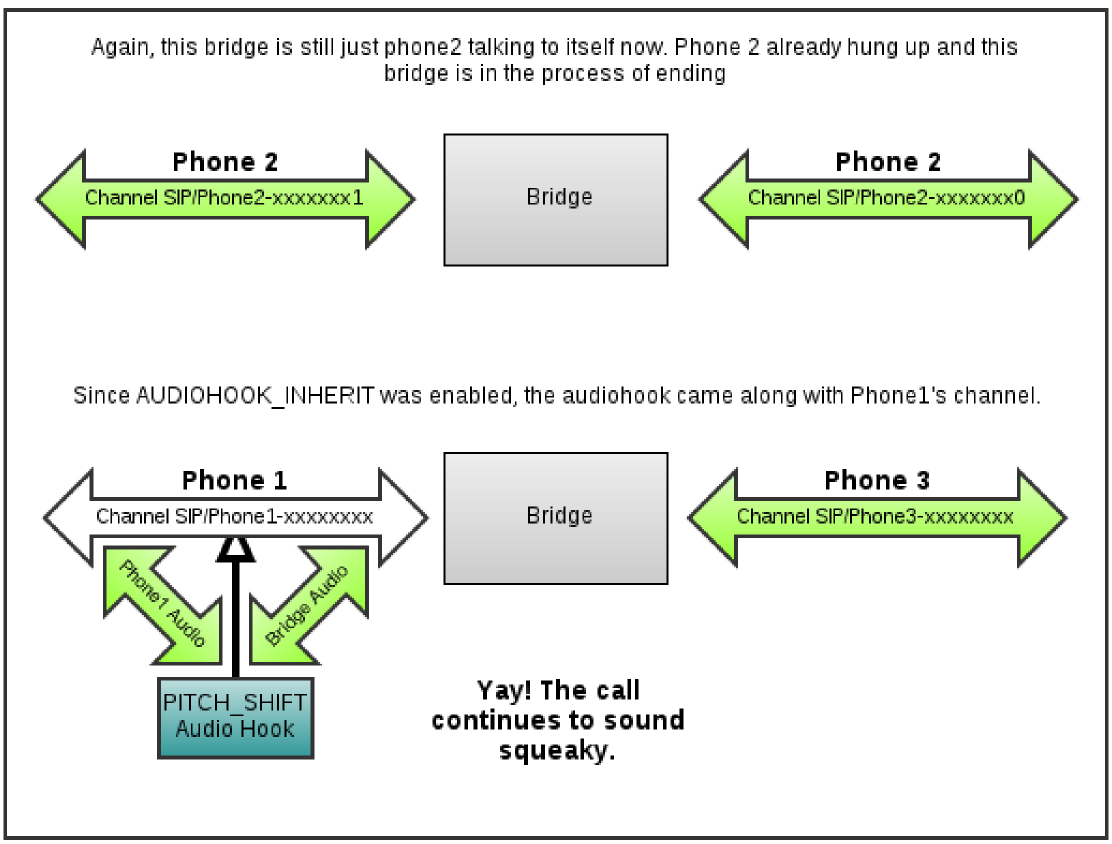
Phone 2 hangs up on Phone 3, initiating the transfer. This requires a masquerade.



Bridges after Transfer: Without AUDIOHOOK_INHERIT(PITCH_SHIFT)



Bridges after Transfer: With AUDIOHOOK_INHERIT(PITCH_SHIFT)



Inheritance of audiohooks can be turned off in the same way by setting

AUDIOHOOK_INHERIT(source)=no.

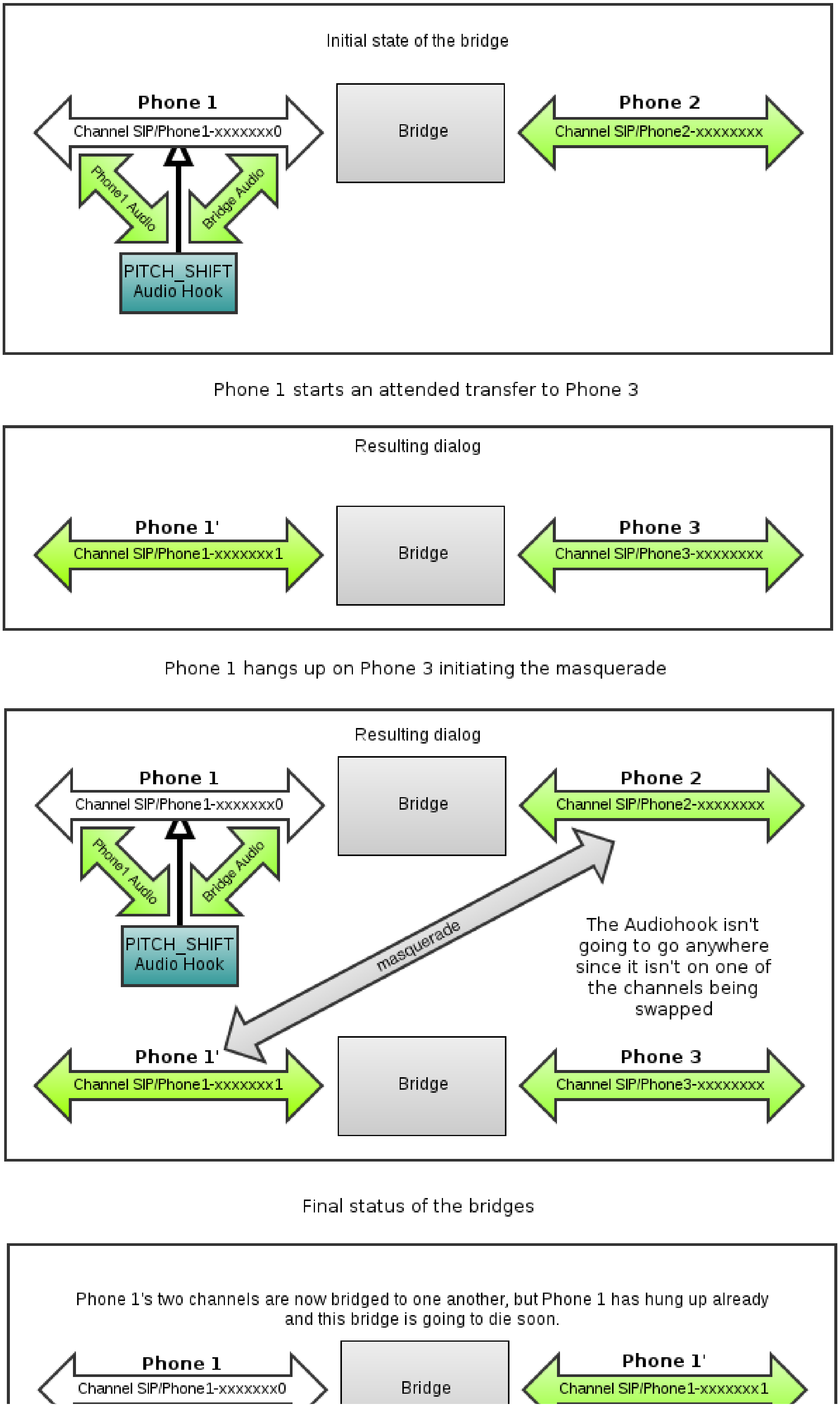
Audiohook Sources

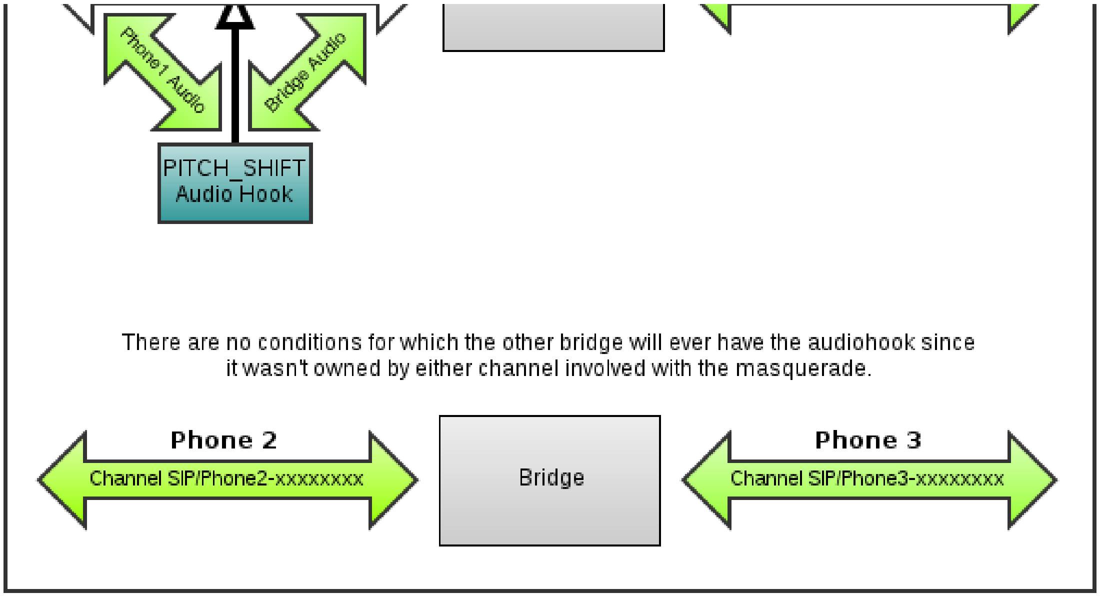
Audiohooks have a source name and can come from a number of sources. An up to date list of possible sources should always be available from the documentation for AUDIOHOOK_INHERIT.

- Chanspy - from app_chanspy
- MixMonitor - app_mixmonitor.c
- Volume - func_volume.c
- Mute - res_mutestream.c
- Speex - func_speex.c
- pitch_shift - func_pitchshift.c
- JACK_HOOK - app_jack.c

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
The Audiohook isn't going to go anywhere since it isn't on one of the channels being swapped
Phone 1'
Channel SIP/Phone1-xxxxxxx1
Bridge
Phone 3
Channel SIP/Phone3-xxxxxxxx
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



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.

Configuration and Operation

Here is the top-level page for all of the Asterisk Reference Information, formerly found in the doc/ and doc/tex subdirectories of the Asterisk source.

It's been there all along, but now it's here, in an easy to view format (no need to install 800MB of dependancies in Debian just to convert .tex into PDF), that's also searchable. Hoo-ray!

Asterisk Calendaring

The Asterisk Calendaring API 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 three calendaring modules that ship with Asterisk that provide support for iCalendar, CalDAV, and Microsoft Exchange Server calendars. All three modules support event notification. Both CalDAV and Exchange support reading and writing calendars, while iCalendar is a read-only format.

Configuring Asterisk Calendaring

All asterisk calendaring modules are configured through calender.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.

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 adminstrator 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 overrided 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 Channel Drivers

All about Asterisk and its Channel Drivers

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.

IAX2 Jitterbuffer

The new jitterbuffer

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

PLC

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

This facility is enabled by default in iLBC and speex, as it has no additional cost. This facility can be enabled in adpcm, alaw, g726, gsm, lpc10, and ulaw by setting genericplc = true in the plc se ction 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.

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.

Local Channel

Introduction to Local Channels

In Asterisk, Local channels are a method used to treat an extension in the dialplan as if it were an external device. In essense, Asterisk will send the call back into the dialplan as the destination of the call, versus sending the call to a device.

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

Local Channel Examples

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

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.

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.

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.

Understanding when to use (slash)n

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

This following dialplan will not perform as expected.

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

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

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

This can have some adverse effects when you're expecting information to be available during the call that gets associated with the Local channel. When the Local channel is optimized out of the call path, any Dial() flags, or channel variables associated with the Local channel are also destroyed and are no longer available to Asterisk.

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

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

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

...into the following:

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

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

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

1. SIP device PHONE_A calls Asterisk via a SIP INVITE
2. Asterisk accepts the INVITE and then starts processing dialplan logic in the [internal] context
3. Our dialplan calls Dial(Local/2@services) - notice no /n
4. The Local channel then executes dialplan at extension 2 within the [services] context
5. Extension 2 within [services] then performs Dial() to PHONE_B with the line: Dial(SIP/PHONE_B,,L(60000:45000:15000))
6. SIP/PHONE_B then answers the call
7. Even though the L option was given when dialing the SIP device, the L information is stored in the channel that is doing the Dial() which is the Local channel, and not the endpoint SIP channel.
8. The Local channel in the middle, containing the information for tracking the time allowance of the call, is then optimized out of the call path, losing all information about when to terminate the call.
9. SIP/PHONE_A and SIP/PHONE_B then continue talking indefinitely.

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

1. Use what we just described, Dial(Local/2@services/n) to cause the Local channel to remain in the call path so that the L() option used inside the Local channel is not discarded when optimization is performed.
2. Place the L() option at the outermost part of the path so that when the middle is optimized out of the call path, the information required to make L() work is associated with the outside channel. The L information will then be stored on the calling channel, which is PHONE_A. For example:

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

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

Local Channel Modifiers

There are additional modifiers for the Local channel as well. They include:

- 'n' - Adding "/n" at the end of the string will make the Local channel not do a native transfer (the "n" stands for "n"o release) upon the remote end answering the line. This is an esoteric, but important feature if you expect the Local channel to handle calls exactly like a normal channel. If you do not have the "no release" feature set, then as soon as the destination (inside of the Local channel) answers the line and one audio frame passes, the variables and dial plan will revert back to that of the original call, and the Local channel will become a zombie and be removed from the active channels list. This is desirable in some circumstances, but can result in unexpected dialplan behavior if you are doing fancy things with variables in your call handling.

- 'j' - Adding "/j" at the end of the string allows you to use the generic jitterbuffer on incoming calls going to Asterisk applications. For

example, this would allow you to use a jitterbuffer for an incoming SIP call to Voicemail by putting a Local channel in the middle. The 'j' option must be used in conjunction with the 'n' option to make sure that the Local channel does not get optimized out of the call.
This option is available starting in the Asterisk 1.6.0 branch.

- 'm' - Using the "/m" option will cause the Local channel to forward music on hold (MoH) start and stop requests. Normally the Local channel acts on them and it is started or stopped on the Local channel itself. This options allows those requests to be forwarded through the Local channel.
  This option is available starting in the Asterisk 1.4 branch.
- 'b' - The "/b" option causes the Local channel to return the actual channel that is behind it when queried. This is useful for transfer scenarios as the actual channel will be transferred, not the Local channel.

This option is available starting in the Asterisk 1.6.0 branch.

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, Threeway call, 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.

Asterisk Configuration

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

General Configuration Information

The top-level page for general (typical) Asterisk configuration information.

Configuration Parser

Introduction

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

Asterisk configuration files are defined as follows:

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

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

```
[section]
label1 = value1
label2 = value2
object => name
label3 = value3
label2 = value4
object2 => name2
```

In this syntax, we create objects with the settings defined above the object creation. Note that settings are inherited from the top, so in the example above object2 has inherited the setting for "label1" from the first object.
For template configurations, the syntax for defining a section is changed to:

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

The options field is used to define templates, refer to templates and hide templates. Any object can be used as a template.
No whitespace is allowed between the closing "]" and the parenthesis "(".
Comments

All lines that starts with semi-colon ";" is treated as comments and is not parsed.
The ";~~" is a marker for a multi-line comment. Everything after that marker will be treated as a comment until the end-marker "~~;" is found. Parsing begins directly after the end-marker.

```
;This is a comment
label = value
;-- This is
a comment -;
;- Comment --; exten=> 1000,1,dial(SIP/lisa)
```

Including other files

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

```
#include myusers.conf
```

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

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

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

The exec directive is used like this:

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

Adding to an existing section

```
[section]
label = value

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

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

Defining a template-only section

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

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

Using templates (or other configuration sections)

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

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

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

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

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

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

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

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

Additional Examples
(in top-level sip.conf)

```
[defaults]
type=friend
nat=yes
qualify=on
dtmfmode=rfc2833
disallow=all
allow=alaw
#include accounts/*/sip.conf
```

(in accounts/customer1/sip.conf)

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

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

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

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

The asterisk.conf file

Asterisk Main Configuration File

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

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

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

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

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

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

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

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

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

[options]
;Under "options" you can enter configuration options
;that you also can set with command line options
; Verbosity level for logging (-v) verbose = 0
; Debug: "No" or value (1-4)
debug = 3

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

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

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

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

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

; Disable ANSI colors (-n)
```

```
nocolor = yes | no

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

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

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

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

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

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

; 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
```

CLI Prompt

Changing the CLI Prompt

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

You may include the following variables, that will be replaced by the current value by Asterisk:

- %d - Date (year-month-date)
- %s - Asterisk system name (from asterisk.conf)
- %h - Full hostname
- %H - Short hostname

- %t - Time
- %u - Username
- %g - Groupname
- %% - Percent sign

- %# - '#' if Asterisk is run in console mode, '' if running as remote console
- %Cn[;n] - Change terminal foreground (and optional background) color to specified A full list of colors may be found in include/asterisk/term.h

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

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

The Asterisk Dialplan

The Asterisk dialplan

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

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

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

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

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

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

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

There are some extensions with important meanings:

- s - What to do when an extension context is entered (unless overridden by the low level channel interface) This is used in macros, and some special cases. "s" is not a generic catch-all wildcard extension.
- i - What to do if an invalid extension is entered
- h - The hangup extension, executed at hangup
- t - What to do if nothing is entered in the requisite amount of time.
- T - This is the extension that is executed when the 'absolute' timeout is reached. See "core show function TIMEOUT" for more information on setting timeouts.
- e - This extension will substitute as a catchall for any of the 'i', 't', or 'T' extensions, if any of them do not exist and catching the error in a single routine is desired. The function EXCEPTION may be used to query the type of exception or the location where it occurred.

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

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.

Table 2.1: Channel Driver QoS Settings

	Signaling	Audio	Video	Text

chan_sip	+	+	+	+
chan_skinny	+	+	+	
chan_mgcp	+	+		
chan_unistm	+	+		
chan_h323		+		
chan_iax2	+			
chan_pjsip	+	+	+	

Table 2.2: Other ToS Settings

	Signaling	Audio	Video	Text
dundi.conf	+ (tos setting)			
iaxprov.conf	+ (tos setting)			

IP TOS values

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

The tos parameters also take numeric values.*

Note that on 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.

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:

Table 2.3: Recommended QoS Settings

	tos	cos
Signaling	cs3	3
Audio	ef	5

Video	af41	4
Text	af41	3
Other	ef	

IAX2

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

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

In sip.conf, there are 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.
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

MP3 Support

MP3 Music On Hold

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

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

ICES

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

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

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.

FreeTDS

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

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

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_sc
reen','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;
```

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.

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 (14:00-25:00,sat-sun,,)
            Voicemail(${EXTEN},b);
        else
        {
            Voicemail(${EXTEN},u);
            NoOp(hi, there!);
        }
        random(51) NoOp(This should appear 51% of the time);
        random( 60 )
        {
                NoOp( This should appear 60% of the time );
        }
        else
        {
                random(75)
                {
                    NoOp( This should appear 30% of the time! );
                }
                else
                {
                    NoOp( This should appear 10% of the time! );
                }
        }
    }
    _777X => {
        switch (${EXTEN}) {
            case 7771:
                NoOp(You called 7771!);
                break;
            case 7772:
                NoOp(You called 7772!);
                break;
            case 7773:
                NoOp(You called 7773!);
                // fall thru-
            pattern 777[4-9]:
                NoOp(You called 777 something!);
            default: NoOp(In the default clause!);
        }
    }
}
```

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(,);
}
```

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.

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 command" 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
```

Device Status Reports with AMI

blank

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 Manger (AJAM)

AJAM is a new technology which allows web browsers or other HTTP enabled applications and web pages to directly access the Asterisk Manger Interface (AMI) via HTTP. Setting up your server to process AJAM involves a few steps:

Setting up the Asterisk HTTP server

1. Uncomment the line "enabled=yes" in /etc/asterisk/http.conf to enable Asterisk's builtin micro HTTP server.
2. If you want Asterisk to actually deliver simple HTML pages, CSS, javascript, etc. you should uncomment "enablestatic=yes"
3. Adjust your "bindaddr" and "bindport" settings as appropriate for your desired accessibility
4. Adjust your "prefix" if appropriate, which must be the beginning of any URI on the server to match. The default is "asterisk" and the rest of these instructions assume that value.

Allow Manager Access via HTTP

1. Make sure you have both "enabled = yes" and "webenabled = yes" setup in /etc/asterisk/manager.conf
2. You may also use "httptimeout" to set a default timeout for HTTP connections.
3. Make sure you have a manager username/secret

Once those configurations are complete you can reload or restart Asterisk and you should be able to point your web browser to specific URI's which will allow you to access various web functions. A complete list can be found by typing "http show status" at the Asterisk CLI.
examples:

- http://localhost:8088/asterisk/manager?action=login&username=foo&secret=bar

This logs you into the manager interface's "HTML" view. Once you're logged in, Asterisk stores a cookie on your browser (valid for the length of httptimeout) which is used to connect to the same session.

