13.2 (25/04/2017)

- mod: Correction HTTPS read timeouts (when SSL datas pendings).

13.1 (25/01/2017)

- mod: Correction of an issue with the MRCPsynth extra parameters.
- mod: Correction memoryleak with a debug trace in the <assign>.
- mod: Correction bargein issue.
- mod: Correction for speech=auto (use the provider set in the account).
- mod: Correction in MD5 functions for cache managment (voximal report).
- add: Internal parse cache configuration.
- add: Added speechverbio parameter to support Verbio bultins grammars.
- mod: Modification to support Verbio ASR engine with uniMRCP (for buitins).
- mod: Added specific traces for the RAM cache managment.
- mod: Correction to avoid sending grammar actions without finishing the playlist queue.
- mod: Added extra traces to measure the load performances.

13.0 (31/05/2016)

- mod: Correction of a memoryleak with dynamic grammars.
- mod: Correction coredump with long speech responses (NLU).

12.1 (11/12/2015)

- mod: Correction in the number and accurency DMTF builtin grammars.
- mod: Correction to not inspect the tags with DTMF grammars.
- add: Support sln format (PCM 16bit 8kHz Raw).
- add: Add a parameter to use CALLERID with originate.
- mod: Correction to parse the cookies parameter 'secure' and 'httponly'.
- mod: Correction to allow VoiceXML execution after throwing the event disconnect.
- add: mrcpsynthparams for accounts too.
- add: Clean text results from Loquendo ASR (speechclean parameter removes spaces and CR).
- mod: Correction to catch and process the error.grammar events.
- mod: Correction to support alternate prompt using <value>.
- mod: Correction to restore the readformat when the speech resource is released.
- mod: Set the Speech-Language property of the uniMRCP (for builtins grammars).
- mod: Correction to support launching https requests from the Vxml() parameter.

12.0 (03/07/2015)

- mod: Correction to support https: *uri as Vxml() parameter. * mod: Correction to support Loquendo NLSML answers. * mod: Correction to close speechrecord files (generate filedescriptors leak). * add: Option to generate the logs to the Asterisk. * mod: Correction to support builtin time and hour grammars results. * mod: Remove the extra voice parameter in the xml:lang for the grammars. * add: Add a parameter to force the grammar text encoding. * mod: Correcion to save the Verbio format in ISO encoding. * mod: Correction of a regression with the catch event hangup. * add: FreePBX module. * mod: Correction of a memory leak associated to the tag <data>. * mod: Correction to avoid disk saturation with the recordutterance option. ====== 11.0 (05/12/2014) ====== * add: Add sessions max duration. * add: Add average sessions counter. * add: Support for Vestec binary grammars. * mod: Correction to disable the account mark. * mod: Correction with the object Curl and Json parsing. * add: Add new log file (voicexml.log) generated from tag <log>. * mod: Add property interdigittimeout default value set to 3s. * add: Add support JSON parsing to the <object> Curl. * mod: Correction to avoid mode='' with inline grammars. * add: Add support for the Asterisk Festival application. * mod: Restore compatibility with Asterisk 1.4. * mod: Disable the grammars of the field before processing its <filled>. * mod: Correction VXI crash if VoiceXML document is <xml></xml> (OVH). * mod: Correction VXI crash if debian 7 64bits. * add: Add mode parseSrgs=dtmf to check always the grammar content. * add: Add DTMF properties to the ASR engine. * mod: Corrections to support the Nuance Call Steering (grammars). * add: Add parameter recordrewind to disable the 1/4s of cut and the end of the record. * mod: Pass the DTMF to the speech API if mode speechbargein is forced. * add: Mode uri for the parseSRGS option (pass the URLs of the grammars to the ASR engine). * mod: Load when enable the dynamic grammars (grammars with srcexpr). * mod: Force UTF8 encoding for the grammar files. * mod: Change the default value of speechscore to 0. * mod: Correction issue with dtmf input without length. * mod: Support 's' unit in maxage and maxstale attributs (for Nuance Call Steering). * mod: Remove error event if multiple varaiable declarations with <var>. * mod: Add shaddow variables if NLSML <instance> have extra tags. * mod: Correction for noinput/nomatch events with uniMRCP. * mod: Remove ;jsession= in the base URL for the TTS cache. * mod: Disable the ECMAscript strict mode. * mod: Correction fo Verbio sensibility. ====== 10.0 (09/04/2014) ====== * mod: Set transfer variable if callee hangup (near_end_disconnect). * add: Internal objects to process function:, set:, get:, execute: and application: prefix. * mod: No wait after a grammar command. * add: Integration of Resource Speech for Google Speech API. * add: Porting to Asterisk 11/12. * mod: Support original NLSML content response from uniMRCP. * mod: Pass SWI* properties (Nuance ASR parameters) to the ASR engine. ====== 8.2 (16/12/2013) ====== * mod: Set VXML_ERROR if an account reach the max limit. * add: Add parameter 'speechmaxtimeout' (use property maxspeechtimeout too). * add: Add parameter 'systemname' for asterisk (force name). * mod: Change SpiderMonket library (1.5 to 1.8). * add: Added the parameter speechtimeout. * mod: Correction to support completetimeout with Verbio. * mod: Correction to pop the ASR/VAD ('s') event if it is a DTMF inband. * mod: Correction to support the option maxlength in the builtin grammars. * mod: Correction to unmatch voice grammars with DTMF results. * mod: Correction throw a nomatch if input different to length parameter with dtmf. * mod: Correction of unclosed SSML files. * mod: Correction to enable global grammars in <block>. * mod: Correction to use count=0 to enable the reprompt feature. * mod: Correction to support termchar with DTMF events. * mod: Correction for Verbio ASR, to call once SPEECH_ENGINE(langiso). * mod: Corrections in internal NLSML conversion/parsing. * mod: Catch hangup event during record. ====== 8.1*

