

## 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 DTMF 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: mrpcsynthparams 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 buitins 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: Correccion 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 speechbargin 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 variable declarations with <var>. \* mod: Add shadow 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 maxspeecheventtimeout too). \* add: Add parameter 'systemname' for asterisk (force name). \* mod: Change SpiderMonket library (1.5 to 1.8). \* add: Added the parameter speecheventtimeout. \* 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 unmatched 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 detection. \* 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: Improvement 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 maxspeechovertimeout 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 audiodfetch. \* 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 announcement 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 generate 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 audiodfetch 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 announcement. \* 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 multipart/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 from 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 property. \* 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: Additional 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 extension 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 redesigned (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 syntax for Lumenvox \* mod: Correction for the jsession \* mod: Refund of the transfer execution \* mod: Return the righth 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 specific 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 configurion 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 (0x7FFFFFFF) \* 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*

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