- http://localhost:8088/asterisk/rawman?action=status

Assuming you've already logged into manager, this URI will give you a "raw" manager output for the "status" command.

- http://localhost:8088/asterisk/mxml?action=status

This will give you the same status view but represented as AJAX data, theoretically compatible with RICO (http://www.openrico.org).

- http://localhost:8088/asterisk/static/ajamdemo.html

If you have enabled static content support and have done a make install, Asterisk will serve up a demo page which presents a live, but very basic, "astman" like interface. You can login with your username/secret for manager and have a basic view of channels as well as transfer and hangup calls. It's only tested in Firefox, but could probably be made to run in other browsers as well.

A sample library (astman.js) is included to help ease the creation of manager HTML interfaces.

⚠ 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 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.

Configuring Call Queues

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/PleaseHoldAndYourCallWillBeAnsweredByOurNextAvailableSalesRepresenta
tive);
        WaitExten(0.3);

Background(digium/AtAnyTimeYouMayPress0ToSpeakWithAnOperatorOr8ToLeaveAMessage);
        Set(CALLERID(name)=Sales);
        Queue(sales-general,t);
        Set(CALLERID(name)=EmptySalQ);
        goto dispatch,s,1;
        Playback(goodbye);
        Hangup();
    }
}
```

Please note that there is only one attempt to queue a call in the sales queue. All sales agents that are logged in will be rung.

```
context customerservice {
    0 => {
        SetCIDName(CSVTrans);
        goto dispatch|s|1;
    }
    8 => Voicemail(${CUSTSERVVM});
    s => {
        Ringing();
        Wait(2);
        Background(digium/ThankYouForCallingDigiumCustomerService);
        WaitExten(0.3);
        notracking:
Background(digium/PleaseWaitForTheNextAvailableCustomerServiceRepresentative);
        WaitExten(0.3);

Background(digium/AtAnyTimeYouMayPress0ToSpeakWithAnOperatorOr8ToLeaveAMessage);
        Set(CALLERID(name)=Cust Svc);
        Set(QUEUE_MAX_PENALTY=10);
        Queue(customerservice,t);
        Set(QUEUE_MAX_PENALTY=0);
        Queue(customerservice,t);
        Set(CALLERID(name)=EmptyCSVQ);
        goto dispatch,s,1;
        Background(digium/NoCustomerServiceRepresentativesAreAvailableAtThisTime);
        Background(digium/PleaseLeaveAMessageInTheCustomerServiceVoiceMailBox);
        Voicemail(${CUSTSERVVM});
        Playback(goodbye);
        Hangup();
    }
}
```

Note that calls coming into customerservice will first be try to queue calls to those agents with a QUEUE_MAX_PENALTY of 10, and if none are available, then all agents are rung.

```
context dispatch {
    s => {
        Ringing();
        Wait(2);
        Background(digium/ThankYouForCallingDigium);
        WaitExten(0.3);
        Background(digium/YourCallWillBeAnsweredByOurNextAvailableOperator);
        Background(digium/PleaseHold);
        Set(QUEUE_MAX_PENALTY=10);
        Queue(dispatch|t);
        Set(QUEUE_MAX_PENALTY=20);
        Queue(dispatch|t);
        Set(QUEUE_MAX_PENALTY=0);
        Queue(dispatch|t);
        Background(digium/NoOneIsAvailableToTakeYourCall);
        Background(digium/PleaseLeaveAMessageInOurGeneralVoiceMailBox);
        Voicemail(${DISPATCHVM});
        Playback(goodbye);
        Hangup();
    }
}
```

And in the dispatch context, first agents of priority 10 are tried, then 20, and if none are available, all agents are tried.

Notice that a common pattern is followed in each of the three queue contexts:

First, you set QUEUE_MAX_PENALTY to a value, then you call Queue(queue-name,option,...) (see the Queue application documetation for details)

In the above, note that the "t" option is specified, and this allows the agent picking up the incoming call the luxury of transferring the call to other parties.

The purpose of specifying the QUEUE_MAX_PENALTY is to develop a set of priorities amongst agents. By the above usage, agents with lower number priorities will be given the calls first, and then, if no-one picks up the call, the QUEUE_MAX_PENALTY will be incremented, and the queue tried
again. Hopefully, along the line, someone will pick up the call, and the Queue application will end with a hangup.

The final attempt to queue in most of our examples sets the QUEUE_MAX_PENALTY to zero, which means to try all available agents.

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.

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)
- 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!
- 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.

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 events, one module that is a consumer of these events has been implemented.

Asterisk trunk was recently updated to include support for dynamic logger levels. This module takes advantage of this functionality to create a custom "security" logger level. Then, when this module is in use, logger.conf can be configured to put security events into a file

security_log => security

The content of this file is a well defined and easily interpretable format for external scripts to read and act upon. The definition for the format of the log file is described later in this chapter.

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, ChallengeResponseFailed, InvalidPassword
- 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, 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.

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 Subversion via the following command:

```
# svn co http://svn.asterisk.org/svn/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

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 notifica- tion 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.

Call Detail Records (CDR)

Top-level page for all things CDR

CDR Applications

- SetAccount - Set account code for billing
- SetAMAFlags - Sets AMA flags
- NoCDR - Make sure no CDR is saved for a specific call
- ResetCDR - Reset CDR
- ForkCDR - Save current CDR and start a new CDR for this call
- Authenticate - Authenticates and sets the account code
- SetCDRUserField - Set CDR user field
- AppendCDRUserField - Append data to CDR User field

For more information, use the "core show application application" command. You can set default account codes and AMA flags for devices in channel configuration files, like sip.conf, iax.conf etc.

CDR Fields

- accountcode: What account number to use, (string, 20 characters)
- src: Caller*ID number (string, 80 characters)
- dst: Destination extension (string, 80 characters)
- dcontext: Destination context (string, 80 characters)
- clid: Caller*ID with text (80 characters)
- channel: Channel used (80 characters)
- dstchannel: Destination channel if appropriate (80 characters)
- lastapp: Last application if appropriate (80 characters)
- lastdata: Last application data (arguments) (80 characters)
- start: Start of call (date/time)
- answer: Answer of call (date/time)
- end: End of call (date/time)
- duration: Total time in system, in seconds (integer), from dial to hangup
- billsec: Total time call is up, in seconds (integer), from answer to hangup
- disposition: What happened to the call: ANSWERED, NO ANSWER, BUSY
- amaflags: What flags to use: DOCUMENTATION, BILL, IGNORE etc, specified on a per channel basis like accountcode.
- user field: A user-defined field, maximum 255 characters

In some cases, uniqueid is appended:

- uniqueid: Unique Channel Identifier (32 characters) This needs to be enabled in the source code at compile time

> ⚠ If you use IAX2 channels for your calls, and allow 'full' transfers (not media-only transfers), then when the calls is transferred the server in the middle will no longer be involved in the signaling path, and thus will not generate accurate CDRs for that call. If you can, use media-only transfers with IAX2 to avoid this problem, or turn off transfers completely (although this can result in a media latency increase since the media packets have to traverse the middle server(s) in the call).

In 1.8 and later

In some CDR backends, the following fields may also be supported:

- linkedid: a unique identifier based on uniqueid. Unlike uniqueid, but spreads to other channels as transfers, dials, etc are performed
- peeraccount: the account code of the bridged channel
- sequence: a field that can be combined with uniqueid and linkedid to uniquely identify a 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

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.

Channel Event Logging (CEL)

Top-level page for all things CEL

CEL Design Goals

CEL, or Channel Event Logging, has been 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, conferenced, forwarded, etc. In general, those doing billing applications in 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.

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.

The design philosophy of CEL is to generate event data that can grouped together to form a billing record. This may not be a simple task, but we hope to provide a few different examples that could be used as a basis for those involved in this effort.

There are definite parallels between Manager 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.

To 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).

This sequence of actions will generate the following annotated list of 42 CEL events:

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","","","DOCUME
NTATION","","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","","","DOCUME
NTATION","","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","(Ou
tgoing 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","","","DOC
UMENTATION","","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","","","DOC
UMENTATION","","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","","","DOCUME
NTATION","","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","ParkedC
all","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","ParkedC
all","701","DOCUMENTATION","","1178736428.5","","" ;;;
"EV_BRIDGE_START","2007-05-09
12:47:12","fxs.50","150","150","","","701","extension","Zap/50-1","ParkedC
all","701","DOCUMENTATION","","1178736428.5","","Zap/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","","","DOCUME
NTATION","","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","ParkedC
all","701","DOCUMENTATION","","1178736428.5","","Zap/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","ParkedC
all","701","DOCUMENTATION","","1178736428.5","","Zap/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","ParkedC
all","701","DOCUMENTATION","","1178736428.5","","Zap/51-1" ;;;
"EV_BRIDGE_START","2007-05-09
12:47:51","fxs.50","150","150","","","701","extension","Zap/50-1","ParkedC
all","701","DOCUMENTATION","","1178736428.5","","Zap/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","","","DOCUME
NTATION","","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","(Ou
tgoing 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","ParkedC
all","701","DOCUMENTATION","","1178736428.5","","Zap/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","ParkedC
all","701","DOCUMENTATION","","1178736428.5","","Zap/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","ParkedC
all","701","DOCUMENTATION","","1178736428.5","","Zap/51-1" ;;;
"EV_BRIDGE_START","2007-05-09
12:48:58","fxs.50","150","150","","","701","extension","Zap/50-1","ParkedC
all","701","DOCUMENTATION","","1178736428.5","","Zap/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","ParkedC
all","701","DOCUMENTATION","","1178736428.5","","Zap/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","","","DOC
UMENTATION","","1178736471.6","","" ;;; 150 hangs up
"EV_CHAN_END","2007-05-09
```

```
12:50:13","fxs.50","150","150","","","h","extension","Zap/50-1","","","DOC
UMENTATION","","1178736471.6","","" ;;; 150 ends
```

In terms of Manager events, the above Events correspond to the following 80 Manager events:

```
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 Events and Fields

While CDRs and the Manager are basically both event tracking mechanisms, CEL tries to track only those events that might pertain to billing issues.

Table of CEL Events

Event	Description
CHAN_START	The time a channel was created
CHAN_END	The time a channel was terminated
ANSWER	The time a channel was answered (ie, phone taken off-hook)
HANGUP	The time at which a hangup occurred
CONF_ENTER	The time a channel was connected into a conference room
CONF_EXIT	The time a channel was removed from a conference room
CONF_START	The time the first person enters a conference room
CONF_END	The time the last person left a conference room (and turned out the lights?)
APP_START	The time a tracked application was started
APP_END	the time a tracked application ended

PARK_START	The time a call was parked
PARK_END	Unpark event
BRIDGE_START	The time a bridge is started
BRIDGE_END	The time a bridge is ended
BRIDGE_UPDATE	This is a replacement channel (Masquerade)
3WAY_START	When a 3-way conference starts (usually via attended transfer)
3WAY_END	When one or all exit a 3-way conference
BLINDTRANSFER	When a blind transfer is initiated
ATTENDEDTRANSFER	When an attended transfer is initiated
TRANSFER	Generic transfer initiated; not used yet...?
PICKUP	This channel picked up the peer channel
FORWARD	This channel is being forwarded somewhere else
HOOKFLASH	So far, when a hookflash event occurs on a DAHDI interface
LINKEDID_END	The last channel with the given linkedid is retired
USER_DEFINED	Triggered from the dialplan, and has a name given by the user

Table of CEL Event Fields

Table 11.2: List of CEL Event Fields

Field	Description
eventtype	The name of the event; see the above list.
eventtime	The time the event happened
cid_name	CID name field
cid_num	CID number field
cid_ani	CID ANI field

cid_rdnis	CID RDNIS field
cid_dnid	CID DNID field
exten	The extension in the dialplan
context	The context in the dialplan
channame	The name assigned to the channel in which the event took place
appname	The name of the current application
appdata	The arguments that will be handed to that application
amaflags	The AMA flags associated with the event; user assignable.
accountcode	A user assigned datum (string)
peeraccount	A user assigned datum (string) on the peer.
uniqueid	Each Channel instance gets a unique ID associated with it.
linkedid	the per-call id, spans several events, possibly.
userfield	A user assigned datum (string)
peer	For bridge or other 2-channel events, this would be the other channel name
userdeftype	User defined event name
extra	Extra information associated with the event.

CEL Applications and Functions

Top-level page for information on CEL Applications and Functions

CEL Function

- THIS IS NO LONGER TRUE REWRITE *****

The CEL function parallels the CDR function, for fetching values from the channel or event. It has some notable notable differences, though! For instance, CEL data is not stored on the channel. Well, not much of it, anyway! You can use the CEL function to set the amaflags, accountcode, and userfield, which are stored on the channel.

Channel variables are not available for reading from the CEL function, nor can any variable name other than what's in the list, be set. CDRs have a structure attached to the channel, where the

CDR function could access the values stored there, or set the values there. CDRs could store their own variable lists, but CEL has no such storage. There is no reason to store any event information, as they are immediately output to the various backends at the time they are generated.

See the description for the CEL function from the CLI: core show function CEL

Here is a list of all the available channel field names:

- cidname
- userfield
- cidnum
- amaflags
- cidani
- cidrdnis
- ciddnid
- appdata
- exten
- accountcode
- context
- uniqueid
- channame
- appname
- peer
- eventtime
- eventtype

CELGenUserEvent Application

This application allows the dialplan to insert custom events into the event stream.
For more information, in the CLI, type: **core show application CELGenUserEvent**
Its arguments take this format:

```
CELGenUserEvent(eventname)
```

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!

CEL Configuration Files

cel.conf

Generating Billing Information from CEL

- - - - This is the Next Big Task ****

CEL Storage Backends

Right now, the CEL package will support CSV, Customized CSV, ODBC, PGSQL, TDS, Sqlite3, and Radius back ends. See the doc/celdriver.tex file for how to use these back ends.

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.

MySQL CEL Backend

Using MySQL for Channel Event records is supported by using ODBC and the cel_odbc module.

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
);
```

SQLite 3 CEL Backend

SQLite version 3 is supported in cel_sqlite3_custom.

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)

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,blabla
```

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

Also, characters special to variable substitution, expression evaluation, etc (see below), can be quoted. For example, to literally use a $ on the string "$1231", quote it with a preceding
. Special characters that must be quoted to be used, are [ ] $ " \. (to write \itself, use a backslash.
).

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

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

About Variables

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

To set a variable to a particular value, do:

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

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

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

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

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

stores to the variable "koko" the value "lala" and to variable "lala" the value "blabla".
In fact, everything contained ${here} is just replaced with the value of the variable "here".

Variable Inheritance

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

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

Variable Inheritance Examples

```
Set(__FOO=bar)
```

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

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

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})
```

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.

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

*"One Thousand Five Hundred" =~ "(T[Expressions Examples^])"
returns: Thousand

- "One Thousand Five Hundred" =~ "T[Expressions Examples^]"
  returns: 8

- "One Thousand Five Hundred" : "T[Expressions Examples^]"
  returns: 0

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

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

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

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

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

- 2+8/2
  returns: 6
  Spaces aren't necessary

- (2+8)/2
  returns: 5
  of course

- (3+8)/2
  returns: 5.5

- TRUNC((3+8)/2)
  returns: 5

- FLOOR(2.5)
  returns: 2

- FLOOR(-2.5)
  returns: -3

- CEIL(2.5)
  returns: 3

- CEIL(-2.5)
  returns: -2

- ROUND(2.5)
  returns: 3

- ROUND(3.5)
  returns: 4
- ROUND(-2.5)
  returns: -3
- RINT(2.5)
  returns: 2
- RINT(3.5)
  returns: 4
- RINT(-2.5)
  returns: -2
- RINT(-3.5)
  returns: -4
- TRUNC(2.5)
  returns: 2
- TRUNC(3.5)
  returns: 3
- TRUNC(-3.5)
  returns: -3

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

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.

Conditionals

There is one conditional application - the conditional goto :

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

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

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

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

Example of use :

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

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.

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 ${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.
- ${DYNAMIC_FEATURENAME} The name of the last triggered 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 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

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.

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

DUNDIQUERY and DUNDIRESULT

The DUNDIQUERY and DUNDIRESULT dialplan functions will let you initiate a DUNDi query from the dialplan, see how many results there are, and access each one. Here is some example usage:

```
exten => 1,1,Set(ID=${DUNDIQUERY(1,dundi_test,b)})
exten => 1,n,Set(NUM=${DUNDIRESULT(${ID},getnum)})
exten => 1,n,NoOp(There are ${NUM} results)
exten => 1,n,Set(X=1)
exten => 1,n,While($[${X} <= ${NUM}])
exten => 1,n,NoOp(Result ${X} is ${DUNDIRESULT(${ID},${X})})
exten => 1,n,Set(X=$[${X} + 1])
exten => 1,n,EndWhile
```

DUNDi Peering Agreement

```
DIGIUM GENERAL PEERING AGREEMENT (TM)
Version 1.0.0, September 2004
Copyright (C) 2004 Digium, Inc.
150 West Park Loop Suite 100, 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.
```

E.164 NUmber Mapping (ENUM)

The ENUMLOOKUP Dialplan Function

The ENUMLOOKUP function is more complex than it first may appear, and this guide is to give a general overview and set of examples that may be well-suited for the advanced user to evaluate in their consideration of ENUM or ENUM-like lookup strategies. This document assumes a familiarity with ENUM (RFC3761) or ENUM-like methods, as well as familiarity with NAPTR DNS records (RFC2915, RFC3401-3404). For an overview of NAPTR records, and the use of NAPTRs in the ENUM global phone-number-to-DNS mapping scheme, please see http://www.voip-info.org/tiki-index.php?page=ENUM for more detail.

Using ENUM within Asterisk can be simple or complex, depending on how many failover methods and redundancy procedures you wish to utilize. Implementation of ENUM paths is supposedly defined by the person creating the NAPTR records, but the local administrator may choose to ignore certain NAPTR response methods (URI types) or prefer some over others, which is in contradiction to the RFC. The ENUMLOOKUP method simply provides administrators a method for determining NAPTR results in either the globally unique ENUM (e164.arpa) DNS tree, or in other ENUM-like DNS trees which are not globally unique. The methods to actually create channels ("dial") results given by the ENUMLOOKUP function is then up to the administrator to implement in a way that best suits their environment.

```
Function:
ENUMLOOKUP(number[,Method-type[,options[,record#[,zone-suffix]]]])
```

Performs an ENUM tree lookup on the specified number, method type, and ordinal record offset, and returns one of four different values:

1. Post-parsed NAPTR of one method (URI) type
2. Count of elements of one method (URI) type
3. Count of all method types
4. Full URI of method at a particular point in the list of all possible methods

ENUMLOOKUP Arguments

- number - Telephone number or search string. Only numeric values within this string are parsed; all other digits are ignored for search, but are re-written during NAPTR regexp expansion.
- service_type - tel, sip, h323, iax2, mailto, ...[any other string], ALL. Default type is "sip". Special name of "ALL" will create a list of method types across all NAPTR records for the search number, and then put the results in an ordinal list starting with 1. The position number specified will then be returned, starting with 1 as the first record (lowest value) in the list. The service types are not hardcoded in Asterisk except for the default (sip) if no other service type specified; any method type string (IANA-approved or not) may be used except for the string "ALL".
- options
    - c - count. Returns the number of records of this type are returned (regardless of order or priority.) If "ALL" is the specified service_type, then a count of all methods will be returned for the DNS record.

- record# - Which record to present if multiple answers are returned integer = The record in priority/order sequence based on the total count of records passed back by the query. If a service_type is specified, all entries of that type will be sorted into an ordinal list starting with 1 (by order first, then priority). The default of options is "1"
- zone_suffix - Allows customization of the ENUM zone. Default is e164.arpa.

ENUMLOOKUP Examples

Let's use this ENUM list as an example (note that these examples exist in the DNS, and will hopefully remain in place as example destinations, but they may change or become invalid over time. The end result URIs are not guaranteed to actually work, since some of these hostnames or SIP proxies are imaginary. Of course, the tel: replies go to directory assistance for New York City and San Francisco...) Also note that the complex SIP NAPTR at weight 30 will strip off the leading "+" from the dialed string if it exists. This is probably a better NAPTR than hard-coding the number into the NAPTR, and it is included as a more complex regexp example, though other simpler NAPTRs will work just as well.