*(30/10/2013) ====== * add: Support dynamic ABNF grammars (from option/menu). * mod: Set default inputmodes to “dtmf voice”. * add: Get interrupted DTMF from synthMRCP. * mod: Correction when speechunload=no (desactivate the grammar if unfree). * mod: Correction bug in the filenames for the cache. * add: Function VXML(mixmonitor), get default mixmonitor filename. * mod: ASR timeout property set to milliseconds. * mod: Small change for MRCPsynth. * add: Feature to play an audio when waiting for an free VoiceXML channel. * mod: Correction in the option cdrdial. * add: Apache logs for service statistics. * mod: Return Ok when the speechunload is set. * mod: Correction to support the ASR 'hypotesis'. * add: Added Flac record format (for google API). * mod: Correction set the max speech timeout from 10s to 60s. ====== 8.0 (28/06/2013) ====== * add: Pass the speech properties to the ASR engine (configure sendProperties). * add: URI checks (rfc2806) with prefix 'tel:' in dest <transfer>. * mod: Add reference uri: for uniMRCP grammars. * mod: Correction of lastresult$.recordingsize. * add: Add the file size recorded (shadow variable $.size). * add: Throw error.noresource if the ASR is disable instead of error.grammar. * mod: Throw disconnect.hangup only once (after exit execution). * add: Record audio silence before the VAD dectection. * add: Support of the timeout property for the record. * mod: Added support connectiontimeout and maxtime for blind transfer. * mod: Correction for special values returned by the Verbio ASR. * mod: Correction to support HTTP proxy rules. * add: Play/record FLV files (for RTMP channel). * mod: Correction to use an executable content inside a <foreach>. * add: Support of the recordutterance property. * mod: Change the cache key for the audio (skip parameters from the url base). * add: Parameter speechconcatenate used to merge the ASR results to a single string. * mod: Correction of transfers results and events. * add: Support of <mark> (added to block too). * add: Support of fetchaudio with delay and min properties. * mod: Correction hangup event when the caller hung up. * mod: Correction VXI crash if bad encoding with a script. * add: Added the builtin grammar 'dtmf/touchtone'. * mod: Support of HawHaw's generated grammars. * add: Support of <data> and XML dom parsing. * mod: Integration of the last W3C schema. * mod: Improvment with <break>. * add: Support of <foreach>. * add: Parameter to force the SSML format. * add: Use of MRCPsynth for the TTS (method MRCP) * add: Support of consultation transfert (mode keepcontext added too). * add: Support of <grammar> srcexpr attribut. * add: Support of <disconnect> namelist attribut. * add: VoiceXML 2.1 support. * mod: Add 'HttpOnly' attribut for the Cookies. * add: Added CLI command vxml reset statistics. * add: Added CLI command vxml set debug. ====== 7.1 (17/09/2012) ====== * mod: Correction for the Verbio ASR with SRGS grammars. * add: Parameter speechunload to disable the unloadgrammar functions. * add: Range numbers and IP for the accounts. * add: Support the prefix “uri:” for the grammars. * add: Parameter originatedelay2 (time to wait for the originate retry). * mod: Correction Object execution (disabled with the traces). * mod: CURL with SSL support enabled. * add: Account parameter to play an audio if account limit reached. * add: Account parameter to change the volume. * mod: Correction coredump in the curl object. * mod: Correction for the score value for Vestec. ====== 7.0 (22/03/2012) ====== * add: Prefix '@' in the account number to select by caller number. * mod: Correction for S_ISREG symbol not found (load module). * add: Special mode for <clear>, namelist with “.” not clear eventcounters. * add: Account parameter 'wait', wait before start the VoiceXML browser. * add: Option account force 'vxml' to disable '@' execution. * add: Prefix “record:” to access to the recorded local messages (record dest). * add: Add attribute 'dest'/'destexpr' to the record (record in recorddirectory). * add: Parameter to control the originate delay. ====== 6.3 (05/01/2012) ====== * add: Support of date/time DTMF builtin grammars. * add: Configuration parameter to disable to touch the entries (no write in the table). * add: Support audio attributes maxage and maxstale. * mod: Correction of the false VXI connection lost. * add: Parameter monitorformat to set the monitor file format. * mod: Support attribut expr in the tag grammar to evaluate the url (not standard). * mod: Support attribut tag in the tag grammar (not standard). * add: New transfer prefix “outgoing:”, better then the standard originate. ======*

