**Asterisk CLI**

The [Asterisk](http://www.voip-info.org/wiki/view/Asterisk) command line interface (CLI) is reached by using the Linux shell command  
**asterisk -r** or **rasterisk**  
  
If you want debugging output, add one or many **v**:s  
**asterisk -vvvvvr**  
  
The Asterisk server has to be running in the background for the CLI to start.

If you want to run a CLI command in a shell script, use the **x** option  
  
**asterisk -rx "logger reload"**  
  
For help in the CLI mode, use the **core show help** command (formerly **help**). To get help on various applications you can use in the [extensions.conf](http://www.voip-info.org/wiki/view/Asterisk+config+extensions.conf" \o "Asterisk config extensions.conf) config file, use the **core show applications** command (formerly **show applications**).

**General commands**

* **!<command>**: Executes a given shell command
* **abort halt**: Cancel a running halt
* **add extension**: Add new extension into context
* **add ignorepat**: Add new ignore pattern
* **add indication**: Add the given indication to the country
* **agent show**: Show status of [agents](http://www.voip-info.org/wiki/view/Asterisk+Agents)
* **debug channel**: Enable debugging on a channel
* **dont include**: Remove a specified include from context
* **help**: Display help list, or specific help on a command
* **include context**: Include context in other context
* **load**: Load a dynamic module by name
* **logger reload**: Reopen log files. Use after rotating the log files.
* **mixmonitor {start|stop|list}**: Execute a MixMonitor command.
* **no debug channel**: Disable debugging on a channel
* [**originate**](http://www.voip-info.org/wiki/view/Asterisk+cli+originate): originate a call.
* **remove extension**: Remove a specified extension
* **remove ignorepat**: Remove ignore pattern from context
* **remove indication**: Remove the given indication from the country
* **save dialplan**: *Overwrites* your current extensions.conf file with an exported version based on the current state of the dialplan. A backup copy of your old extensions.conf is *not* saved. The initial values of global variables defined in the [globals] category retain their previous initial values; the current values of global variables are not written into the new extensions.conf. (:exclaim:) Using "save dialplan" will result in losing any comments in your current extensions.conf.
* **dialplan save (1.4)**: BROKEN, doesn't parse correctly. *Overwrites* your current extensions.conf file with an exported version based on the current state of the dialplan. A backup copy of your old extensions.conf is *not* saved. The initial values of global variables defined in the [globals] category retain their previous initial values; the current values of global variables are not written into the new extensions.conf. (:exclaim:) Using "save dialplan" will result in losing any comments in your current extensions.conf.
* **set verbose**: Set level of verboseness
* **show applications**: [Shows registered applications](http://www.voip-info.org/wiki/view/Asterisk+-+documentation+of+application+commands)
* **show application**: Describe a specific application
* **show channel**: Display information on a specific channel
* [**show channels**](http://www.voip-info.org/wiki/view/show+channels): Display information on channels
* [**show codecs**](http://www.voip-info.org/wiki/view/Asterisk+cli+show+codecs): Display information on codecs
* **show conferences**: Show status of conferences
* **show dialplan**: Show dialplan
* [**show hints**](http://www.voip-info.org/wiki/view/Asterisk+cli+show+hints): Show registered hints
* **show image formats**: Displays image formats
* **show indications**: Show a list of all [country/indications](http://www.voip-info.org/wiki/view/Asterisk+config+indications.conf)
* **show locals**: Show status of local channels
* **show manager command**: Show manager commands
* **show manager connect**: Show connected manager users
* **show parkedcalls**: Lists parked calls
* **show queues**: Show status of queues, see details [here](http://www.voip-info.org/wiki/view/asterisk+cli+command+show+queue)
* **show switches**: Show alternative switches
* **show translation**: Display translation matrix
* **soft hangup**: Request a hangup on a given channel - in Asterisk 1.6.2: "channel request hangup <name>"
* **show voicemail users**: List defined voicemail boxes
* **show voicemail zones**: List zone message formats
* [**devstate change**](http://www.voip-info.org/wiki/view/Asterisk+cli+devstate+change): Change state of a custom device (new in Asterisk 1.6.0)