```
0.2.0.1.1.6.5.1.0.3.1.loligo.com. 3600 IN NAPTR 10 100 "u" "E2U+tel"
"Unable to render embedded object: File (+12125551212) not found." .
0.2.0.1.1.6.5.1.0.3.1.loligo.com. 3600 IN NAPTR 21 100 "u" "E2U+tel"
"Unable to render embedded object: File (+14155551212) not found." .
0.2.0.1.1.6.5.1.0.3.1.loligo.com. 3600 IN NAPTR 25 100 "u" "E2U+sip"
"Unable to render embedded object: File (2203@sip.fox-den.com) not found."
.
0.2.0.1.1.6.5.1.0.3.1.loligo.com. 3600 IN NAPTR 26 100 "u" "E2U+sip"
"Unable to render embedded object: File (1234@sip-2.fox-den.com) not
found." .
0.2.0.1.1.6.5.1.0.3.1.loligo.com. 3600 IN NAPTR 30 100 "u" "E2U+sip"
"Unable to render embedded object: File (\\1@sip-3.fox-den.com) not found."
.
0.2.0.1.1.6.5.1.0.3.1.loligo.com. 3600 IN NAPTR 55 100 "u" "E2U+mailto"
"Unable to render embedded object: File (jtodd@fox-den.com) not found." .
```

Example 1: Simplest case, using first SIP return (use all defaults except for domain name)

```
exten => 100,1,Set(foo=${ENUMLOOKUP(+13015611020,,,,loligo.com)})
```

returns: ${foo}="2203@sip.fox-den.com"

Example 2: What is the first "tel" pointer type for this number? (after sorting by order/preference; default of "1" is assumed in options field)

```
exten => 100,1,Set(foo=${ENUMLOOKUP(+13015611020,tel,,,loligo.com)})
```

returns: ${foo}="+12125551212"

Example 3: How many "sip" pointer type entries are there for this number?

```
exten => 100,1,Set(foo=${ENUMLOOKUP(+13015611020,sip,c,,loligo.com)})
```

returns: ${foo}=3

Example 4: For all the "tel" pointer type entries, what is the second one in the list? (after sorting by preference)

```
exten => 100,1,Set(foo=${ENUMLOOKUP(+13015611020,tel,,2,loligo.com)})
```

returns: ${foo}="+14155551212"

Example 5: How many NAPTRs (tel, sip, mailto, etc.) are in the list for this number?

```
exten => 100,1,Set(foo=${ENUMLOOKUP(+13015611020,ALL,c,,loligo.com)})
```

returns: ${foo}=6

Example 6: Give back the second full URI in the sorted list of all NAPTR URIs:

```
exten => 100,1,Set(foo=${ENUMLOOKUP(+13015611020,ALL,,2,loligo.com)})
```

returns: ${foo}="tel:+14155551212" [note the "tel:" prefix in the string]

Example 7: Look up first SIP entry for the number in the e164.arpa zone (all defaults)

```
exten => 100,1,Set(foo=${ENUMLOOKUP(+437203001721)})
```

returns: ${foo}="enum-test@sip.nemox.net" [note: this result is subject to change as it is "live" DNS and not under my control]

Example 8: Look up the ISN mapping in freenum.org alpha test zone

```
exten => 100,1,Set(foo=${ENUMLOOKUP(1234*256,,,,freenum.org)})
```

returns: ${foo}="1234@204.91.156.10" [note: this result is subject to change as it is "live" DNS]

Example 9: Give back the first SIP pointer for a number in the enum.yoydynelabs.com zone (invalid lookup)

```
exten => 100,1,Set(foo=${ENUMLOOKUP(1234567890,sip,,1,enum.yoyodynelabs.com)})
```

returns: ${foo}=""

ENUMLOOKUP Usage Notes and Subtle Features

- The use of "" in lookups is confusing, and warrants further explanation. All E.164 numbers ("global phone numbers") by definition need a leading "" during ENUM lookup. If you neglect to add a leading "", you may discover that numbers that seem to exist in the DNS aren't getting matched by the system or are returned with a null string result. This is due to the NAPTR reply requiring a "" in the regular expression matching sequence. Older versions of Asterisk add a "" from within the code, which may confuse administrators converting to the new function. Please ensure that all ENUM (e164.arpa) lookups contain a leading "" before lookup, so ensure your lookup includes the leading plus sign. Other DNS trees may or may not require a leading "" - check before using those trees, as it is possible the parsed NAPTRs will not provide correct results unless you have the correct dialed string. If you get console messages like "WARNING[24907]: enum.c:222 parse_naptr: NAPTR Regex match failed." then it is very possible that the returned NAPTR expects a leading "" in the search string (or the returned NAPTR is mis-formed.)
- If a query is performed of type "c" ("count") and let's say you get back 5 records and then some seconds later a query is made against record 5 in the list, it may not be the case that the DNS resolver has the same answers as it did a second or two ago - maybe there are only 4 records in the list in the newest query. The resolver should be the canonical storage location for DNS records, since that is the intent of ENUM. However, some obscure future cases may have wildly changing NAPTR records within several seconds. This is a corner case, and probably only worth noting as a very rare circumstance. (note: I do not object to Asterisk's dnsmgr method of locally caching DNS replies, but this method needs to honor the TTL given by the remote zone master. Currently, the ENUMLOOKUP function does not use the dnsmgr method of caching local DNS replies.)
- If you want strict NAPTR value ordering, then it will be necessary to use the "ALL" method to incrementally step through the different returned NAPTR pointers. You will need to use string manipulation to strip off the returned method types, since the results will look like "sip:12125551212" in the returned value. This is a non-trivial task, though it is required in order to have strict RFC compliance and to comply with the desires of the remote party who is presenting NAPTRs in a particular order for a reason.
- Default behavior for the function (even in event of an error) is to move to the next priority, and the result is a null value. Most ENUM lookups are going to be failures, and it is the responsibility of the dialplan administrator to manage error conditions within their dialplan. This is a change from the old app_enumlookup method and it's arbitrary priority jumping based on result type or failure.
- Anything other than digits will be ignored in lookup strings. Example: a search string of "+4372030blah01721" will turn into 1.2.7.1.0.0.3.0.2.7.3.4.e164.arpa. for the lookup. The NAPTR parsing may cause unexpected results if there are strings inside your NAPTR lookups.
  If there exist multiple records with the same weight and order as a result of your query, the function will RANDOMLY select a single NAPTR from those equal results.
- Currently, the function ignores the settings in enum.conf as the search zone name is now specified within the function, and the H323 driver can be chosen by the user via the dialplan. There were no other values in this file, and so it becomes deprecated.
- The function will digest and return NAPTRs which use older (deprecated) style, reversed method strings such as "sip+E2U" instead of the more modern "E2U+sip"
- There is no provision for multi-part methods at this time. If there are multiple NAPTRs with (as an example) a method of "E2U+voice:sip" and then another NAPTR in the same DNS record with a method of ""E2U+sip", the system will treat these both as method "sip" and they will be separate records from the perspective of the function. Of course, if both records point to the same URI and have equal priority/weight (as is often the case) then this will cause no serious difficulty, but it bears mentioning.
- ISN (ITAD Subscriber Number) usage: If the search number is of the form ABC*DEF (where ABC and DEF are at least one numeric digit) then perform an ISN-style lookup where the lookup is manipulated to C.B.A.DEF.domain.tld (all other settings and options apply.) See http://www.freenum.org/ for more details on ISN lookups. In the unlikely event you wish to avoid ISN re-writes, put an "n" as the first digit of the search string - the "n" will be ignored for the search.

More ENUMLOOKUP Examples

All examples below except where noted use "e164.arpa" as the referenced domain, which is the default domain name for ENUMLOOKUP. All numbers are assumed to not have a leading "+" as dialed by the inbound channel, so that character is added where necessary during

ENUMLOOKUP function calls.

extensions.conf

```
; example 1
;
; Assumes North American international dialing (011) prefix.
; Look up the first SIP result and send the call there, otherwise
; send the call out a PRI. This is the most simple possible
; ENUM example, but only uses the first SIP reply in the list of
; NAPTR(s).
;
exten => _011.,1,Set(enumresult=${ENUMLOOKUP(${EXTEN:3})})
exten => _011.,n,Dial(SIP/${enumresult})
exten => _011.,n,Dial(DAHDI/g1/${EXTEN})

;
; example 2
;
; Assumes North American international dialing (011) prefix.
; Check to see if there are multiple SIP NAPTRs returned by
; the lookup, and dial each in order. If none work (or none
; exist) then send the call out a PRI, group 1.
;
exten => _011.,1,Set(sipcount=${ENUMLOOKUP(${EXTEN:3},sip,c)}|counter=0)
exten => _011.,n,While($["${counter}"<"${sipcount}"])
exten => _011.,n,Set(counter=$[${counter}+1])
exten => _011.,n,Dial(SIP/${ENUMLOOKUP(${EXTEN:3},sip,,${counter})})
exten => _011.,n,EndWhile
exten => _011.,n,Dial(DAHDI/g1/${EXTEN})

;
; example 3
;
; This example expects an ${EXTEN} that is an e.164 number (like
; 14102241145 or 437203001721)
; Search through e164.arpa and then also search through e164.org
; to see if there are any valid SIP or IAX termination capabilities.
; If none, send call out via DAHDI channel 1.
;
; Start first with e164.arpa zone...
;
exten => _X.,1,Set(sipcount=${ENUMLOOKUP(${EXTEN},sip,c)}|counter=0)
exten => _X.,2,GotoIf($["${counter}"<"${sipcount}"]?3:6)
exten => _X.,3,Set(counter=$[${counter}+1])
exten => _X.,4,Dial(SIP/${ENUMLOOKUP(${EXTEN},sip,,${counter})})
exten => _X.,5,GotoIf($["${counter}"<"${sipcount}"]?3:6)
;
exten => _X.,6,Set(iaxcount=${ENUMLOOKUP(${EXTEN},iax2,c)}|counter=0)
exten => _X.,7,GotoIf($["${counter}"<"${iaxcount}"]?8:11)
exten => _X.,8,Set(counter=$[${counter}+1])
exten => _X.,9,Dial(IAX2/${ENUMLOOKUP(${EXTEN},iax2,,${counter})})
exten => _X.,10,GotoIf($["${counter}"<"${iaxcount}"]?8:11)
;
exten => _X.,11,NoOp("No valid entries in e164.arpa for ${EXTEN} - checking in
e164.org")
;
```

```
; ...then also try e164.org, and look for SIP and IAX NAPTRs...
;
exten => _X.,12,Set(sipcount=${ENUMLOOKUP(${EXTEN},sip,c,,e164.org)}|counter=0)
exten => _X.,13,GotoIf($["${counter}"<"${sipcount}"]?14:17)
exten => _X.,14,Set(counter=$[${counter}+1])
exten => _X.,15,Dial(SIP/${ENUMLOOKUP(${EXTEN},sip,,${counter},e164.org)})
exten => _X.,16,GotoIf($["${counter}"<"${sipcount}"]?14:17)
;
exten => _X.,17,Set(iaxcount=${ENUMLOOKUP(${EXTEN},iax2,c,,e164.org)}|counter=0)
exten => _X.,18,GotoIf($["${counter}"<"${iaxcount}"]?19:22)
exten => _X.,19,Set(counter=$[${counter}+1])
exten => _X.,20,Dial(IAX2/${ENUMLOOKUP(${EXTEN},iax2,,${counter},e164.org)})
exten => _X.,21,GotoIf($["${counter}"<"${iaxcount}"]?19:22)
;
; ...then send out PRI.
;
exten => _X.,22,NoOp("No valid entries in e164.org for ${EXTEN} - sending out via
```

```
DAHDI")
exten => _X.,23,Dial(DAHDI/g1/${EXTEN})
```

Features

Miscellaneous documents that talk about Asterisk functionality. All of this needs to be integrated into Configuration and Operation.

Asterisk Applications

This page is a container page for Asterisk applications, e.g. those things that appear in the apps source directory.

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.

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...

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 Command Line Interface

In addition to being the console for Asterisk, the CLI also sports several features that make it very helpful to use for obtaining information and affecting system configuration. The console can also be seen by starting a remote console, which connects to the running daemon and shows much of the same information as if using the daemon in foreground mode.

Connecting a remote console is as easy as using the -r or -R flags. The only difference between these flags is that the uppercase variation (-R) will automatically reconnect to the daemon (or at least retry) if the daemon restarts. To exit a remote console, simply type 'quit' or 'exit'. Please note that you can differentiate between a remote console and the 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, you can use the 'stop' set of commands, such as 'stop now',
'stop gracefully', or 'stop when convenient'.

Once on the console, the 'help' command may be used to see a list of commands available for use. Note that in addition to the 'help' command, the Asterisk CLI sports tab command line completion on all commands, including many arguments. To use tab command line completion, 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.

The 'help' command may also be used to obtain more detailed information on how to use a particular command. 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.

Asterisk Manager Interface (AMI) Changes

Container page for AMI related content.

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

...

Building Queues

Building Queues

Written by: Leif Madsen
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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.

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.

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:

```
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]
persistantmembers=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
```

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.

Call Completion Supplementary Services

Placeholder page for CCSS information.

Call Queues

Template holder while we wait for input on a good README for call queues. Please open a bug report and add text to this document

General advice on the agent channel

Using dynamic queue members

SIP channel configuration

Queues depend on the channel driver reporting the proper state for each member of the queue. To get proper signalling on
queue members that use the SIP channel driver, you need to enable a call limit (could be set to a high value so it
is not put into action) and also make sure that both inbound and outbound calls are accounted for.

Example:

```
[general]
limitonpeer = yes

[peername]
type=friend
call-limit=10
```

Other references

- queuelog.txt
- queues-with-callback-members.txt

(Should we merge those documents into this?)

Channel Drivers

placeholder page to store the various channel driver subpages

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>
===
=============================================================
```

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.

Distributed Device State

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.

DUNDi - Distributed Universal Number Discovery

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.
```

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` com mand 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.

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.
```

IAX2 Security

IAX2 Security

Document Version 1.0

09/03/09

Asterisk Development Team <asteriskteam@digium.com>

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.

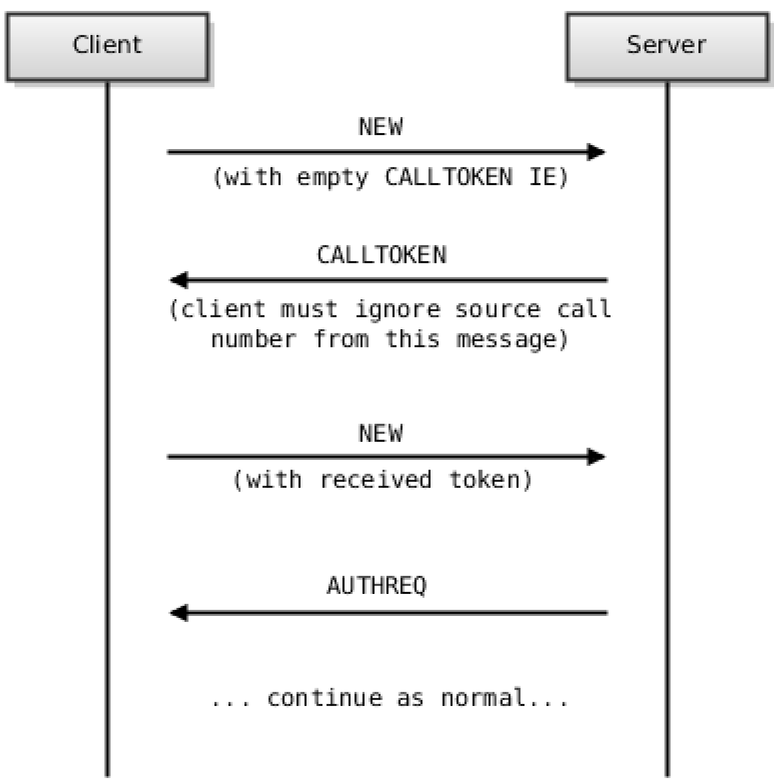
This request MUST include the CALLTOKEN IE with a copy of the value of the CALLTOKEN IE received in the CALLTOKEN response. The IE value is an opaque value. Clients MUST be able to accept a CALLTOKEN payload of any length, up to the maximum length allowed in an IAX2 IE.

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.

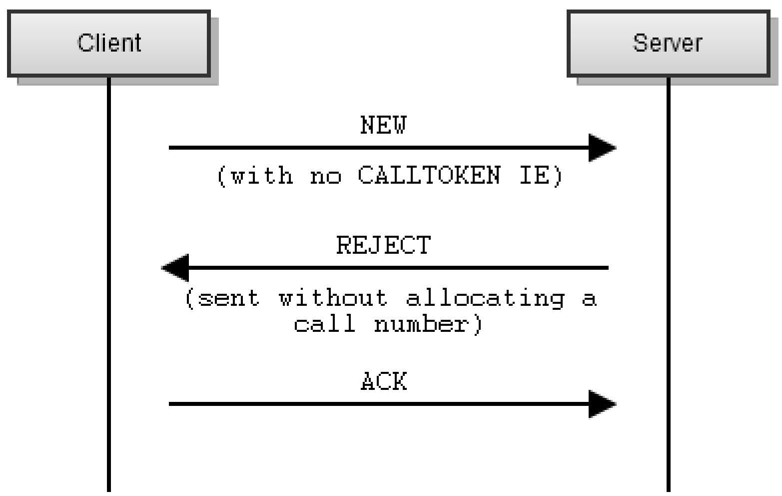
If a server receives a request with an invalid CALLTOKEN IE value, then it MUST drop it and not respond.

Example Message Exchanges

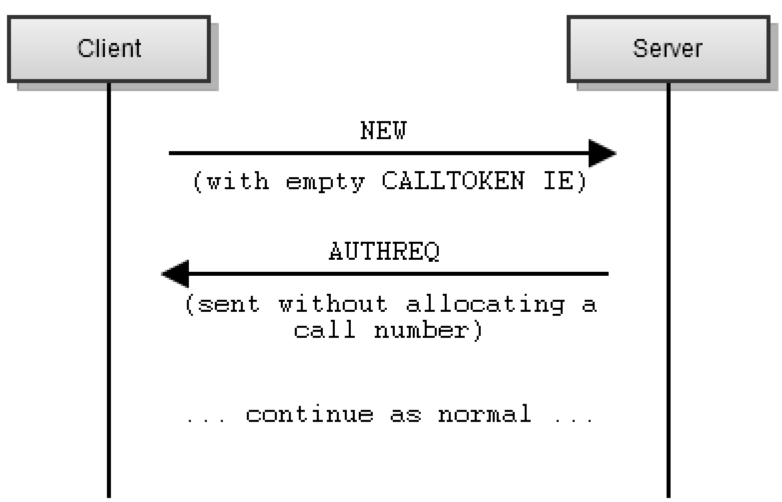
Call Setup



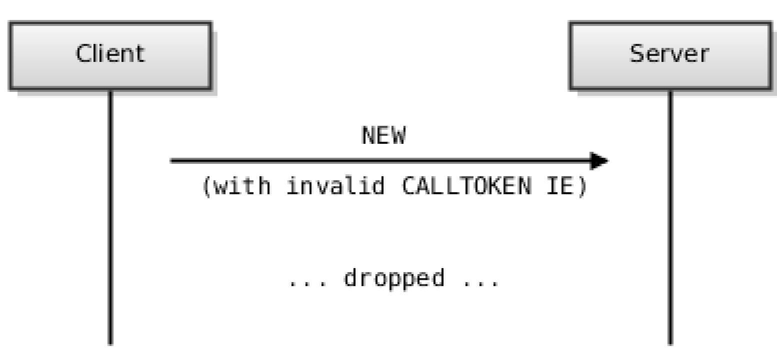
Call Setup, client does not support CALLTOKEN



Call Setup, client supports CALLTOKEN, server does not



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.