*6.2 (11/08/2011) ====== * mod: Correction to support the Dial alternative without the tel: prefix. * mod: Correction add a fixe maxspeechtimeout to extend the timeout. * add: Option to execute a CLI command after the load (for chan_h323). * add: Option to load a dynamic library (for chan_h323). * add: Monitor option to record the VoiceXML session. * mod: Correction in Verbio/builtin grammar support. ====== 6.1 (25/05/2011) ====== * add: Support builtin grammar with uniMPRC (Loquendo). * mod: Support of audio calls with VAD for the speech/ASR (Loquendo issue). * add: Added XML/Text parser with the CURL object. * mod: Correction to not send 100-Continue when body is empty. * mod: Default optimize mode set to disabled. * add: Added mixmonitor parameter to enable the call recording (randomize possible). * mod: Corrections in the speechrecord parameter (randomize records). * mod: Corrections in the ASR result response. ====== 6.0 (22/03/2011) ====== * add: License option to disable the VoiceXML session. * add: CLI command originate to generate a call (and link it to the vxml application). * add: Callback feature (transfer blind, with connecttimeout null). * add: Modes autoanswer=ringing and force=ringing. * mod: Correction crash with the HTTP VoiceXML objects. * add: Option to record the speech sequences (audio + results). * add: Alternative TTS/TTV URLs. * add: Feature to Dial 2 channels simultaneously, or as an alternative. * add: Option to decode the URL in the Asterisk module. * add: TTY/TDD send/receive from VoiceXML (first step). * add: Support RFC5552 (exit/result in the BYE), update the SIP channel patch. * add: Support RFC5552 (only the VoiceXML URL in the INVITE), needs a SIP channel patch. ====== 5.2 (30/11/2010) ====== * mod: Modification to support the audiofetch. * mod: Correction in the fonction wait/silence (wait for openvxi). * add: Add the parameter dialnumbersonly to filter called numbers. * mod: Change the open sequence for better reactivity. * mod: Increase the number of accounts to 200. * add: File descriptors counters (with show top). * add: Average statistiques (duration, response and CAPS). * mod: Replace the nanohttp library by the libcurl. * add: Extra parameters in the transfer (after mark ',') for to the Dial command. * add: Command line parameters -U and -G to change the OpenVXI linux user/group. * mod: Correction for uniMRCP to stop the speech/ASR engine. * mod: Correction for disable bargein with the speech/ASR. * add: Increase the Asterisk compatibility (disable the using the channel context). * mod: Ignore the ASR result if a DTMF interaction occured. * add: Support the attribut repeat for SRGS/XML DTMF grammars. * add: Options to pass all the SRGS/XML grammars to the ASR engine (voice and DTMF). * add: Support for Asterisk 1.8. * add: Add the context support in the app_vxml redirect ('@exten@context'). * mod: Correction for MDTel (shadow 'out' value set). * add: Option parseSRGS to disable the SRGS parsing in the browser. * mod: Correction for speech unimrcp and Nuance (support <tag>out=). * mod: Correction of the session contexts initalisation (ctx→url). * mod: Enable the cdrupdate in case of vxml(@) using. * mod: Correction to enable the H323 license option. * add: Add the parameter param (to force the session.param variable in the VoiceXML context). * add: Add the parameter priorityevents. * mod: Correction for Vestec ASR (word compare without case sensitive). ====== 5.1 (02/09/2010) ====== * mod: Corrections for the ASR (speech). * mod: Correction to use the separator char parameter. * add: Dialer, H232, RTMP options. * mod: Correction of uninitialized account index in the context. * mod: Correction of the nomatch when digits/minlength=1. * mod: Correction to return the result/value of an executed application. * mod: Corrections of the annoncement feature (Karaoke function with the record). * mod: Correction of the Verbio Speech timeout. * mod: Close the opendir when monitor action tries to purge the cache. * mod: Remove “{out=value}” in the gererate ABNF grammars. * mod: Disable the CDR modifications with cdrupdate=no. ====== 5.0 (25/05/2010) ====== * mod: Add force Flash/RTMP for the future RTMP channel. * mod: Correction to avoid the crash when the object http don't get parameters. * mod: Correction to use h324m with Asterisk 1.6. * add: Support audiofetch attribut. * add: Cache manager to purge automatically the TTS cache. * mod: Use UTF-8 for srgs/xml grammars. * add: Porting of Vestec ASR (with asterisk speech API).*