**Server management**

* **restart gracefully**: Restart Asterisk gracefully, i.e. stop receiving new calls and restart at empty call volume
* **restart now**: Restart Asterisk immediately
* **restart when convenient**: Restart Asterisk at empty call volume

**Note for Asterisk 1.2:** Restart now is like a reload, not a real restart it just run the reload routines (thus open ports are not closed). Often you don't need really need to restart asterisk, instead just need to issue e.g. 'unload chan\_sip.so' and 'load chan\_sip.so'.

* **reload**: Reload configuration
* **stop gracefully**: Gracefully shut down Asterisk, i.e. stop receiving new calls and shut down at empty call volume
* **stop now**: Shut down Asterisk imediately
* **stop when convenient**: Shut down Asterisk at empty call volume
* **dialplan reload**: Reload extensions and *only* extensions (formerly extensions reload)
* **unload**: Unload a dynamic module by name
* **show modules**: List [modules](http://www.voip-info.org/wiki/view/Asterisk+modules) and info about them
* **show uptime**: Show uptime information
* **show version**: Display Asterisk version info

**AGI commands**

* **show agi**: Show AGI commands or specific help
* **dump agihtml**: Dumps a list of agi command in html format

**Database handling commands**

* **database del**: Removes database key/value
* **database deltree**: Removes database keytree/values
* **database get**: Gets database value
* [**database put**](http://www.voip-info.org/wiki/view/Asterisk+cli+database+put): Adds/updates database value
* **database show**: Shows database contents
* **database showkey**: Shows database contents: An alternative to showing keys by family with *database show*, this command shows **all** the families with a particular key

**IAX Channel commands**

* **iax2 debug**: Enable IAX debugging
* **iax2 no debug**: Disable IAX debugging
* **iax2 set jitter**: Sets IAX jitter buffer
* **iax2 show cache**: Display IAX cached dialplan
* **iax2 show channels**: Show active IAX channels
* [**iax2 show netstats**](http://www.voip-info.org/wiki/view/Asterisk+cli+iax2+show+netstats): Show network and jitter buffer statistics for active IAX calls
* **iax2 show peers**: Show defined IAX peers
* **iax2 show registry**: Show IAX registration status
* **iax2 show stats**: Display IAX statistics
* **iax2 show users**: Show defined IAX users
* **iax2 trunk debug**: Request IAX trunk debug

* **iax debug**: Enable IAX debugging
* **iax no debug**: Disable IAX debugging
* **iax set jitter**: Sets IAX jitter buffer
* **iax show cache**: Display IAX cached dialplan
* **iax show channels**: Show active IAX channels
* **iax show peers**: Show defined IAX peers
* **iax show registry**: Show IAX registration status
* **iax show stats**: Display IAX statistics
* **iax show users**: Show defined IAX users
* **init keys**: Initialize RSA key passcodes
* **show keys**: Displays RSA key information

**H323 channel commands**

* **h.323 debug**: Enable chan\_h323 debug
* **h.323 gk cycle**: Manually re-register with the Gatekeper
* **h.323 hangup**: Manually try to hang up a call
* **h.323 no debug**: Disable chan\_h323 debug
* **h.323 no trace**: Disable H.323 Stack Tracing
* **h.323 show codecs**: Show enabled codecs
* **h.323 show tokens**: Manually try to hang up a call
* **h.323 trace**: Enable H.323 Stack Tracing