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
```

Open Settlement Protocol (OSP) User Guide

OSP User Guide for Asterisk V1.6

9 February 2007

Table of Contents

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Revision History
Revision Date of Issue Description

```
1        26 Jul 2005   OSP Module User Guide for Asterisk V1.2
1.4      16 Jun 2006   OSP Module User Guide for Asterisk V1.4
1.6.0    13 Dec 2006   OSP Module User Guide for Asterisk V1.6
1.6.1    4 Jan 2007    Clarifying edits, add revision history, add general
                       purpose extensions.conf example
1.6.2    9 Feb 2007    Replace OSP Toolkit site from SIPfoundry with
                       SourceForge
```

1 Introduction
This document provides instructions on how to build and configure Asterisk V1.6 with the OSP Toolkit to enable secure, multi-lateral peering. This document is also available in the Asterisk source package as doc/osp.txt. The OSP Toolkit is an open source implementation of the OSP peering protocol and is freely available from https://sourceforge.net/projects/osp-toolkit. The OSP standard defined by the European Telecommunications Standards Institute (ETSI TS 101 321) www.etsi.org. If you have questions or need help, building Asterisk with the OSP Toolkit, please post your question on the OSP mailing list at https://lists.sourceforge.net/lists/listinfo/osp-toolkit-client.

2 OSP Toolkit
Please reference the OSP Toolkit document "How to Build and Test the OSP Toolkit" available from https://sourceforge.net/projects/osp-toolkit.

2.1 Build OSP Toolkit
The software listed below is required to build and use the OSP Toolkit:

- OpenSSL (required for building) - Open Source SSL protocol and Cryptographic Algorithms (version 0.9.7g recommended) from www.openssl.org. Pre-compiled OpenSSL binary packages are not recommended because of the binary compatibility issue.
- Perl (required for building) - A programming language used by OpenSSL for compilation. Any version of Perl should work. One version of Perl is available from www.activestate.com/Products/ActivePer. If pre-compiled OpenSSL packages are used, Perl package is not required.
- C compiler (required for building) - Any C compiler should work. The GNU Compiler Collection from www.gnu.org is routinely used for building the OSP Toolkit for testing.

- OSP Server (required for testing) - Access to any OSP server should work. An open source reference OSP server developed by Cisco System is available at http://www.vovida.org/applications/downloads/openosp/. RAMS, a java based open source OSP server is available at https://sourceforge.net/projects/rams. A free version of the TransNexus commercial OSP server may be downloaded from http://www.transnexus.com/OSP%20Toolkit/Peering_Server/VoIP_Peering_Server.htm.

2.1.1 Unpacking the Toolkit

After downloading the OSP Toolkit (version 3.3.6 or later release) from www.sourceforge.net, perform the following steps in order:

- Copy the OSP Toolkit distribution into the directory where it will reside. The default directory for the OSP Toolkit is /usr/src.

*Un-package the distribution file by executing the following command:

```
gunzip –c OSPToolkit-###.tar.gz | tar xvf –
```

Where ### is the version number separated by underlines. For example, if the version is 3.3.6, then the above command would be:

```
gunzip –c OSPToolkit-3_3_6.tar.gz | tar xvf –
```

A new directory (TK-3_3_6-20060303) will be created within the same directory as the tar file.

- Go to the TK-3_3_6-20060303 directory by running this command:

```
cd TK-3_3_6-20060303
```

Within this directory, you will find directories and files similar to what is listed below if the command "ls -F" is executed):

```
ls -F
enroll/
RelNotes.txt    lib/
README.txt    license.txt
bin/     src/
crypto/     test/
include/
```

2.1.2 Preparing to build the OSP Toolkit

- Compile OpenSSL according to the instructions provided with the OpenSSL distribution (You would need to do this only if you don’t have openssl already).
- Copy the OpenSSL header files (the *.h files) into the crypto/openssl directory within the osptoolkit directory. The OpenSSL header files

are located under the openssl/include/openssl directory.

- Copy the OpenSSL library files (libcrypto.a and libssl.a) into the lib directory within the osptoolkit directory. The OpenSSL library files are located under the openssl directory.
  Note: Since the Asterisk requires the OpenSSL package. If the OpenSSL package has been installed, steps 4 through 6 are not necessary.
- Optionally, change the install directory of the OSP Toolkit. Open the Makefile in the /usr/src/TK-3_3_6-20060303/src directory, look for the install path variable – INSTALL_PATH, and edit it to be anywhere you want (defaults /usr/local).
  Note: Please change the install path variable only if you are familiar with both the OSP Toolkit and the Asterisk.

2.1.3 Building the OSP Toolkit

- From within the OSP Toolkit directory (/usr/src/TK-3_3_6-20060303), start the compilation script by executing the following commands:

```
cd src
make clean; make build
```

2.1.4 Installing the OSP Toolkit

The header files and the library of the OSP Toolkit should be installed. Otherwise, you must specify the OSP Toolkit path for the Asterisk.

- Use the make script to install the Toolkit.

```
make install
```

The make script is also used to install the OSP Toolkit header files and the library into the INSTALL_PATH specified in the Makefile.

> Please make sure you have the rights to access the INSTALL_PATH directory. For example, in order to access /usr/local directory, root privileges are required.

2.1.5 Building the Enrollment Utility

Device enrollment is the process of establishing a trusted cryptographic relationship between the VoIP device and the OSP Server. The Enroll program is a utility application for establishing a trusted relationship between an OSP client and an OSP server. Please see the document "Device Enrollment" at http://www.transnexus.com/OSP%20Toolkit/OSP%20Toolkit%20Documents/Device_Enrollment.pdf for more information about the enroll application.

- From within the OSP Toolkit directory (example: /usr/src/TK-3_3_6-20060303), execute the following commands at the command prompt:

```
cd enroll
make clean; make linux
```

Compilation is successful if there are no errors in the compiler output. The enroll program is now located in the OSP Toolkit/bin directory (example: /usr/src/ TK-3_3_6-20060303/bin).

2.2 Obtain Crypto Files

The OSP module in Asterisk requires three crypto files containing a local certificate (localcert.pem), private key (pkey.pem), and CA certificate (cacert_0.pem). Asterisk will try to load the files from the Asterisk public/private key directory - /var/lib/asterisk/keys. If the files are not present, the OSP module will not start and the Asterisk will not support the OSP protocol. Use the enroll.sh script from the toolkit distribution to enroll Asterisk with an OSP server and obtain the crypto files. Documentation explaining how to use the enroll.sh script (Device Enrollment) to enroll with an OSP server is available at http://www.transnexus.com/OSP%20Toolkit/OSP%20Toolkit%20Documents/Device_Enrollment.pdf. Copy the files generated by the enrollment process to the Asterisk /var/lib/asterisk/keys directory.

> ⚠ The osptestserver.transnexus.com is configured only for sending and receiving non-SSL messages, and issuing signed tokens. If you need help, post a message on the OSP mailing list at https://lists.sourceforge.net/lists/listinfo/osp-toolkit-client

The enroll.sh script takes the domain name or IP addresses of the OSP servers that the OSP Toolkit needs to enroll with as arguments, and then generates pem files – cacert_#.pem, certreq.pem, localcert.pem, and pkey.pem. The '#' in the cacert file name is used to differentiate the ca certificate file names for the various SP's (OSP servers). If only one address is provided at the command line, cacert_0.pem will be generated. If 2 addresses are provided at the command line, 2 files will be generated – cacert_0.pem and cacert_1.pem, one for each SP (OSP server). The example below shows the usage when the client is registering with osptestserver.transnexus.com.

```
./enroll.sh osptestserver.transnexus.com
Generating a 512 bit RSA private key
........................++++++++++++
.........++++++++++++
writing new private key to 'pkey.pem'
-----
You are about to be asked to enter information that will be incorporated
into your certificate request.
What you are about to enter is what is called a Distinguished Name or a DN.
There are quite a few fields but you can leave some blank
For some fields there will be a default value,
If you enter '.', the field will be left blank.
-----
Country Name (2 letter code) [AU]: _______
State or Province Name (full name) [Some-State]: _______
Locality Name (eg, city) []:_______
Organization Name (eg, company) [Internet Widgits Pty Ltd]: _______
Organizational Unit Name (eg, section) []:_______
Common Name (eg, YOUR name) []:_______
Email Address []:_______

Please enter the following 'extra' attributes
to be sent with your certificate request
A challenge password []:_______
An optional company name []:_______

Error Code returned from openssl command : 0

CA certificate received
[SP: osptestserver.transnexus.com]Error Code returned from getcacert command : 0

output buffer after operation: operation=request
output buffer after nonce: operation=request&nonce=1655976791184458
X509 CertInfo context is null pointer
Unable to get Local Certificate
depth=0 /CN=osptestserver.transnexus.com/O=OSPServer
verify error:num=18:self signed certificate
verify return:1
depth=0 /CN=osptestserver.transnexus.com/O=OSPServer
verify return:1
The certificate request was successful.
Error Code returned from localcert command : 0
```

The files generated should be copied to the /var/lib/asterisk/keys directory.

> ⚠ The script enroll.sh requires AT&T korn shell (ksh) or any of its compatible variants. The /usr/src/TK-3_3_6-20060303/bin directory should be in the PATH variable. Otherwise, enroll.sh cannot find the enroll file.

3 Asterisk

In Asterisk, all OSP support is implemented as dial plan functions. In Asterisk V1.6, all combinations of routing between OSP and non-OSP enabled networks using any combination of SIP, H.323 and IAX protocols are fully supported. Section
3.1 describes the three easy steps to add OSP support to Asterisk:

1. Build Asterisk with OSP Toolkit
2. Configure osp.conf file
3. Cut and paste to extensions.conf

Sections 3.2 and 3.3 provide a detailed explanation of OSP dial plan functions and configuration examples. The detailed information provided in Sections 3.2 and 3.3 is not required for operating Asterisk with OSP, but may be helpful to developers who want to customize their Asterisk OSP implementation.

3.1 Configure for OSP Support

3.1.1 Build Asterisk with OSP Toolkit

The first step is to build Asterisk with the OSP Toolkit. If the OSP Toolkit is installed in the default install directory, /usr/local, no additional configuration is required. Compile Asterisk according to the instructions provided with the Asterisk distribution.

If the OSP Toolkit is installed in another directory, such as /myosp, Asterisk must be configured with the location of the OSP Toolkit. See the example below.

```
--with-osptk=/myosp
```

> ⚠ Please change the install path only if you familiar with both the OSP Toolkit and the Asterisk. Otherwise, the change may result in Asterisk not supporting the OSP protocol.

3.1.2 osp.conf

The /etc/asterisk/osp.conf file, shown below, contains configuration parameters for using OSP. Two parameters, servicepoint and source must be configured. The default values for all other parameters will work well for standard OSP implementations.

```
;
; Open Settlement Protocol Sample Configuration File
;
; This file contains configuration of OSP server providers that
; are used by the Asterisk OSP module.  The section "general" is
; reserved for global options.  All other sections describe specific
```

```
; OSP Providers.  The provider "default" is used when no provider is
; otherwise specified.
:
: The "servicepoint" and "source" parameters must be configured.  For
; most implementations the other parameters in this file can be left
; unchanged.
;
[general]
;
; Enable cryptographic acceleration hardware.
;
accelerate=no
;
; Defines the status of tokens that Asterisk will validate.
; 0 - signed tokens only
; 1 - unsigned tokens only
; 2 - both signed and unsigned
; The default value is 0, i.e. the Asterisk will only validate signed
; tokens.
;
tokenformat=0
;
[default]
;
; List all service points (OSP servers) for this provider.  Use
; either domain name or IP address.  Most OSP servers use port 5045.
;
;servicepoint=http://osptestserver.transnexus.com:5045/osp
servicepoint=http://OSP server IP:5045/osp
;
; Define the "source" device for requesting OSP authorization.
: This value is usually the domain name or IP address of the
: the Asterisk server.
;
;source=domain name or [IP address in brackets]
source=[host IP]
;
; Define path and file name of crypto files.
; The default path for crypto file is /var/lib/asterisk/keys.  If no
; path is defined, crypto files should be in
; /var/lib/asterisk/keys directory.
;
; Specify the private key file name.
; If this parameter is unspecified or not present, the default name
; will be the osp.conf section name followed by "-privatekey.pem"
; (for example: default-privatekey.pem)
;
privatekey=pkey.pem
;
; Specify the local certificate file.
; If this parameter is unspecified or not present, the default name
; will be the osp.conf section name followed by "- localcert.pem "
; (for example: default-localcert.pem)
;
localcert=localcert.pem
;
; Specify one or more Certificate Authority key file names.  If none
; are listed, a single Certificate Authority key file name is added
; with the default name of the osp.conf section name followed by
```

```
; "-cacert_0.pem " (for example: default-cacert_0.pem)
;
cacert=cacert_0.pem
;
; Configure parameters for OSP communication between Asterisk OSP
; client and OSP servers.
;
; maxconnections: Max number of simultaneous connections to the
;                 provider OSP server (default=20)
; retrydelay:     Extra delay between retries (default=0)
; retrylimit:     Max number of retries before giving up (default=2)
; timeout:        Timeout for response in milliseconds (default=500)
;
maxconnections=20
retrydelay=0
retrylimit=2
timeout=500
;
; Set the authentication policy.
; 0 - NO        - Accept all calls.
; 1 – YES       - Accept calls with valid token or no token.
;                  Block calls with invalid token.
; 2 – EXCLUSIVE – Accept calls with valid token.
;                  Block calls with invalid token or no token.
; Default is 1,
;
authpolicy=1
;
; Set the default destination protocol. The OSP module supports
; SIP, H323, and IAX protocols.  The default protocol is set to SIP.
```

```
;
defaultprotocol=SIP
```

3.1.3 extensions.conf

OSP functions are implemented as dial plan functions in the extensions.conf file. To add OSP support to your Asterisk server, simply copy and paste the text box below to your extensions.conf file. These functions will enable your Asterisk server to support all OSP call scenarios. Configuration of your Asterisk server for OSP is now complete.

```
[globals]
DIALOUT=DAHDI/1

[SrcGW] ; OSP Source Gateway
exten => _XXXX.,1,NoOp(OSPSrcGW)
; Set calling number if necessary
exten => _XXXX.,n,Set(CALLERID(numner)=1234567890)
; OSP lookup using default provider, if fail/error jump to lookup+101
exten => _XXXX.,n(lookup),OSPLookup(${EXTEN}||j)
; Deal with outbound call according to protocol
exten => _XXXX.,n,Macro(outbound)
; Dial to destination, 60 timeout, with call duration limit
exten => _XXXX.,n,Dial(${OSPDIALSTR},60,oL($[${OSPOUTTIMELIMIT}*1000]))
; Wait 1 second
exten => _XXXX.,n,Wait,1
; Hangup
exten => _XXXX.,n,Hangup
; Deal with OSPLookup fail/error
exten => _XXXX.,lookup+101,Hangup
exten => h,1,NoOp()
; OSP report usage
exten => h,n,OSPFinish(${HANGUPCAUSE})

[DstGW] ; OSP Destination Gateway
exten => _XXXX.,1,NoOp(OSPDstGW)
; Deal with inbound call according to protocol
exten => _XXXX.,n,Macro(inbound)
; Validate token using default provider, if fail/error jump to auth+101
exten => _XXXX.,n(auth),OSPAuth(|j)
; Ringing
exten => _XXXX.,n,Ringing
; Wait 1 second
exten => _XXXX.,n,Wait,1
; Check inbound call duration limit
exten => _XXXX.,n,GoToIf($[${OSPINTIMELIMIT}=0]?100:200)
; Without duration limit
exten => _XXXX.,100,Dial(${DIALOUT},15,o)
exten => _XXXX.,n,Goto(1000)
; With duration limit
exten => _XXXX.,200,Dial(${DIALOUT},15,oL($[${OSPINTIMELIMIT}*1000]))
exten => _XXXX.,n,Goto(1000)
; Wait 1 second
```

```
exten => _XXXX.,1000,Wait,1
; Hangup
exten => _XXXX.,n,Hangup
; Deal with OSPAuth fail/error
exten => _XXXX.,auth+101,Hangup
exten => h,1,NoOp()
; OSP report usage
exten => h,n,OSPFinish(${HANGUPCAUSE})

[GeneralProxy] ; Proxy
exten => _XXXX.,1,NoOp(OSP-GeneralProxy)
; Deal with inbound call according to protocol
exten => _XXXX.,n,Macro(inbound)
; Validate token using default provider, if fail/error jump to auth+101
exten => _XXXX.,n(auth),OSPAuth(|j)
; OSP lookup using default provider, if fail/error jump to lookup+101
exten => _XXXX.,n(lookup),OSPLookup(${EXTEN}||j)
; Deal with outbound call according to protocol
exten => _XXXX.,n,Macro(outbound)
; Dial to destination, 14 timeout, with call duration limit
exten => _XXXX.,n,Dial(${OSPDIALSTR},14,oL($[${OSPOUTTIMELIMIT}*1000]))
; OSP lookup next destination using default provider, if fail/error jump to next1+101
exten => _XXXX.,n(next1),OSPNext(${HANGUPCAUSE}||j)
; Deal with outbound call according to protocol
exten => _XXXX.,n,Macro(outbound)
; Dial to destination, 15 timeout, with call duration limit
exten => _XXXX.,n,Dial(${OSPDIALSTR},15,oL($[${OSPOUTTIMELIMIT}*1000]))
; OSP lookup next destination using default provider, if fail/error jump to next2+101
exten => _XXXX.,n(next2),OSPNext(${HANGUPCAUSE}||j)
; Deal with outbound call according to protocol
exten => _XXXX.,n,Macro(outbound)
; Dial to destination, 16 timeout, with call duration limit
exten => _XXXX.,n,Dial(${OSPDIALSTR},16,oL($[${OSPOUTTIMELIMIT}*1000]))
; Hangup
exten => _XXXX.,n,Hangup
; Deal with OSPAuth fail/error
exten => _XXXX.,auth+101,Hangup
; Deal with OSPLookup fail/error
exten => _XXXX.,lookup+101,Hangup
; Deal with OSPNext fail/error
exten => _XXXX.,next1+101,Hangup
; Deal with OSPNext fail/error
exten => _XXXX.,next2+101,Hangup
exten => h,1,NoOp()
; OSP report usage
exten => h,n,OSPFinish(${HANGUPCAUSE})

[macro-inbound]
exten => s,1,NoOp(inbound)
; Get inbound protocol
exten => s,n,Set(CHANTECH=${CUT(CHANNEL,/,1)})
exten => s,n,GoToIf($["${CHANTECH}"="H323"]?100)
exten => s,n,GoToIf($["${CHANTECH}"="IAX2"]?200)
exten => s,n,GoToIf($["${CHANTECH}"="SIP"]?300)
exten => s,n,GoTo(1000)
; H323 --------------------------------------------------------
; Get peer IP
exten => s,100,Set(OSPPEERIP=${H323CHANINFO(peerip)})
; Get OSP token
```

```
exten => s,n,Set(OSPINTOKEN=${H323CHANINFO(osptoken)})
exten => s,n,GoTo(1000)
; IAX ----------------------------------------------------------
; Get peer IP
exten => s,200,Set(OSPPEERIP=${IAXPEER(CURRENTCHANNEL)})
; Get OSP token
exten => s,n,Set(OSPINTOKEN=${IAXCHANINFO(osptoken)})
exten => s,n,GoTo(1000)
; SIP ----------------------------------------------------------
; Get peer IP
exten => s,300,Set(OSPPEERIP=${SIPCHANINFO(peerip)})
; Get OSP token
exten => s,n,Set(OSPINTOKEN=${SIP_HEADER(P-OSP-Auth-Token)})
exten => s,n,GoTo(1000)
; --------------------------------------------------------------
exten => s,1000,MacroExit

[macro-outbound]
exten => s,1,NoOp(outbound)
; Set calling number which may be translated
exten => s,n,Set(CALLERID(num)=${OSPCALLING})
; Check destinatio protocol
exten => s,n,GoToIf($["${OSPTECH}"="H323"]?100)
exten => s,n,GoToIf($["${OSPTECH}"="IAX2"]?200)
exten => s,n,GoToIf($["${OSPTECH}"="SIP"]?300)
; Something wrong
exten => s,n,Hangup
exten => s,n,GoTo(1000)
; H323 ---------------------------------------------------------
; Set call id
exten => s,100,Set(H323CHANINFO(callid)=${OSPOUTCALLID})
; Set OSP token
exten => s,n,Set(H323CHANINFO(osptoken)=${OSPOUTTOKEN})
exten => s,n,GoTo(1000)
; IAX ----------------------------------------------------------
; Set OSP token
exten => s,200,Set(IAXCHANINFO(osptoken)=${OSPOUTTOKEN})
exten => s,n,GoTo(1000)
; SIP ----------------------------------------------------------
exten => s,300,GoTo(1000)
```

```
; ------------------------------------------------------------
exten => s,1000,MacroExit
```

3.1.4 dahdi/sip/iax/h323/ooh323.conf

There is no configuration required for OSP.

3.2 OSP Dial Plan Functions

This section provides a description of each OSP dial plan function.

3.2.1 OSPAuth

OSP token validation function.
Input:

- OSPPEERIP: last hop IP address
- OSPINTOKEN: inbound OSP token
- provider: OSP service provider configured in osp.conf. If it is empty, default provider is used.
- priority jump

Output:

- OSPINHANDLE: inbound OSP transaction handle
- OSPINTIMELIMIT: inbound call duration limit
- OSPAUTHSTATUS: OSPAuth return value. SUCCESS/FAILED/ERROR

3.2.2 OSPLookup

OSP lookup function.

Input:

- OSPPEERIP: last hop IP address
- OSPINHANDLE: inbound OSP transaction handle
- OSPINTIMELIMIT: inbound call duration limit
- exten: called number
- provider: OSP service provider configured in osp.conf. If it is empty, default provider is used.
- priority jump
- callidtypes: Generate call ID for the outbound call. h: H.323; s: SIP; i: IAX. Only h, H.323, has been implemented.

Output:

- OSPOUTHANDLE: outbound transaction handle
- OSPTECH: outbound protocol
- OSPDEST: outbound destination IP address
- OSPCALLED: outbound called nummber
- OSPCALLING: outbound calling number
- OSPOUTTOKEN: outbound OSP token
- OSPRESULTS: number of remaining destinations
- OSPOUTTIMELIMIT: outbound call duration limit

- OSPOUTCALLIDTYPES: same as input callidtypes
- OSPOUTCALLID: outbound call ID. Only for H.323
- OSPDIALSTR: outbound dial string
- OSPLOOKUPSTATUS: OSPLookup return value. SUCCESS/FAILED/ERROR

3.2.3 OSPNext

OSP lookup next function.

Input:

- OSPINHANDLE: inbound transaction handle
- OSPOUTHANDLE: outbound transaction handle
- OSPINTIMELIMIT: inbound call duration limit
- OSPOUTCALLIDTYPES: types of call ID generated by Asterisk.
- OSPRESULTS: number of remain destinations
- cause: last destination disconnect cause
- priority jump

Output:

- OSPTECH: outbound protocol
- OSPDEST: outbound destination IP address
- OSPCALLED: outbound called number
- OSPCALLING: outbound calling number
- OSPOUTTOKEN: outbound OSP token
- OSPRESULTS: number of remain destinations
- OSPOUTTIMELIMIT: outbound call duration limit
- OSPOUTCALLID: outbound call ID. Only for H.323
- OSPDIALSTR: outbound dial string
- OSPNEXTSTATUS: OSPLookup return value. SUCCESS/FAILED/ERROR

3.2.4 OSPFinish

OSP report usage function.

Input:

- OSPINHANDLE: inbound transaction handle
- OSPOUTHANDLE: outbound transaction handle
- OSPAUTHSTATUS: OSPAuth return value
- OSPLOOKUPTSTATUS: OSPLookup return value
- OSPNEXTSTATUS: OSPNext return value
- cause: last destination disconnect cause
- priority jump

Output:

- OSPFINISHSTATUS: OSPLookup return value. SUCCESS/FAILED/ERROR

3.3 extensions.conf Examples

The extensions.conf file example provided in Section 3.1 is designed to handle all OSP call scenarios when Asterisk is used as a source or destination gateway to the PSTN or as a proxy between VoIP networks. The extenstion.conf examples in this section are designed for specific use cases only.

3.3.1 Source Gateway

The examples in this section apply when the Asterisk server is being used as a TDM to VoIP gateway. Calls originate on the TDM network and are converted to VoIP by Asterisk. In these cases, the Asterisk server queries an OSP server to find a route to a VoIP destination. When the call ends, Asterisk sends a CDR to the OSP server.
For SIP protocol.

