13.2 (25/04/2017 devel)

- 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