**SIP channel commands**

* Debugging
  + Enable
    - **sip debug**
    - **sip set debug on** (valid on 1.6.2.7)
  + Disable
    - **sip no debug**
    - **sip set debug off** (valid on 1.6.2.7)
* **sip reload**: Reload sip.conf (added after 0.7.1 on 2004-01-23)
* **sip show channels**: Show active SIP channels
* **sip show channel**: Show detailed SIP channel info
* [**sip show inuse**](http://www.voip-info.org/wiki/view/asterisk+cli+command+sip+show+inuse): List all inuse/limit
* **sip show peers**: Show defined SIP peers (clients that register to your Asterisk server), see details [here](http://www.voip-info.org/wiki/view/asterisk+cli+command+sip+show+peers)
* **sip show registry**: Show SIP registration status (when Asterisk registers as a client to a SIP Proxy)
* **sip show subscriptions**: Lists all sip presence (busy lamp indication) subscriptions
* **sip show users**: Show defined SIP users

**Zap channel commands**

* **zap destroy channel**: Destroy a channel
* **zap show channels**: Show active zapata channels
* **zap show channel**: Show information on a channel
* **zap show status**: lists all the Zaptel spans. A span will apear here whether or not its channels are configured with chan\_zap.
* **zap show cadences**: Show the configured ring cadences (available e.g with Zap/1r2).
* **zap set swgain**(<= 1.6): set the (software) gain for a hannel. Temporary equivalents of rxgain and txgain in zapata.conf.
* **zap set hwgain**(<=1.6): set the hardware gain for channels that support it.
* **zap set dnd**(<=1.6) set a channel's do-not-disturb mode on or off.

The following commands are available if the channel is built with support for libpri:

* **pri debug span**: Enables PRI debugging on a span
* **pri intense debug span**: Enables REALLY INTENSE PRI debugging
* **pri no debug span**: Disables PRI debugging on a span
* **pri show spans**: List spans and their status.
* **pri show span**: Information about a span.
* **pri show debug**: show where debug is enabled.

See section 3 of [AT&T tr41459\_99](http://www.att.com/cpetesting/pdf/tr41459_99.pdf) to better understand what the output generated by the pri debug command means.  
  
[Bristuff](http://www.voip-info.org/wiki/view/Bristuff) adds **bri debug** which is an alias for pri debug.

[**Console channel**](http://www.voip-info.org/wiki/view/Asterisk+console+channels)**commands**

* [**dial**](http://www.voip-info.org/wiki/view/Asterisk+CLI+dial) : Dials the given extension, if specified, from the console. Can be used to initiate a call, or to dial digits during an existing call.
* **answer**: Answer a call if one is currently ringing on the console.
* **hangup**: Hangup the call if there is currently one on the console.

**Asterisk channel MGCP commands**

* **mgcp audit endpoint**: Audit specified MGCP endpoint
* **mgcp debug**: Enable MGCP debugging
* **mgcp no debug**: Disable MGCP debugging
* **mgcp show endpoints**: Show defined MGCP endpoints

[**skinny channel**](http://www.voip-info.org/wiki/view/Asterisk+channel+skinny)**commands**

* **skinny debug**: Enable Skinny debugging
* **skinny no debug**: Disable Skinny debugging
* **skinny show lines**: Show defined Skinny lines per device

**Asterisk channel CAPI commands**

* **capi debug**: Enable CAPI debugging
* **capi no debug**: Disable CAPI debugging
* **capi info**: Show CAPI info

[**Sirrix ISDN channel**](http://www.voip-info.org/wiki/view/Asterisk+Sirrix+ISDN+channels)**commands**

* **srx reload**: Reload channel driver configuration; active calls are not being terminated!
* **srx show ccmsgs**: Disable / enable output of incoming callcontrol messages.
* **srx show chans**: Show info about B-Channels
* **srx show globals**: Show info about global settings
* **srx show groups**: Show info about configured groups
* **srx show layers**: Show info about ISDN stack (Layer 1, 2, 3)
* **srx show sxpvts**: Show private info about active channels
* **srx show timers**: Show info about running timers