```
[SIPSrcGW]
exten => _XXXX.,1,NoOp(SIPSrcGW)
; Set calling number if necessary
exten => _XXXX.,n,Set(CALLERID(numner)=CallingNumber)
; OSP lookup using default provider, if fail/error jump to lookup+101
exten => _XXXX.,n(lookup),OSPLookup(${EXTEN}||j)
; Set calling number which may be translated
exten => _XXXX.,n,Set(CALLERID(num)=${OSPCALLING})
; Dial to destination, 60 timeout, with call duration limit
exten => _XXXX.,n,Dial(${OSPDIALSTR},60,oL($[${OSPOUTTIMELIMIT}*1000]))
; Wait 3 seconds
exten => _XXXX.,n,Wait,3
; Hangup
exten => _XXXX.,n,Hangup
; Deal with OSPLookup fail/error
exten => _XXXX.,lookup+101,Hangup
exten => h,1,NoOp()
; OSP report usage
exten => h,n,OSPFinish(${HANGUPCAUSE})
```

For IAX protocol.

```
[IAXSrcGW]
exten => _XXXX.,1,NoOp(IAXSrcGW)
; Set calling number if necessary
exten => _XXXX.,n,Set(CALLERID(numner)=CallingNumber)
; OSP lookup using default provider, if fail/error jump to lookup+101
exten => _XXXX.,n(lookup),OSPLookup(${EXTEN}||j)
; Set outbound OSP token
exten => _XXXX.,n,Set(IAXCHANINFO(osptoken)=${OSPOUTTOKEN})
; Set calling number which may be translated
exten => _XXXX.,n,Set(CALLERID(num)=${OSPCALLING})
; Dial to destination, 60 timeout, with call duration limit
exten => _XXXX.,n,Dial(${OSPDIALSTR},60,oL($[${OSPOUTTIMELIMIT}*1000]))
; Wait 3 seconds
exten => _XXXX.,n,Wait,3
; Hangup
exten => _XXXX.,n,Hangup
; Deal with OSPLookup fail/error
exten => _XXXX.,lookup+101,Hangup
exten => h,1,NoOp()
; OSP report usage
exten => h,n,OSPFinish(${HANGUPCAUSE})
```

For H.323 protocol.

```
[H323SrcGW]
exten => _XXXX.,1,NoOp(H323SrcGW)
; Set calling number if necessary
exten => _XXXX.,n,Set(CALLERID(numner)=CallingNumber)
; OSP lookup using default provider, if fail/error jump to lookup+101
; "h" parameter is used to generate a call id
; Cisco OSP gateways use this call id to validate OSP token
exten => _XXXX.,n(lookup),OSPLookup(${EXTEN}||jh)
; Set outbound call id
exten => _XXXX.,n,Set(OH323CHANINFO(callid)=${OSPOUTCALLID})
; Set outbound OSP token
exten => _XXXX.,n,Set(OH323CHANINFO(osptoken)=${OSPOUTTOKEN})
; Set calling number which may be translated
exten => _XXXX.,n,Set(CALLERID(num)=${OSPCALLING})
; Dial to destination, 60 timeout, with call duration limit
exten => _XXXX.,n,Dial(${OSPDIALSTR},60,oL($[${OSPOUTTIMELIMIT}*1000]))
; Wait 3 seconds
exten => _XXXX.,n,Wait,3
; Hangup
exten => _XXXX.,n,Hangup
; Deal with OSPLookup fail/error
exten => _XXXX.,lookup+101,Hangup
exten => h,1,NoOp()
; OSP report usage
exten => h,n,OSPFinish(${HANGUPCAUSE})
```

3.3.2 Destination Gateway

The examples in this section apply when Asterisk is being used as a VoIP to TDM gateway. VoIP calls are received by Asterisk which validates the OSP peering token and completes to the TDM network. After the call ends, Asterisk sends a CDR to the OSP server.

For SIP protocol

```
[SIPDstGW]
exten => _XXXX.,1,NoOp(SIPDstGW)
; Get peer IP
exten => _XXXX.,n,Set(OSPPEERIP=${SIPCHANINFO(peerip)})
; Get OSP token
exten => _XXXX.,n,Set(OSPINTOKEN=${SIP_HEADER(P-OSP-Auth-Token)})
; Validate token using default provider, if fail/error jump to auth+101
exten => _XXXX.,n(auth),OSPAuth(|j)
; Ringing
exten => _XXXX.,n,Ringing
; Wait 1 second
exten => _XXXX.,n,Wait,1
; Dial phone, timeout 15 seconds, with call duration limit
exten => _XXXX.,n,Dial(${DIALOUTANALOG}/${EXTEN:1},15,oL($[${OSPINTIMELIMIT}*1000]))
; Wait 3 seconds
exten => _XXXX.,n,Wait,3
; Hangup
exten => _XXXX.,n,Hangup
; Deal with OSPAuth fail/error
exten => _XXXX.,auth+101,Hangup
exten => h,1,NoOp()
; OSP report usage
exten => h,n,OSPFinish(${HANGUPCAUSE})
```

For IAX protocol

```
[IAXDstGW]
exten => _XXXX.,1,NoOp(IAXDstGW)
; Get peer IP
exten => _XXXX.,n,Set(OSPPEERIP=${IAXPEER(CURRENTCHANNEL)})
; Get OSP token
exten => _XXXX.,n,Set(OSPINTOKEN=${IAXCHANINFO(osptoken)})
; Validate token using default provider, if fail/error jump to auth+101
exten => _XXXX.,n(auth),OSPAuth(|j)
; Ringing
exten => _XXXX.,n,Ringing
; Wait 1 second
exten => _XXXX.,n,Wait,1
; Dial phone, timeout 15 seconds, with call duration limit
exten => _XXXX.,n,Dial(${DIALOUTANALOG}/${EXTEN:1},15,oL($[${OSPINTIMELIMIT}*1000]))
; Wait 3 seconds
exten => _XXXX.,n,Wait,3
; Hangup
exten => _XXXX.,n,Hangup
; Deal with OSPAuth fail/error
exten => _XXXX.,auth+101,Hangup
exten => h,1,NoOp()
; OSP report usage
exten => h,n,OSPFinish(${HANGUPCAUSE})
```

For H.323 protocol

```
[H323DstGW]
exten => _XXXX.,1,NoOp(H323DstGW)
; Get peer IP
exten => _XXXX.,n,Set(OSPPEERIP=${H323CHANINFO(peerip)})
; Get OSP token
exten => _XXXX.,n,Set(OSPINTOKEN=${H323CHANINFO(osptoken)})
; Validate token using default provider, if fail/error jump to auth+101
exten => _XXXX.,n(auth),OSPAuth(|j)
; Ringing
exten => _XXXX.,n,Ringing
; Wait 1 second
exten => _XXXX.,n,Wait,1
; Dial phone, timeout 15 seconds, with call duration limit
exten => _XXXX.,n,Dial(${DIALOUTANALOG}/${EXTEN:1},15,oL($[${OSPINTIMELIMIT}*1000]))
; Wait 3 seconds
exten => _XXXX.,n,Wait,3
; Hangup
exten => _XXXX.,n,Hangup
; Deal with OSPAuth fail/error
exten => _XXXX.,auth+101,Hangup
exten => h,1,NoOp()
; OSP report usage
exten => h,n,OSPFinish(${HANGUPCAUSE})
```

3.3.3 Proxy

The example in this section applies when Asterisk is a proxy between two VoIP networks.

```
[GeneralProxy]
exten => _XXXX.,1,NoOp(GeneralProxy)
; Get peer IP and inbound OSP token
; SIP, un-comment the following two lines.
;exten => _XXXX.,n,Set(OSPPEERIP=${SIPCHANINFO(peerip)})
;exten => _XXXX.,n,Set(OSPINTOKEN=${SIP_HEADER(P-OSP-Auth-Token)})
; IAX, un-comment the following 2 lines
;exten => _XXXX.,n,Set(OSPPEERIP=${IAXPEER(CURRENTCHANNEL)})
;exten => _XXXX.,n,Set(OSPINTOKEN=${IAXCHANINFO(osptoken)})
; H323, un-comment the following two lines.
;exten => _XXXX.,n,Set(OSPPEERIP=${OH323CHANINFO(peerip)})
;exten => _XXXX.,n,Set(OSPINTOKEN=${OH323CHANINFO(osptoken)})
;----------------------------------------------------------------
; Validate token using default provider, if fail/error jump to auth+101
exten => _XXXX.,n(auth),OSPAuth(|j)
; OSP lookup using default provider, if fail/error jump to lookup+101
; “h” parameter is used to generate a call id for H.323 destinations
; Cisco OSP gateways use this call id to validate OSP token
exten => _XXXX.,n(lookup),OSPLookup(${EXTEN}||jh)
; Set outbound call id and OSP token
; IAX, un-comment the following line.
;exten => _XXXX.,n,Set(IAXCHANINFO(osptoken)=${OSPOUTTOKEN})
; H323, un-comment the following two lines.
;exten => _XXXX.,n,Set(OH323CHANINFO(callid)=${OSPOUTCALLID})
;exten => _XXXX.,n,Set(OH323CHANINFO(osptoken)=${OSPOUTTOKEN})
;----------------------------------------------------------------
; Set calling number which may be translated
exten => _XXXX.,n,Set(CALLERID(num)=${OSPCALLING})
; Dial to destination, 14 timeout, with call duration limit
exten => _XXXX.,n,Dial(${OSPDIALSTR},14,oL($[${OSPOUTTIMELIMIT}*1000]))
; OSP lookup next destination using default provider, if fail/error jump to next1+101
exten => _XXXX.,n(next1),OSPNext(${HANGUPCAUSE}||j)
; Set outbound call id and OSP token
; IAX, un-comment the following line.
;exten => _XXXX.,n,Set(IAXCHANINFO(osptoken)=${OSPOUTTOKEN})
; H323, un-comment the following two lines.
;exten => _XXXX.,n,Set(OH323CHANINFO(callid)=${OSPOUTCALLID})
;exten => _XXXX.,n,Set(OH323CHANINFO(osptoken)=${OSPOUTTOKEN})
;----------------------------------------------------------------
; Set calling number which may be translated
exten => _XXXX.,n,Set(CALLERID(num)=${OSPCALLING})
; Dial to destination, 15 timeout, with call duration limit
exten => _XXXX.,n,Dial(${OSPDIALSTR},15,oL($[${OSPOUTTIMELIMIT}*1000]))
; OSP lookup next destination using default provider, if fail/error jump to next2+101
exten => _XXXX.,n(next2),OSPNext(${HANGUPCAUSE}||j)
; Set outbound call id and OSP token
; IAX, un-comment the following line.
;exten => _XXXX.,n,Set(IAXCHANINFO(osptoken)=${OSPOUTTOKEN})
; H323, un-comment the following two lines.
;exten => _XXXX.,n,Set(OH323CHANINFO(callid)=${OSPOUTCALLID})
;exten => _XXXX.,n,Set(OH323CHANINFO(osptoken)=${OSPOUTTOKEN})
;----------------------------------------------------------------
; Set calling number which may be translated
```

```
exten => _XXXX.,n,Set(CALLERID(num)=${OSPCALLING})
; Dial to destination, 16 timeout, with call duration limit
exten => _XXXX.,n,Dial(${OSPDIALSTR},16,oL($[${OSPOUTTIMELIMIT}*1000]))
; Hangup
exten => _XXXX.,n,Hangup
; Deal with OSPAuth fail/error
exten => _XXXX.,auth+101,Hangup
; Deal with OSPLookup fail/error
exten => _XXXX.,lookup+101,Hangup
; Deal with 1st OSPNext fail/error
exten => _XXXX.,next1+101,Hangup
; Deal with 2nd OSPNext fail/error
exten => _XXXX.,next2+101,Hangup
exten => h,1,NoOp()
```

```
; OSP report usage
exten => h,n,OSPFinish(${HANGUPCAUSE})
```

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
```

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

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, please use the asterisk-ss7 or asterisk-dev mailing lists (I monitor them regularly) or email me directly.

Matthew Fredrickson
creslin@digium.com

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.

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.

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
```

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
```

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

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 your 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
```

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.

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)

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.

Using the Hoard Memory Allocator with Asterisk

Using the Hoard Memory Allocator with Asterisk

Install the Hoard Memory Allocator

Download Hoard from http://www.hoard.org/ either via a package or the source tarball.

If downloading the source, unpack the tarball and follow the instructions in the README file to build libhoard for your platform.

Configure asterisk

Run ./configure in the root of the asterisk source directory, passing the **--with-hoard** option specifying the location of the libhoard shared library.

For example:

```
./configure --with-hoard=/usr/src/hoard-371/src/
```

Note that we don't specify the full path to libhoard.so, just the directory where it resides.

Enable Hoard in menuselect

Run 'make menuselect' in the root of the asterisk source distribution. Under 'Compiler Flags' select the 'USE_HOARD_ALLOCATOR' option. If the option is not available (shows up with XXX next to it) this means that configure was not able to find libhoard.so. Check that the path you passed to the **--with-hoard** option is correct. Re-run **./configure** with the correct option and then repeat this step.

Make and install asterisk

Run the standard build commands:

```
# make
# make install
```

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, ...)

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!

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
- Context Includes
- Loops
- Variables
- Applications
- Macros/GoSub
- Goto

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.

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 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 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 or additional files.

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.

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 will immediately update the caller with a Re-INVITE.

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.

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}.

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}.

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
```

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.

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. This means that PLC cannot be used if the codecs on either side of the bridge are the same or do not require a translation to slin in order to translate between them. For instance, a ulaw - ulaw call will not use PLC since no translation is required. In addition, a ulaw - alaw call will also not use PLC since the translation path does not include any step involving slin. 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.

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. Enable (and potentially force) jitter buffers on channels
3. Two channels must be bridged together for PLC to be used (no Meetme or one-legged calls)
4. 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.

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>

${PP_EACH_USER(<item><fn>%{DISPLAY_NAME}</fn><ct>%{CALLERID}</ct><bw>1</bw></item>|${M
AC})}
  </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 Astersik. From the CLI, type "http show status". An example output:

```
HTTP Server Status:
Prefix: /asterisk
Server Enabled and Bound to 192.168.1.1:8088

Enabled URI's:
/asterisk/httpstatus => Asterisk HTTP General Status
/asterisk/phoneprov/... => Asterisk HTTP Phone Provisioning Tool
/asterisk/manager => HTML Manager Event Interface
/asterisk/rawman => Raw HTTP Manager Event Interface
/asterisk/static/... => Asterisk HTTP Static Delivery
/asterisk/mxml => XML Manager Event Interface
Enabled Redirects:
 None.
POST mappings:
 None.
```

There should be a phoneprov URI listed. Next, from the CLI, type "phoneprov show routes" and verify that the information there is correct. An example output for Polycom phones woud look like:

```
Static routes

Relative URI Physical location
sip.ver configs/sip.ver
sip.ld configs/sip.ld
bootrom.ver configs/bootrom.ver
sip.cfg configs/sip.cfg
bootrom.ld configs/bootrom.ld
custom.cfg configs/custom.cfg
Dynamic routes
Relative URI Template
deadbeef4dad.cfg 000000000000.cfg
deadbeef4dad-directory.xml
000000000000-directory.xml
deadbeef4dad-phone.cfg
000000000000-phone.cfg
config/deadbeef4dad polycom.xml
```

With the above examples, the phones would be pointed to:

http://192.168.1.1:8080/asterisk/phoneprov

for pulling config files.

Templates would all be placed in AST_DATA_DIR/phoneprov and static files would be placed in AST_DATA_DIR/phoneprov/configs. Examples of valid URIs would be:

- http://192.168.1.1:8080/asterisk/phoneprov/sip.cfg
- http://192.168.1.1:8080/asterisk/phoneprov/deadbeef4dad.cfg
- http://192.168.1.1:8080/asterisk/phoneprov/config/deadbeef4dad

Reference Information Introduction

Introductory section to the Asterisk Reference Manual. Read this section first.

License Information

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

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Digium, Inc.
445 Jan Davis Drive NW
Huntsville, AL 35806
United States

Hold Music License

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.

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

Telephony Hardware

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

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

DAHDI compatible hardware

Digium, Inc (Primary Developer of Asterisk) DAHDI compatible hardware

Analog Interfaces

TDM410 and AEX410 - The TDM410P and AEX410 are modular four-port half-length PCI 2.2 and PCI-Express x1, respectively, cards that supports FXS and FXO station interfaces for connecting analog telephones and analog POTS lines through a PC.

TDM800 and AEX800 - The TDM800 and AEX800 are modular eight-port half-length PCI 2.2 and PCI-Express x1, respectively, cards that support FXS and FXO station interfaces for connecting analog telephones and analog POTS lines through a PC.

TDM2400 and AEX2400 - The TDM2400 and AEX2400 are modular twenty-four port full-length PCI 2.2 and PCI-Express x1, respectively, cards that support FXS and FXO station interfaces for connecting analog telephones and analog POTS lines through a PC.

Digital Interfaces

TE122 and TE121 - The TE122 and TE122 are half-length, half-height PCI 2.2 and PCI-Express x1, respectively, single-span E1/T1/J1 PRI/PRA software-configurable interfaces for terminating digital telephony through a PC.
TE205/7, TE210/12 and TE220 - The TE205, TE210 and TE220 are half-length PCI 2.2 5.0V, PCI 2.2 3.3V, and PCI-Express x1 dual-span E1/T1/J1 PRI/PRA software-configurable interfaces for terminating digital telephony through a PC.
TE405/7, TE410/12 and TE420 - The TE405/7, TE410/12 and TE420 are half-length PCI 2.2 5.0V, PCI 2.2 3.3V, and PCI-Express x1 quad-span E1/T1/J1 PRI/PRA software-configurable interfaces for terminating digital telephony through a PC.
HA8 and HB8 - The HA8 and HB8 are half-length PCI2.2 and PCI-Express x1, respectively, modular eight-port EuroISDN BRI and Analog FXS/FXO interfaces for terminating telephony through a PC.
B410P - The B410P is a half-length PCI-2.2 fixed four-port EuroISDN BRI interface for terminating digital telephony through a PC.
Packet Interfaces

TC400 and TCE400 - The TC400 and TCE400 are half-length, half-height PCI 2.2 and PCI-Express x1, respectively, codec translation cards for DSP-based translations of patent-encumbered codecs such as G.729a and G.723.1

Non-DAHDI compatible hardware

Non-DAHDI compatible hardware

QuickNet, Inc. http://www.quicknet.net

Internet PhoneJack - Single FXS interface. Supports Linux telephony interface. DSP compression built-in.