** mod: Correction to remove null file generated by the HTTP/TTS connector. * add: use transfer/dest extra value ';ani=xxx' to change the CALLERID number and name. * mod: Resize the messaging buffer. * mod: Correction to support refered file for the dial annnouncement. * mod: Correction to play mp3 streams with mp3player. * add: ASR/speech uniMRCP support. * mod: Correction in the start/strop script for Suse. * mod: Replace usleep function (some server don't sleep). * mod: Correction to pass the parameters for with “execute:” (Asterisk 1.4). * mod: Correction to force ULAW/ALAW native format (parameter force=alaw). * mod: Correction to set different filename during the multiparts/POST. ====== 4.4 (03/03/2010) ====== * add: Complete DTMF buffering during HTTP long requests. * add: Add paramter threshold to configure the VAD/silence (record). * add: Add parameter autoexit to kill asterisk if the connection with VXI is lost. * add: Set record maxtime shadow variable. * add: Improve prompt hangup and bargein (skip HTTP processing, limit queue-fill). * mod: Select the first account with redirection(s). * add: Add clean support of noinput and hangup event during the record. * add: Add the account parameter “force” to set Transfercapability=VIDEO. * mod: Improvement of the bridge transfer (use with transcode). * mod: Disable the msgqlock. * add: Add parameter videoprofile (to controle the video codec transcoder). * add: Check the account in the vxml(@) execution. * mod: Correction to control the call answer. * add: bridge and spawn modes for localformat. * mod: Add the DOCTYPE in the grammars. * mod: Correction in the session release (wait for playall). * mod: Correction for better speech support. 4.3 (11/01/2010) ———— * add: Property announcememory for the <transfer>. * mod: Corrections in the <object> “property”. * mod: Corrections in the <transfer> features (maxtime property added). * add: Parameter durationlimit in the accounts. * add: Use the 3rd language item as the voice (en-UK-brian). * add: EC2 and Xen support. * mod: Correction to not cut float numbers and hours with the option cutprompt. * mod: Support of GRXML in DTMF mode. * add: Parameter messaging to control the MWI notification. * mod: Correction for the originate feature. * mod: Correction to allow the Video Conference (Konference integration). * add: Optimization option (disable full parsing). * add: builtin:amd grammar to detect the Ansering Machines. * mod: Get the variable result of a application, application:app()=RES. * mod: Corrections for the Fax support (return number of pages) * mod: Enable to execute Applications after the hangup. * add: Integration with the Dialer. * add: Build modules for Asterisk 1.6.0.x and 1.6.1.x * mod: Correction in the license lock. * mod: Set the DTMF dial timeout and duration (dial:xxxx,timeout,duration) * mod: Use the values 99/98 in the confidence attribut for events/field. * mod: Updates for the Asterisk 1.6.1.x (ast_dsp_set_digitmode). 4.2 (17/09/2009) ————– * * mod: Correction for speech using (disable speech if no grammar active). * mod: Support returning speech results during controled prompts. * add: Support VOICE/SRGS (application/srgs+xml) load grammars. * add: Record types “format/…” to support all the Asterisk formats. * add: Support DTMF/JSGF (text/x-grammar-choice-dtmf) load grammars. * add: blindapplication parameter (select an application or use a end dial). * add: CDR option cdrconference to enable CDR generation for the conferences. * add: Add append variable/function feature (with '+='). * mod: Correction in the read function feature. * add: Asterisk 1.6.1.x Porting. * add: Add addition version informations. * add: Support VCR command from the ASR/speech. * add: Support of 'hotword' with the bargeintype. * mod: Correction in the “uri:” prefix. * add: Memory for media files. * mod: Correction to support an empty URL. * mod: Correction for the 64bits portability. * add: object 'delete' * add: URI with “originate:” to generate outgoing calls frome the <transfer> tag. * mod: Corrections in the internal objects (string conversions). * add: Add the maxEvents configuration parameter. * mod: Support passing parameters to the conference (conf:test/M). * add: URI with “app:” to execute an application from the <audio> tag. * mod: Stop the speech resource on DTMF key pressed. * mod: Correction to check the structure of speech results. * mod: Correction to match an account with the name. * add: Add parameter speechbargein to disable the voice bargein. 4.1 (29/05/2009) ————– * add: CDR option cdrspeech to enable CDR generation for the speech/ASR (recognize). * mod: Modification in the object 'pick'. * add:*