Internet LineJack - Single FXS or FXO interface. Supports Linux telephony interface.

mISDN compatible hardware

mISDN compatible hardware

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

- **Digium, Inc.**
  **B410P - also compatible with mISDN
- **beroNet** http://www.beronet.com
  **BN4S0 - 4 Port BRI card (TE/NT)
  **BN8S0 - 8 Port BRI card (TE/NT)
- **Billion Card - Single Port BRI card (TE (/NT with crossed cable))**

Miscellaneous other interfaces

Miscellaneous other interfaces

ALSA http://www.alsa-project.org
Any ALSA compatible full-duplex sound card
OSS http://www.opensound.com

Any OSS compatible full-duplex sound card

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

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.

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.

The Asterisk SIP configuration

Now, let's configure Asterisk to use TLS.

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.

Configuring a TLS-enabled SIP peer within Asterisk

Next, you'll need to configure a SIP peer within Asterisk to use TLS as a transport type. Here's an example:

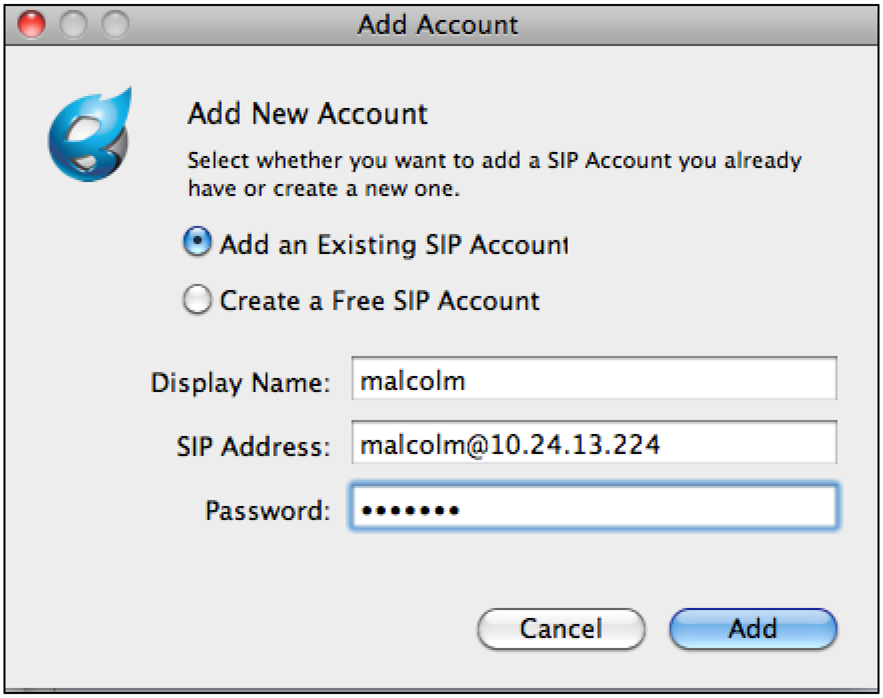
```
[malcolm]
type=peer
secret=malcolm ;note that this is NOT a secure password
host=dynamic
context=local
dtmfmode=rfc2833
disallow=all
allow=g722
transport=tls
context=local
```

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.

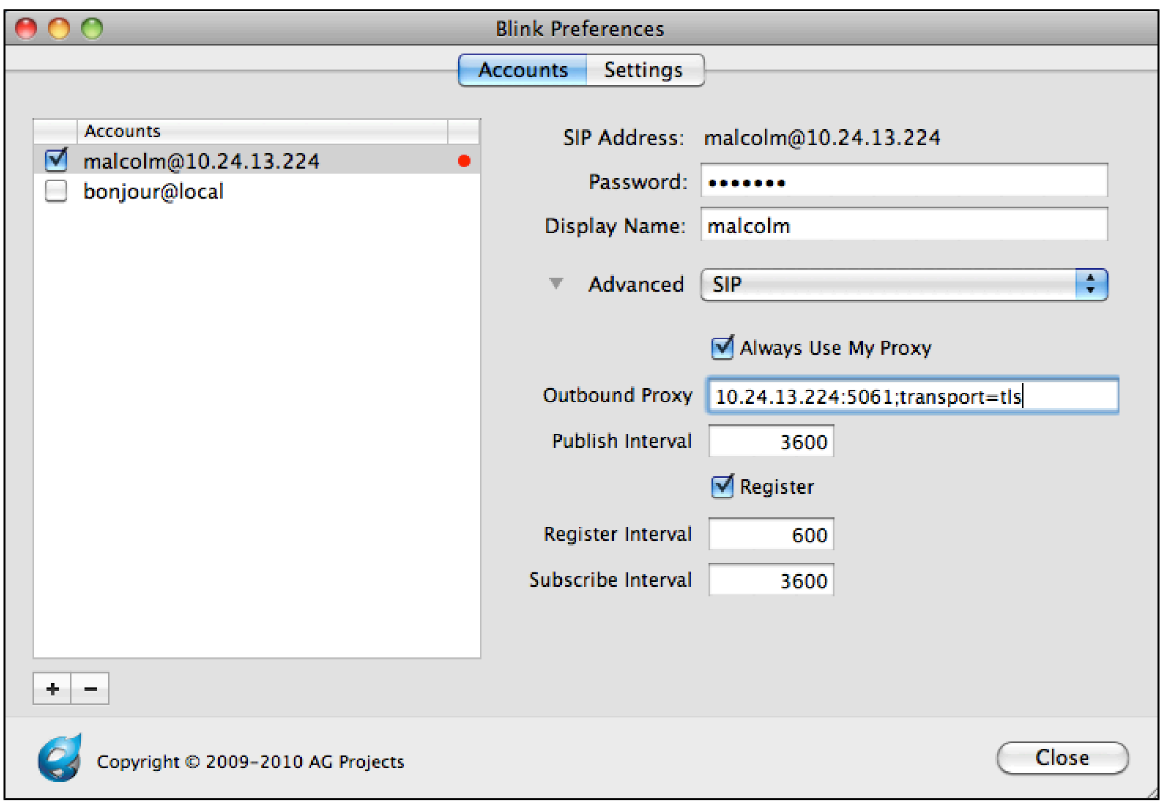
Configuring a TLS-enabled SIP client to talk to Asterisk

Next, we'll configure Blink.

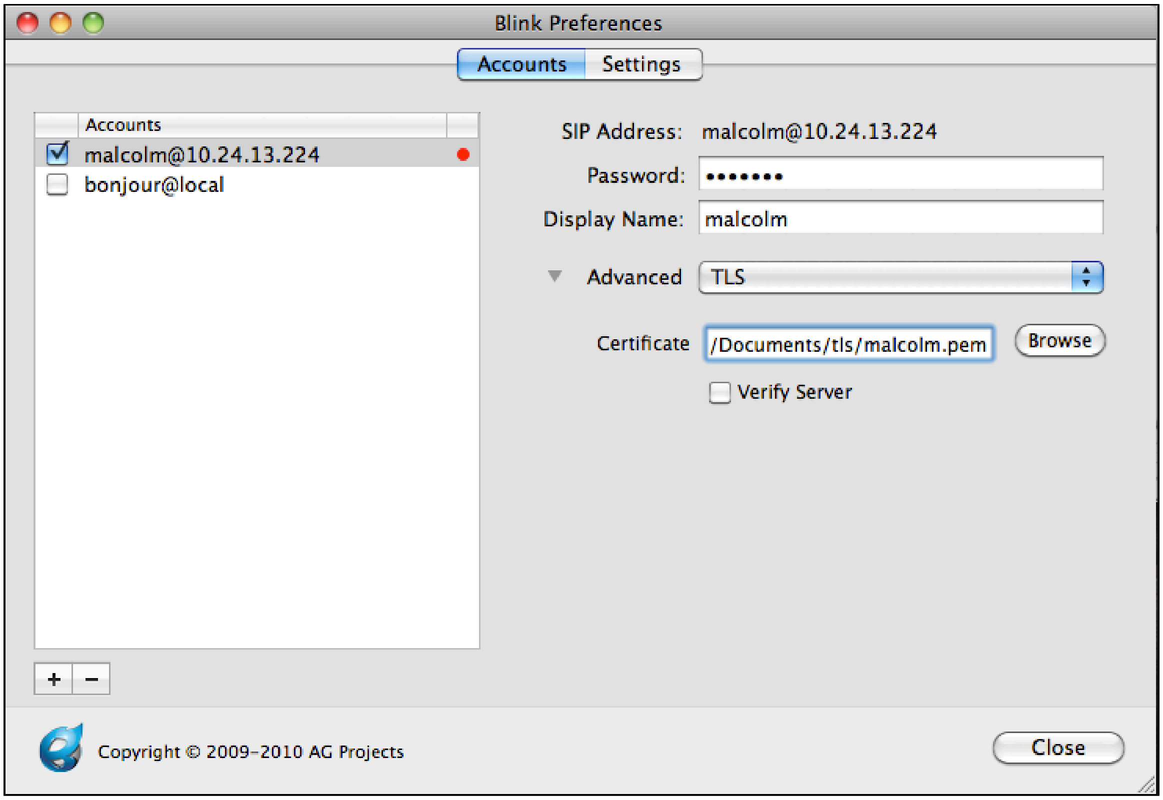
First, let's add a new account.



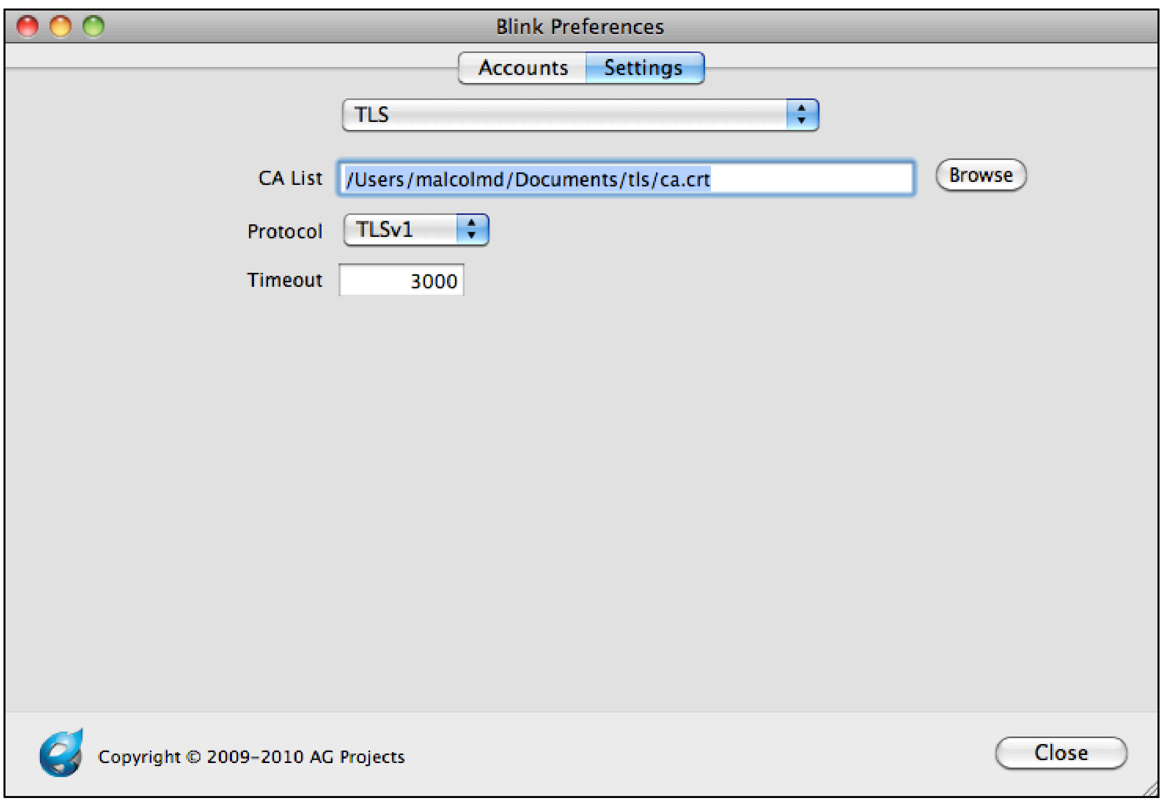
Then, we need to modify the Account Preferences, and under the SIP Settings, we need to set the outbound proxy to connect to the TLS port and transport type on our Asterisk server. In this case, there's an Asterisk server running on port 5061 on host 10.24.13.233.



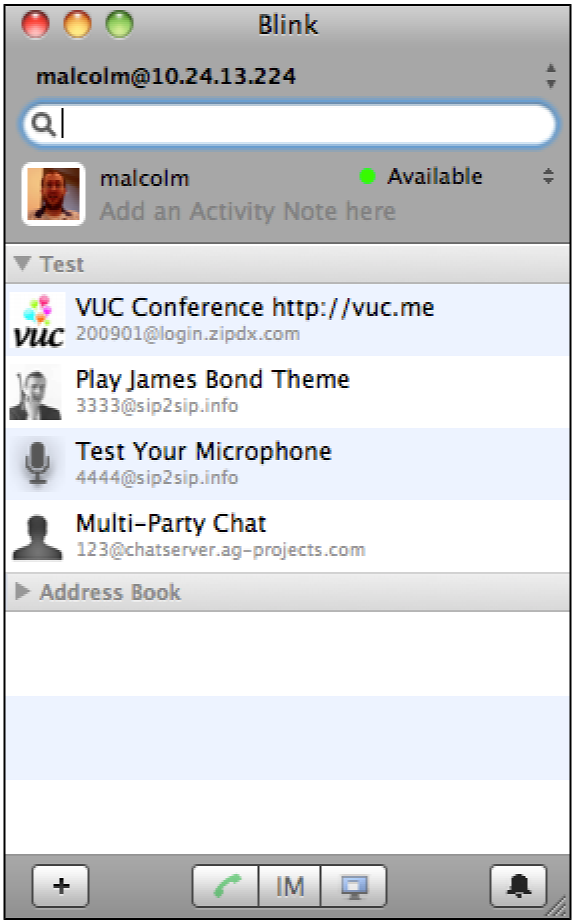
Now, we need to point the TLS account settings to the client certificate (malcolm.pem) that we copied to our computer.



Then, we'll point the TLS server settings to the ca.crt file that we copied to our computer.



Press "close," and you should see Blink having successfully registered to Asterisk.

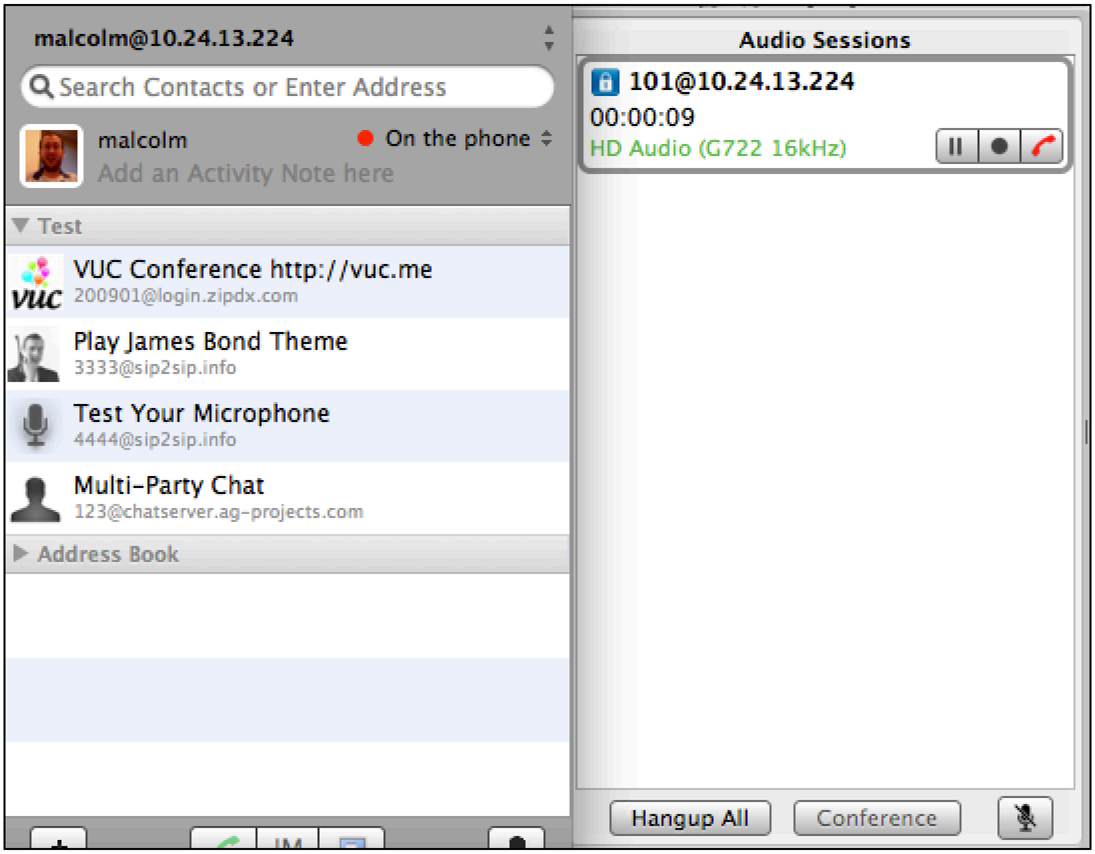


Depending on your Asterisk CLI logging levels, you should see something like:

```
-- 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:



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:

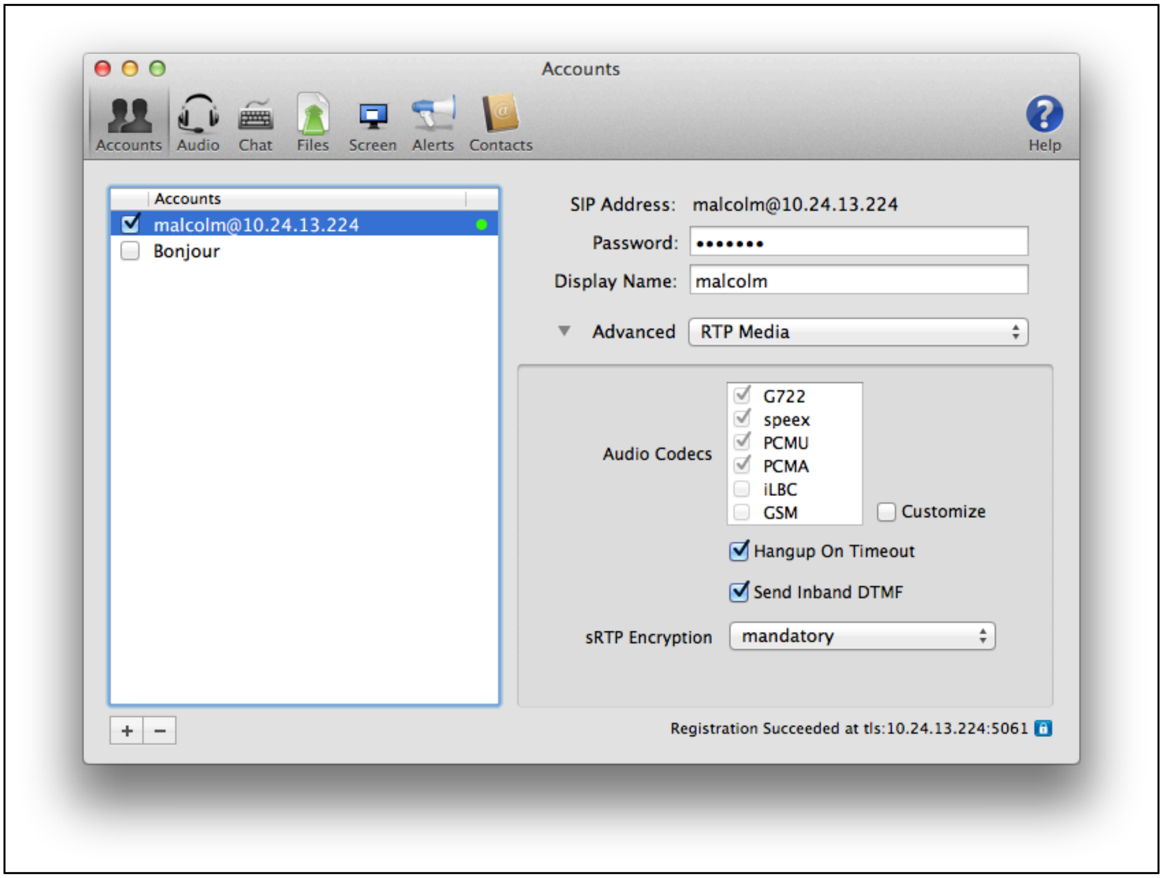
```
[Jan 24 09:29:16] ERROR[10167]: chan_sip.c:27987 setup_srtp: No SRTP module
loaded, can't setup SRTP session.
```

on your Asterisk CLI. If you do see that, install libsrtp (and the development headers), and then reinstall Asterisk (./configure; make; make install).

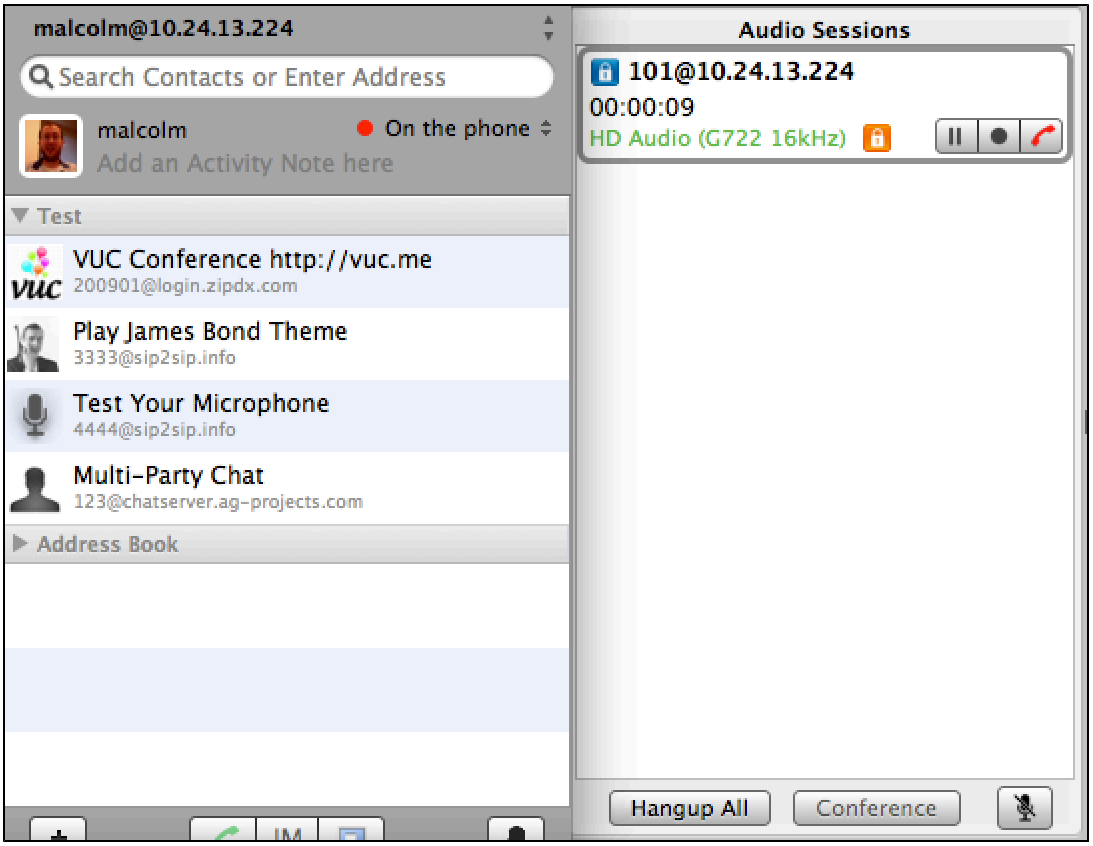
With that complete, let's first go back into our peer definition in **sip.conf.** We're going to add a new encryption line, like:

```
[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:



Reload Asterisk's SIP configuration (sip reload), make a call, and voilà:



We're making secure calls with TLS (signaling) and SRTP (media).

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.

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

Voicemail

All things voicemail

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.

Asterisk SIP Connections

Fishook: We need a single spot to compile the definitive guide to Asterisk's SIP capabilities.

Content coming. WIP beginning here:
1.8 SIP Scratchpad

Asterisk GUI

Introduction to Asterisk GUI

Asterisk GUI is a framework for the creation of graphical interfaces for configuring Asterisk. Some sample graphical interfaces for specific vertical markets are included for reference or for actual use and extension.

Software License

Asterisk GUI HTML and Javascript files Copyright (C) 2006-2011 Digium, Inc.

This program is free software; you can redistribute it and/or modify it under the terms of the GNU General Public License as published by the Free Software Foundation, version 2 only. This software is also available under commercial terms from Digium, Inc.

This program is distributed in the hope that it will be useful, but WITHOUT ANY WARRANTY; without even the implied warranty of MERCHANTABILITY or FITNESS FOR A PARTICULAR PURPOSE. See the GNU General Public License for more details.

Please contact Digium for information on alternative licensing arrangements for Asterisk GUI.

Download

While package release is inconsistent and infrequent, you can always get a current copy of Asterisk GUI from subversion. The current stable version will always be under `branches` and is currently located in `branches/2.0`.

Support

Please note that Asterisk GUI is not officially supported, though bugs, patches, and feature requests may be submitted at http://issues.asterisk.org and should reference the Asterisk GUI project. You may also find peer support in the #asterisk-gui IRC channel and the Asterisk GUI forum.

Installation and Configuration

Installation

```
$ ./configure
$ make
$ make install
$ make checkconfig
```

Configuration

You may install sample configuration files by doing "make samples". Also you will need to edit your Asterisk configuration files to enable Asterisk GUI properly, specifically:

http.conf:

Enable http.

```
[general]
enabled=yes
enablestatic=yes
#bindaddr=0.0.0.0 # allow GUI to be accessible from all IP addresses.
bindaddr=127.0.0.1 # require access from the machine Asterisk is running on
bindport=8088
```

manager.conf

Enable manager access.

```
[general]
enabled = yes
webenabled = yes
```

Create an appropriate entry in `manager.conf` for the administrative user (PLEASE READ THE security.txt FILE!)

```
[admin]
secret = thiswouldbeaninsecurepassword
read = system,call,log,verbose,command,agent,config,read,write,originate
write = system,call,log,verbose,command,agent,config,read,write,originate
```

Access

Access Asterisk GUI via a URL formatted in the following way, where $IP is the IP address on which both Asterisk and Asterisk GUI are installed, $PORT is `bindport` from `http.conf`, and $PREFIX is the `prefix` from `http.conf`, and it can be omitted if blank.

```
http://$IP:$PORT/$PREFIX/static/config/index.html
```

Troubleshooting

Check your filesystem permissions:

```
$ chown -R asterisk:asterisk /etc/asterisk/ /var/lib/asterisk /usr/share/asterisk # if
asterisk runs as the user "asterisk"
$ chmod 644 /etc/asterisk/*
```

Check that the bindaddr value in `/etc/asterisk/http.conf` matches the IP address of the machine you're using to **access** Asterisk GUI, not necessarily the IP address Asterisk GUI is running on.

Check on the Asterisk CLI that Asterisk is receiving the values you've set.

```
http show status
manager show settings
```

Output should look like this:

```
amelia*CLI> http show status
HTTP Server Status:
Prefix: /asterisk
Server Enabled and Bound to 0.0.0.0:8088

Enabled URI's:
/asterisk/httpstatus => Asterisk HTTP General Status
/asterisk/phoneprov/... => Asterisk HTTP Phone Provisioning Tool
/asterisk/amanager => HTML Manager Event Interface w/Digest authentication
/asterisk/arawman => Raw HTTP Manager Event Interface w/Digest authentication
/asterisk/manager => HTML Manager Event Interface
/asterisk/rawman => Raw HTTP Manager Event Interface
/asterisk/static/... => Asterisk HTTP Static Delivery
/asterisk/amxml => XML Manager Event Interface w/Digest authentication
/asterisk/mxml => XML Manager Event Interface

Enabled Redirects:
  None.
amelia*CLI> manager show settings

Global Settings:
----------------
  Manager (AMI):             Yes
  Web Manager (AMI/HTTP):    Yes
  TCP Bindaddress:           0.0.0.0:5038
  HTTP Timeout (minutes):    60
  TLS Enable:                No
  TLS Bindaddress:           Disabled
  TLS Certfile:              asterisk.pem
  TLS Privatekey:
  TLS Cipher:
  Allow multiple login:      Yes
  Display connects:          Yes
  Timestamp events:          No
  Channel vars:
  Debug:                     No
  Block sockets:             No
```

Check that the ports you've specified are open by using telnet from another computer.

```
telnet $ASTERISK_SERVER_IP_ADDRESS 5038
telnet $ASTERISK_SERVER_IP_ADDRESS 8088
```

Check that the `dahdi_genconf` script runs correctly and creates `/etc/asterisk/dahdi_guiread.conf`.

Check the existence of `/etc/asterisk/guipreferences.conf` and inside that file, the existence of the following line:

```
config_upgraded = yes
```

Check the last modified date of `/etc/asterisk/http.conf`. Asterisk GUI updates the timestamp on this file every time it is loaded. If the timestamp is not getting updated, your HTTP request is either not making it to Asterisk or it is not being processed correctly by Asterisk. This indicates a configuration error.

Check that the user Asterisk runs as has a login shell. Asterisk GUI depends on Asterisk being able to use the System application.

```
su - asterisk
```

If you installed Asterisk GUI via 'yum' or 'apt-get', you may need to symlink `/var/lib/asterisk/static-http` to `/usr/share/asterisk/static-http`. Unfortunately `/usr/share/asterisk/static-http` will already exist, but fortunately it does not contain any useful files.

```
ls /var/lib/asterisk/static-http/config && rm -rf /usr/share/asterisk/static-http &&
ln -s /var/lib/asterisk/static-http /usr/share/asterisk/static-http
```

Development

Debugging

To turn on debug messages, open `config/js/session.js`. On line 30, set "log" to "true":

```
log: true, /**< boolean toggling logging */
```

Hide Menu Categories

Hide menu categories by changing the HTML class attribute to "AdvancedMode" in index.html. Show them by enabling Advanced Options in the GUI.

Historical Pages

What is a Historical Page?

A Historical Page is old documentation that no longer applies to the current version of Asterisk. Ways of doing things have changed, perhaps, or a particular bit of functionality is no longer available / supported / recommended.

Jabber in Asterisk

> This information applies to Asterisk 10 and earlier versions. XMPP Support for Asterisk 11 has been completely rewritten

XMPP (Jabber) is an XML based protocol primarily for presence and messaging. It is an open standard and there are several open server implementations, such as ejabberd, jabberd(2), openfire, and others, as well as several open source clients, Psi, gajim, gaim etc. XMPP differs from other IM applications as it is immensely extendable. This allows us to easily integrate Asterisk with XMPP. The Asterisk XMPP Interface is provided by res_jabber.so.

`res_jabber` allows for Asterisk to connect to any XMPP (Jabber) server and is also used to provide the connection interface for `chan_jingle` and `chan_gtalk`.

Functions (`JABBER_STATUS`, `JABBER_RECEIVE`) and applications (`JabberSend`) are exposed to the dialplan.

You'll find examples of how to use these functions/applications hereafter. We assume that 'asterisk-xmpp' is properly configured in `jabber.conf`.

JabberSend

`JabberSend` sends an XMPP message to a buddy. Example:

extensions.ael

```
context default {
 _XXXX => {
       JabberSend(asterisk-xmpp,buddy@gmail.com,${CALLERID(name)} is calling
${EXTEN});
       Dial(SIP/${EXTEN}, 30);
       Hangup();
 }
}
```

JABBER_STATUS

> As of Asterisk 1.6.0, the corresponding application `JabberStatus` is still available, but marked as deprecated in favor of this function.

`JABBER_STATUS` stores the status of a buddy in a dialplan variable for further use. Here is an AEL example of how to use it:

extensions.ael

```
1234 => {
 Set(STATUS=${JABBER_STATUS(asterisk-xmpp,buddy@gmail.com)});
 if (${STATUS}=1) {
  NoOp(User is online and active, ring his Gtalk client.);
  Dial(Gtalk/asterisk-xmpp/buddy@gmail.com);
 } else {
  NoOp(Prefer the SIP phone);
  Dial(SIP/1234);
 }
}
```

JABBER_RECEIVE

`JABBER_RECEIVE` waits (up to X seconds) for a XMPP message and returns its content. Used along with `JabberSend` (or `SendText`, provided it's implemented in the corresponding channel type), `JABBER_RECEIVE` helps Asterisk interact with users while calls flow through the dialplan.

`JABBER_RECEIVE`/`JabberSend` are not tied to the XMPP media modules `chan_gtalk` and `chan_jingle`, and can be used anywhere in the dialplan. In the following example, calls targeted to extension 1234 (be it accessed from SIP, DAHDI or whatever channel type) are controlled by user bob@domain.com. Asterisk notifies him that a call is coming, and asks him to take an action. This dialog takes place over an XMPP chat.

extensions.ael

```
context from-ext {
 1234 => {
  Answer();
  JabberSend(asterisk-xmpp,bob@jabber.org,Call from $CALLERID(num) - choose an option
to process the call);
  JabberSend(asterisk-xmpp,bob@jabber.org,1 : forward to cellphone);
  JabberSend(asterisk-xmpp,bob@jabber.org,2 : forward to work phone);
  JabberSend(asterisk-xmpp,bob@jabber.org,Default action : forward to your voicemail);
  Set(OPTION=${JABBER_RECEIVE(asterisk-xmpp,bob@jabber.org,20)});
  switch (${OPTION}) {
   case 1:
    JabberSend(asterisk-xmpp,bob@jabber.org,(Calling cellphone...);
    Dial(SIP/987654321);
    break;
   case 2:
    JabberSend(asterisk-xmpp,bob@jabber.org,(Calling workphone...);
    Dial(SIP/${EXTEN});
    break;
   default:
    Voicemail(${EXTEN}|u)
  }
 }
}
```

When calling from a GoogleTalk or Jingle client, the `CALLERID(name)` is set to the XMPP id of the caller (i.e. his JID). In the following example, Asterisk chats back with the caller identified by the caller id. We also take advantage of the `SendText` implementation in `chan_gtalk` (available in `chan_jingle`, and `chan_sip` as well), to allow the caller to establish SIP calls from his GoogleTalk client:

extensions.ael

```
context gtalk-in {
 s => {
   NoOp(Caller id : ${CALLERID(all)});
   Answer();
   SendText(Please enter the number you wish to call);
   Set(NEWEXTEN=${JABBER_RECEIVE(asterisk-xmpp,${CALLERID(name)})});
   SendText(Calling ${NEWEXTEN} ...);
   Dial(SIP/${NEWEXTEN);
   Hangup();
 }
}
```

The maintainer of `res_jabber` is Philippe Sultan <philippe.sultan@gmail.com>.

Old Calling using Google

Note that Google's changes to their Google Voice system have rendered the functionality of chan_gtalk and res_jabber unreliable. Additionally, ongoing maintenance of of chan_gtalk and res_jabber for Asterisk versions prior to Asterisk 11 is not provided by Digium and is instead in the charge of the community. For more information, please see the current page Calling using Google, where things have changed, as well as 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_gtalk Channel Driver and the res_jabber Resource module. Before proceeding, please ensure that both are compiled and part of your installation. Compilation of res_jabber and chan_gtalk 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 1.8.1.1 or greater. The 1.6.x versions of Asterisk only support calls made using the legacy GoogleTalk external client.

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.

Gtalk configuration

The chan_gtalk Channel Driver is configured with the gtalk.conf configuration file, typically located in /etc/asterisk. What follows is the minimum required configuration for successful operation.

Minimum Gtalk Configuration

```
[general]
context=local
allowguest=yes
bindaddr=0.0.0.0
externip=216.208.246.1

[guest]
disallow=all
allow=ulaw
context=local
connection=asterisk
```

This general section of this configuration specifies several items.

1. That calls will terminate to or originate from the **local** context; context=local
2. That guest calls are allowd; allowguest=yes
3. That RTP sessions will be bound to a local address (an IPv4 address must be present); bindaddr=0.0.0.0
4. (optional) That your external (the one outside of your NAT) IP address is defined; externip=216.208.246.1

The guest section of this configuration specifies several items.

1. That no codecs are allowed; disallow=all
2. That the ulaw codec is explicitly allowed; allow=ulaw
3. That calls received by the guest user will be terminated into the **local** context; context=local
4. That the Jabber connection used for guest calls is called "asterisk;" connection=asterisk

Jabber Configuration

The res_jabber Resource is configured with the jabber.conf configuration file, typically located in /etc/asterisk. What follows is the minimum required configuration for successful operation.

Minimum Jabber Configuration

```
[general]
autoregister=yes

[asterisk]
type=client
serverhost=talk.google.com
username=your_google_username@gmail.com/Talk
secret=your_google_password
port=5222
usetls=yes
usesasl=yes
statusmessage="I am an Asterisk Server"
timeout=100
```

The general section of this configuration specifies several items.

1. Debug mode is enabled, so that XMPP messages can be seen on the Asterisk CLI; debug=yes
2. Automated buddy pruning is disabled, otherwise buddies will be automatically removed from your list; autoprune=no
3. Automated registration of users from the buddy list is enabled; autoregister=yes

The asterisk 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/resource, where our resource is "Talk;" username=your_google_username@gmail.com/Talk
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. TLS encryption is required by Google; usetls=yes
7. Simple Authentication and Security Layer (SASL) is used by Google; usesasl=yes
8. We set a status message so other Google chat users can see that we're an Asterisk server; statusmessage="I am an Asterisk Server"
9. We set a timeout for receiving message from Google that allows for plenty of time in the event of network delay; timeout=100

Phone configuration

Now, let's place a phone into the same context as the Google calls. 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:

```
exten => s,1,Answer()
exten => s,n,Wait(2)
exten => s,n,SendDTMF(1)
exten => s,n,Dial(SIP/malcolm,20)
```

Note that you might have to adjust the "Wait" time from 2 (in seconds) to something larger, like 8, depending on the current mood of Google. Otherwise, your incoming calls might not be successfully picked up.

This example uses the "s" unmatched extension, because Google does not forward any DID when it sends the call to Asterisk.

In this example, we're Answering the call, Waiting 2 seconds, 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 the D dial option. 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. 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,Set(crazygooglecid=${CALLERID(name)})
exten => s,n,Set(stripcrazysuffix=${CUT(crazygooglecid,@,1)})
exten => s,n,Set(CALLERID(all)=${stripcrazysuffix})
exten => s,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(gtalk/asterisk/mybuddy@gmail.com)
```

Where the technology is "gtalk," the dialing peer is "asterisk" as defined in jabber.conf, and the dial string is the Google account name.

Outgoing calls made to Google Voice take the form of:

```
exten => _1XXXXXXXXXX,1,Dial(gtalk/asterisk/+${EXTEN}@voice.google.com)
```

Where the technology is "gtalk," the dialing peer is "asterisk" as defined in jabber.conf, and the dial string is a full E.164 number (plus character followed by country code, followed by the rest of the digits).

Interactive Voice with Text Response (IVTR)

Because the Google Talk web client provides both audio and text interface, you can use it to provide a text-based way of traversing Interactive Voice Response (IVR) menus. This is necessary since the client does not have any DTMF inputs.

In the following dialplan example, we will answer the call, wait a bit, send some text that will show up in the caller's Google Talk client, play back a prompt, capture the caller's text-based response, and then dial the appropriate SIP device.