*Prompt cut feature, cutPrompt parameter and promptcut proprerty. * add: Option to force a Silence (after prompts). * add: Support maxlength and minlength in the DTMF digits builtin grammar. * add: Option autohangup to hangup the call after the VoiceXML session. * add: Support Asterisk .h263, .h263p and .h264 files. * add: Set and function write from the transfer tag. * mod: Send DTMF modifications (for Asterisk 1.6). * mod: Link VXML_PARAM and VXML_AAI to session.connection.aai. * mod: Account matching (refund and Asterisk Dialplan patterns support). * add: Parameter mark for the accounts (to add a mark in the OpenVXI traces). * add: Additionnal marks in the logs/traces for VoiceXML hosting. * add: VXML_URL2 as VXML_URL alias because the channel SIP use the same variable name. * add: Protection against infinite loops in the account redirections. * mod: Set a minimal size for the msgq. * mod: Correction for Asterisk 1.6 (disable build options sum check). * add: New lock system for the i6net modules. * add: Add autoreload parameter alias for the configuration. * mod: Change the format of the prompt CDR. * add: Transfer “tel:function(X)” get the function value of X with the shadow $.value. * add: Dump the installation date. * add: Values of the prompt properties used for the prompt cache key. * mod: Reload configuration if the configuration file date change. * mod: Correction to disable the debug traces. * add: CDR option cdrparam to set the userfield with VXML_PARAM. * mod: Add local/remote info in the session dump. * mod: Mode speech=emulation (disable messages). * add: Use '@' in the url to redirect an account to another. * mod: Correction for accounts, '*' to catch all the numbers. * add: Transfer “tel:get(X)” get the extention variable value of X with the shadow $.value. * add: CDR option cdroverwrite to update the CDR with the variables VXML_LOCAL and VXML_DISTANT. * add: CDR option cdrprompt to enable CDR generation for the prompt (audio). * add: CDR option cdrdial to disable the CDR creation with Dial (transfer). * mod: Correction for the property promptbackground * mod: Correction for the grammars generated with <option> 4.0 (18/01/2009) ————– * add: Modules for each Asterisk releases (1.4, 1.6 and videocaps) * add: Celudan/3Gbuilder application control * add: Asterisk 1.6 support * add: Multiple ASR/speech configuration * add: Set the record/termchar shadow variable * add: ASR score result * add: ASR configuration (enable GRXML dynamic grammars) * mod: ASR/speech integration redisgned (use Asterisk application instead the API) * add: ASR configuration (enable isolated and ABNF dynamic grammars) * add: objects 'save' and 'pick' * add: dialformatvideo to set the transfer parameters for the video * mod: Enable to customize the transfer applicatuin used * mod: Bug license code correction for the 64bits version * add: Update CDR/accountcode with the name of the vxml account * mod: Correction of the SRGS sytnax for Lumenvox * mod: Correction for the jsession * mod: Refund of the transfer execution * mod: Return the rigth duration after a transfer * add: Keep CDR after the transfer (with Dial) * mod: Http 302 support (as Mozilla) * add: Internals parameters and dump information * add: Default timeout and interdigittimout configurable * mod: Bug timeout correction (due to timeout in prompts) * add: MaxLoopIterations and MaxDocuments configurable * add: Trace level for VoiceXML development * mod: Cache/localfile bug correction * mod: Simplification of the properties dump * add: Parameter speechscore to throw a nomatch * add: Support to audiomaxage and audiomaxstale * add: Option to auto reload the configuration * mod: Support timeout attribut of prompt section * mod: Correction for the coredump when using the ASR (speech) * add: Top dump (from the CLI Asterisk) * add: Add a specifc HTTP connector for the TTS/TTV (allow HTTP connected) * add: Enable/disable interpreter traces * add: Get the delay for the first command after an open 3.1 (31/08/2008) ————– * add: additional properties for the TextToVideo * mod: Disable SIGPIPE generation * add: Specific video URL in the accounts * add: Video detection * add: Counters (PEAK, DENIED, SPEECHS) * add: Set VXML_ERROR if the session cannot be open (content the cause) * add: End date to the session dump * add: Use Number (calledif)) to identify the account * mod: Open sessions locks * mod: support Jsession (java sessions) * mod: Start/stop script (without safe_openvxi) * add: Option mute to openvxi * mod: Disable log Stdout by default * add: CLI* admin commands * mod: Remove direct chan access * add: alias mimitype video/3gp * add: VXML() asterisk function to get/set parameters * add: Porting for Asterisk 1.2 * add: Priority configuration * add: Sessions dump * add: CDRupdate parameter * add: Asterisk vxml application dates * mod: vxml show application * add: Add the dial: transfer prefix * add: CDR updates at the end of the VoiceXML session * add: .alaw and .ulaw formats for the TTS * add: ASR automatic allocation * add: speech configuration for the accounts * mod: Correction in the offset object * mod: Small correction for CLI commands * add: Object property to get internal properties values * mod: Correction to use the MP3Player application * mod: Correction to support exec: in the transfer * add: Configuration of DTMF controls 3.0 (07/05/2008) ——————– * mod: Remove applications integration (call from Asterisk) * add: Conference from the transfer tag with conf: * add: Call a Asterisk application fom the VoiceXML session. * mod: Prompt local file not exits correction (item increment removed) * add: Support 3gp file format extension * mod: Update from sip.fontventa.com (12/01) * mod: Update from sip.fontventa.com (15/12) * add: Test tool to check the code * mod: Update from sip.fontventa.com (29/11) * add: builtin and dynamic speech grammar supported * mod: Number of account changed to 100 * mod: Update from sip.fontventa.com (22/10) * mod: License bug correction * add: First step of speech integration 2.2 (08/10/2007) ——————– * mod: Update from sip.fontventa.com (08/10) * mod: Correction to prompt local file * mod: Not overwrite the client.cfg for the upgrades * mod: Start/stop script option “kill” to purge * add: Offset support with audio/wav (“.wav:1233”) and property <control> * mod: Change context item type : form int to long (offset) * mod: Remove TTS/HTTP overwriting gobal HTTP request paramaters ====== 2.1 (12/09/2007) ====== * add: Infinite loop dealock detection (generate error event) * add: First step to support UTF-8 message from gtalk * mod: Update from sip.fontventa.com (18/08) * add: DTMF interrupt / bargein during RTSP * add: Support simple <break> tags. * add: Control VCR function for wav clips with property “control” * add: Dial format in the account configuration. * add: Support RTSP uri with <audio> * mod: Correction for multi-records * add: Accounts managment * mod: Bugfix for SpiderMonkey (js_FreeRuntimeScriptState) * mod: Invert DNI and ANI (with the same shift in the configurtion file) * mod: Update the ChangeLog ====== 2.0 (01/03/2007) ====== * Official release for Etch (gcc4) * add: QOS counters (prompts, recognizes, records …) * add: MP4 from Asterisk-Video (Sergio / sipfonta). * mod: remove ffmpeg dependencies * mod: remove mov / 3gp format ====== 1.5.beta (xx/03/2007) ====== * mod: Etch support ====== 1.4.beta2 (13/02/2007) ====== * mod: Remove the message “Bad video codec” after the configuration load. * mod: Allow record multiple files * add: Makefile options * mod: Bug wrong ID grammars (0x7FFFFFF) * mod: Don't lock if OpenVXI is not started * add: Support of mp4 file format (from app_mp4) * mod: termchar empty disable the “#” default key ====== 1.4.beta (04/10/2006) ====== * Initial Release (for distribution) * add: Add information files * add: Add licensing files * add: Add asterisk application source for GPL (app_vxml) * mod: Change the licensing interface * add: Option recordsilence * add: Option videoupdate * mod: Corrections for video recording