```
exten => s,1,Answer()
exten => s,n,SendText("If you know the extension of the party you wish to reach, dial
it now.")
exten => s,n,Background(if-u-know-ext-dial)
exten => s,n,Set(OPTION=${JABBER_RECEIVE(asterisk,${CALLERID(name)::15},5)})
exten => s,n,Dial(SIP/${OPTION},20)
```

Note that with the JABBER_RECEIVE function, we're receiving the text from **asterisk** which we defined earlier in this page as our connection to Google. We're also specifying with **${CALLERID(name)::15}** that we want to strip off the last 15 characters from the CallerID name string - which is the number of characters that Google is appending, as of this writing, to represent an internal call ID number, and that we want to wait **5** seconds for a response.

Webinar

A Webinar was conducted on Tuesday, March 24, 2011 detailing Asterisk configuration for calling using Google as well as several usage cases. A copy of the slides, in PDF format, is available here - Google Calling Webinar - Public.pdf

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.

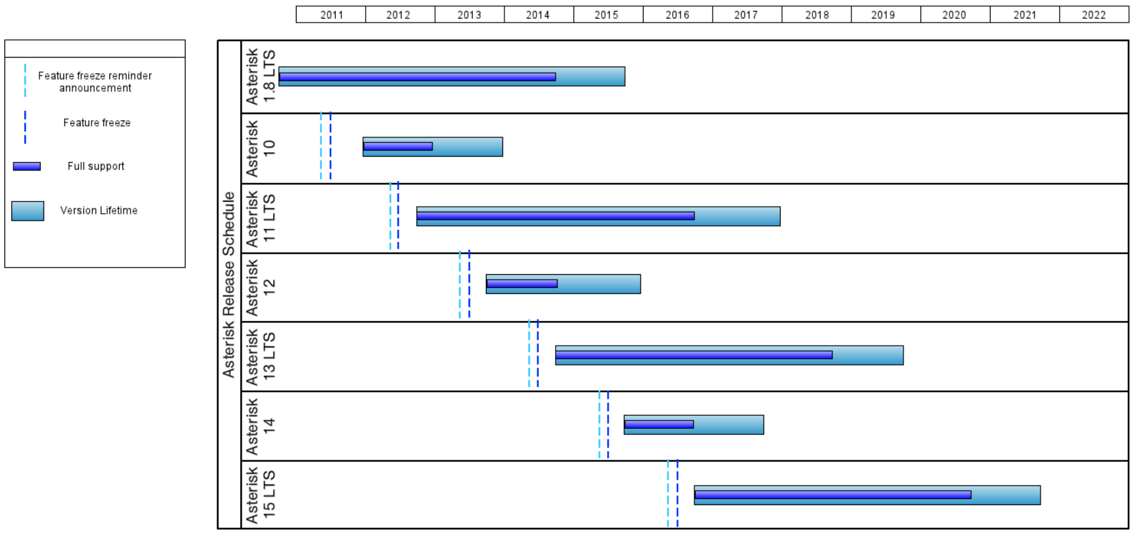
The following table shows the release time lines for all releases of Asterisk, including those that have reached End of Life.

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	2013-10 (tentative)	2014-10 (tentative)	2015-10 (tentative)
13.x	LTS	2014-10 (tentative)	2018-10 (tentative)	2019-10 (tentative)

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 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

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. Issues reported against these modules may have a low level of support. Some extended modules have active community developers monitoring issues though.

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

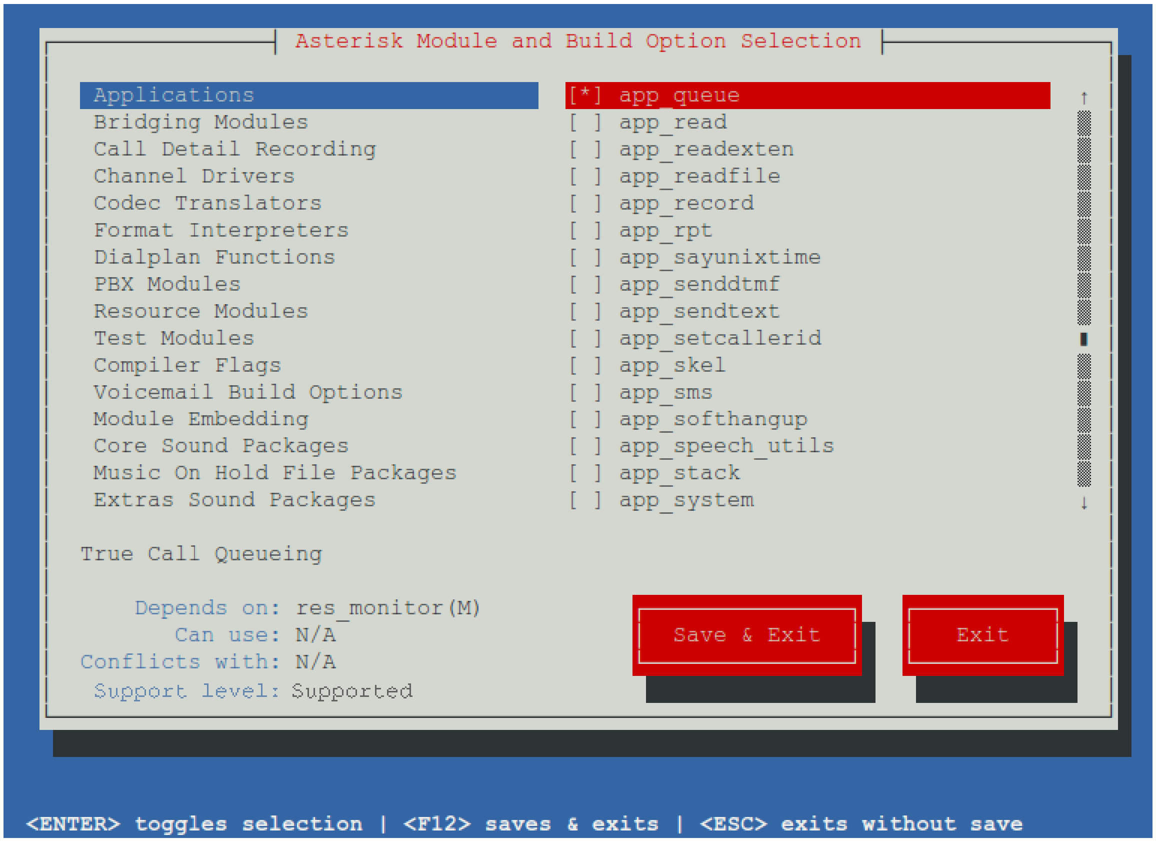
The following two images show the suggested menuselect output based on the addition of the *<support_level>* and *<replacement>* tags. This would be a new line that has not been used before, and therefore would be added to menuselect as that new line.

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**).

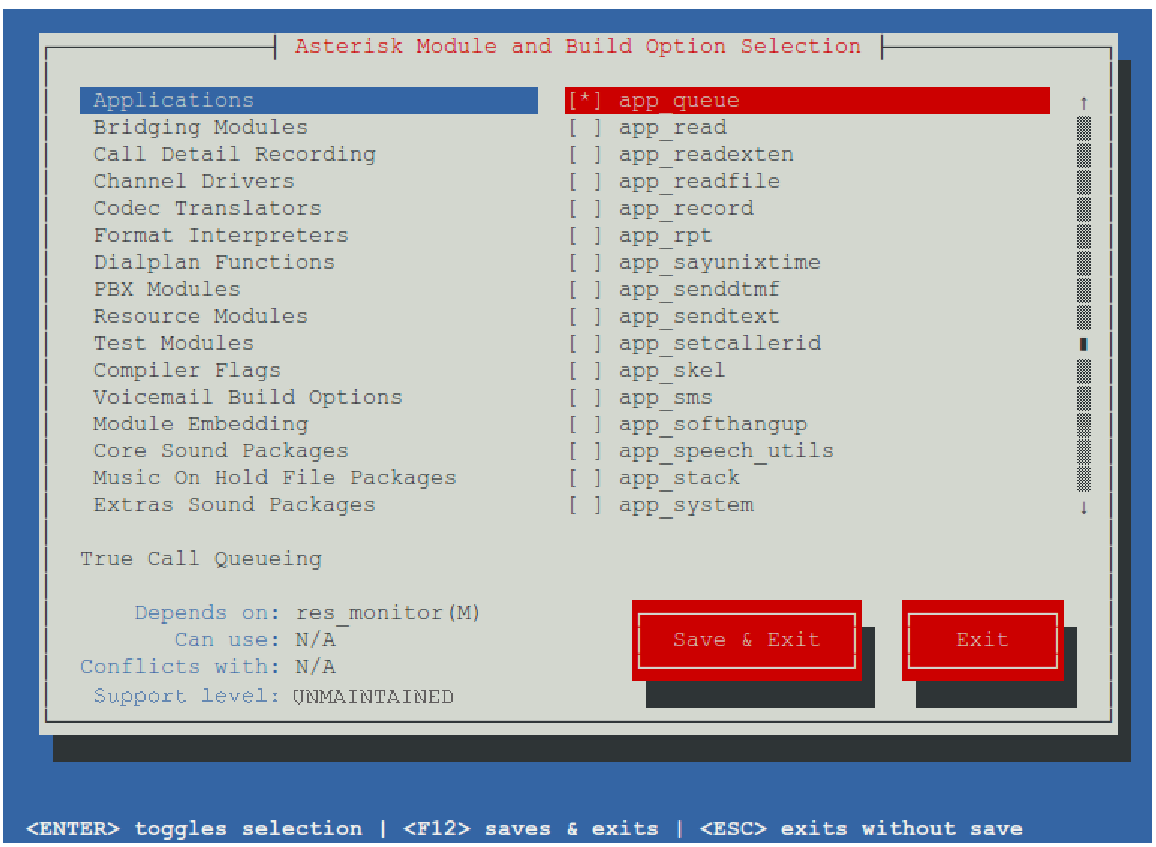
If the **<support_level>** tag is used, then the value of *extended* would cause the additional text of ** EXTENDED ** to be displayed.

```
     ************************************************
         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 ***
```

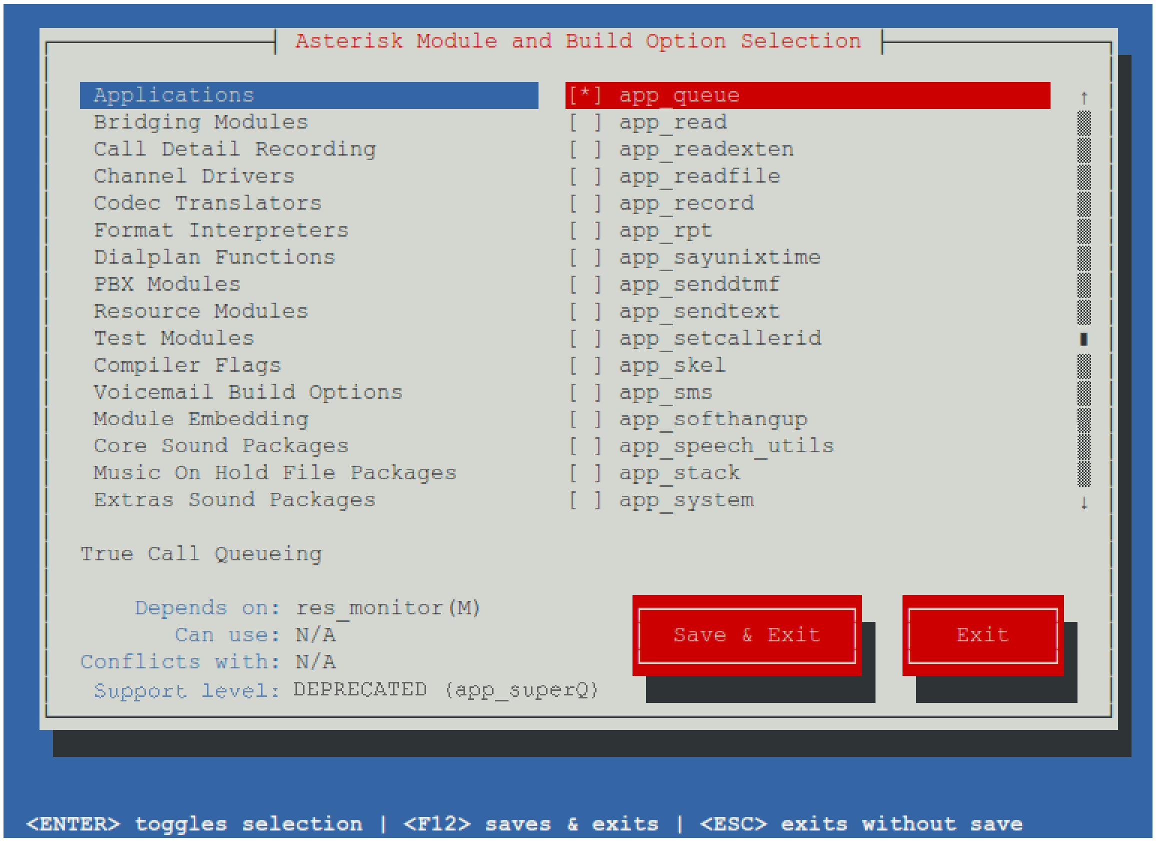
Display in menuselect-newt for supported modules. If no <support_level> is specified, then it is assumed the module is supported:



Display in menuselect-newt for extended modules:



Display in menuselect-newt for deprecated modules:



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 not accepted through the issue tracker and in general 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 or IRC channels 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 http://www.asterisk.org/community
- **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
- **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

Why should you read this?

To get the attention of helpful individuals and the Asterisk development team, you'll want to read, understand and act on this article. The steps here will help you provide all the information the Asterisk team needs to solve your issue.

How to report a bug

STOP! BEFORE YOU EVEN THINK ABOUT FILING A BUG REPORT...

Your issue may not be a bug or could have been fixed already. Run through the checklist below to verify you have done your due diligence.

- **Verify you are on a supported version of Asterisk:** https://wiki.asterisk.org/wiki/display/AST/Asterisk+Versions, i.e. a version of Asterisk that has not reached End of Life
- **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 point release. http://svnview.digium.com/svn/asterisk/tags/
- **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?** The documentation is located here: https://wiki.asterisk.org/wiki/display/AST/Home
- **Have you searched the Asterisk bug tracker to see if an issue is already filed for this potential bug?** https://issues.asterisk.org/jira you'll find the search field in the top right of the page.
- **Can you reproduce the problem?** If you can't find a way to simulate or reproduce the issue, then at least get the system to a point where it's capturing relevant debug during the times the seemingly random failure occurs. Yes that could mean running debug for days or weeks if necessary.

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.

Submitting the bug report - information requirements

You'll report the issue through the tracker, under the Asterisk project. https://issues.asterisk.org/jira/browse/ASTERISK

0. 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.

1.Create a new issue in the tracker

The "Create Issue" button in the top right hand corner of the page will prompt you for the project and issue type. Pick the Asterisk project, and choose an appropriate issue type.

Issue types:

- Bug
- New feature
- Improvement
- Information request (This type is not currently used, and your issue is likely to be closed or redefined)

2. 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'll also need SIP traces for anything involving a SIP channel plus backtraces and valgrind output for crashes or memory issues. Unless the issue is extremely trivial, you'll need to also include an Asterisk debug log, including DEBUG and VERBOSE type messages with at least a level of 5. You can find instructions HERE and HERE.

> 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.

3. 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 above and include all information that we require. Watch for emails where we may ask for additional data and help us test any patches or possible fixes for the issue. If you didn't follow the information requirements - it'll often be the first thing we ask for. A developer **can not** fix your issue until we have sufficient data to verify the problem.

Reasons your report may be closed without resolution

If your bug is determined to be a configuration error or something that is clearly not reproducible, it may be closed immediately by a bug marshal. If you believe this to be in error, please go in and edit the bug and say so, or comment to the bug marshal that made the closure (visible on the ticket itself.) If that fails, bring it up on the mailing list(s) and it will be sorted out by the community.

Common reasons your issue may be closed without resolution:

- The issue can't be reproduced from the original report.
- The report lacks sufficient information to investigate, and you haven't responded to our requests for specific information.
- The issue may exist, but can't be fixed due to underlying architecture or infrastructure issues.
- The issue is not a bug, but is a support request. The reporter will be directed to community resources for support.

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 SVN 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. However just writing a patch doesn't guarantee it'll make it into an Asterisk release. See the paragraphs below on Patch and Code submission.

Do you have a feature request and a patch to go with it?

Great, then move on to the Patch and Code submission section below.

Patch and Code submission

Basic patch requirements:

- Make sure you read the coding guidelines and make your code compliant with these.
- Add comments that explain your code in Doxygen format (Qt style) to make it easier to understand both for those that commit and for other developers and users.
- Read the Digium License Agreement section below, and then sign the agreement within JIRA.

Format of patch: Please use "diff -u" or "svn diff" on all your patches. Patches which include alternate formatting are almost certainly going to be thrown out or ignored; there are too few hours in the day to wade through difficult-to-follow C code fixes without the help of "diff -u".

Features and Improvements

We all love new features, but new features bring new complexities, and new complexities bring new bugs. So, to help cut down on new bugs being introduced with your new features, please consider as much of what you're touching as possible. In other words: everything is linked in very subtle ways; make sure you look very closely at each piece of what your patch influences before you submit it. Don't be afraid to submit revised copies of the patch if that's what's required. Bug marshals are there to help you and their suggestions are based on their experience of working with Asterisk.

Once you have a patch that follows the requirements noted above then you can post it on the issue tracker in the Asterisk project as type "Improvement" or "New Feature".

Patch testing & Review board

After you create a new feature or bug patch submission, it will need to be tested before it can be committed to the SVN repository. By "tested", we mean that multiple persons other than yourself should patch/run/abuse the patch in various manners so that all possible variations of input/output are run through the test. In instances where there are limited test environments, please document exactly how and why you were unable to get others to test the patch so that the developers know that this simply isn't "waiting for testing" and is in fact ready to go. Insufficient testing is a sure way to have your feature put on the back burner - the SVN maintainers don't have time to run exhaustive tests on each new feature. Find people on the #asterisk or #asterisk-dev IRC channels who may be able to assist you in testing, and develop a working relationship with other Asterisk users so that you can swap testing routines; this greatly speeds the process. Mail the mailing lists, informing other Asterisk users about your patch and ask for feedback.

If you have submitted code in the past and are great at following the coding guidelines, then you may want to request access to Review Board. You can find the instructions for review board here. Code submitted to Review Board is likely to be included much faster than an untested patch just sitting in the tracker. The development team will provide access to Review Board for those that submit code often, write quality code and have a history of following the coding guidelines.

Who commits the code into Asterisk?

There are two ways code gets pushed into Asterisk.

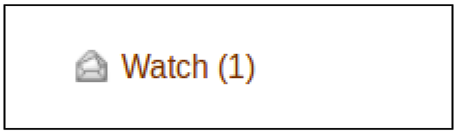
1. Patches can always be merged by a member of the Asterisk Developer Community. Folks with commit access are those who have shown a willingness to work with the review process and are trusted shepherds of the project. Anyone with commit access can take ownership of a proposed patch and work to get it included in trunk - hence why the asterisk-dev mailing list is the best place to discuss patches.
2. Digium works through the queue of all issues on the tracker, both bugs and improvements. There's typically hundreds of issues in the backlog, including the work that the Asterisk Developer Community commits to at AstriDevCon every year. As such, we try to address as much as we can but there's no guarantee that we will get to any one issue.

What is the status of my issue?

The status of all issues in the tracker are reflected in the tracker. For any particular issue you'll want to look at the "Status" field and the comments tab under "Activity".

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.

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 to avoid cluttering up the main project. The clone being created does not indicate any particular status.

All comments, work, and anything of relevance to the patch or issue is done in the ASTERISK project

Why did my feature not make it into Asterisk?

Many times, a feature that might sound good to you and might work as expected just isn't important enough to make it into the system if it isn't deemed useful to a large enough audience. This is, of course, a subjective decision by the SVN maintainers and the bug marshals. Their opinions can be swayed if enough users get together and rally around a particular feature that seems to be useful to them, so getting others to chime in on your neat request will perhaps advance it further towards the top of the queue. Sometimes, it's just a matter of timing - nobody has had the time to look at it yet. It's perfectly legitimate (and good) for someone who has a bug or feature request to go to IRC and lobby for it with the bug marshal to push it through, schedule time, etc.

Digium Submission License Agreement

If you are going to contribute code to the Asterisk project, then please thoroughly read and then electronically sign the Digium Open Source Software Project Submission Agreement upon logging in to the Asterisk issue tracker. You'll see a "Sign a License Agreement" link at the top of the JIRA interface.

The gist of the agreement is that your contribution must not introduce any encumbrance to the Asterisk code base; Digium does not own your contribution, and they cannot take released Asterisk out of GPL. Relax; it's a very fair and reasonable license, and does not remove your rights or threaten the open source nature of the project. See the mailing list archives for long explanations of why everyone who contributes agrees that it's a fair and sane thing to do. You only need to sign the agreement once; it applies to all code that you send in via the issue tracker.

The issue tracker keeps a record of contributions and the license agreements that were in effect when those contributions were uploaded; it will currently restrict you from uploading a patch or documentation without a signed license agreement.

How to properly upload an attachment that requires a Submission License Agreement:

- When using the "Attach Files" form on a particular issue, be sure to select the "Yes, this is a code or documentation contribution" radio button.
- If you don't have a user license agreement, you'll see the message
  " You dont appear to have a current, signed submission license agreement on file. Please sign one before attempting to upload a code or documentation contribution."
- To sign a submission license agreement, use the "Sign a license Agreement" link at the top navigation bar in JIRA.
- The license agreement will be reviewed by Digium's legal department, then you'll be notified by E-mail of acceptance or rejection. If accepted you can now repeat the upload process and you'll be able to upload the code with a proper license associated to your account and showing by the attachment.

If you upload a patch and do not mark it as a code submission, and a bug marshal or developer later determines it to be a code submission, they may have to delete your patch and ask you to re-upload it, properly marking it as a code submission so that your license will be associated.

Get involved in Asterisk Development - influence the future of Asterisk

Reading the guidelines here is a great start! There are plenty of other wiki pages on Asterisk development which include coding guidelines. Feel free to start submitting relevant patches, documentation additions or fixes as soon as you can. There are many minor features and changes that need to be tested, reviewed and possibly expanded or tweaked before they can go into Asterisk.

Start associating with other Asterisk developers where they hang out http://www.asterisk.org/community/. You'll find them on primarily on IRC and the mailing lists. The forums are populated primarily by users.

Once you get to know the community and you really want to influence the Asterisk road-map, you'll want to check out AstriDevCon. AstriDevCon is an event held alongside AstriCon where the developers get together to review what was done in the past year and plan Asterisk's future.

Asterisk Community

- Asterisk Community Services
- Community Services Signup
- IRC
- Mailing Lists

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 maili ng list. If any of the community services are unavailable outside of a scheduled maintenance time, please notify the Asterisk development team.

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.

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-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
asterisknow	AsteriskNOW Discussion
svn-commits	SVN commits to the Digium repositories
Test-results	Results from automated testing

Development List Note

This list is geared toward the development of Asterisk itself. For discussions about developing applications that *use* Asterisk, please see the asterisk-users list.