Voximal ChangeLog



The old name of Voximal is VXI* : full VXI* ChangeLog

Voximal is the new generation of VXI* 12.0 , and integrates branch 13.0

14.2

02/01/2020

- mod: Improve the AAI support. (Can be linked to User-To-User SIP header).
- mod: Improve the fetchaudio feature.
- mod: Add Google Assistant Dialogflow payload support.
- mod: Correction to ignore XML tags in the NLSML <input>.
- mod: Important correction in the earlyspeech feature.
- add: Add ConfBridge conference application support (Transfer with conference: prefix).
- add: Add a property to manage the Google Credentials files name and allows serveral GoogleSpeech/GoogleDialogflow connexions.
- mod: Correction coredumps with command “clear cache” and files older than x days.
- add: Add <script> extension to execute variable contents as an ECMA script.
- mod: Correction to force the score with Google Speech when empty final result.
- mod: Modification to send speechprovider with load grammar URI.
- mod: Correction aoround cut/break features.
- add: Add property to define the TTS text/ssml encoding for MRCP contents.
- mod: Add parameter a speechunanswered to keep grammars errors during unanswered processing.
- mod: Correction for MRCP (bad streaming state check).
- mod: Correction to extend the buffer to receive long ASR/STT results.
- mod: Correction crash with grammars and unaswered mode.
- mod: Update to the last openssl, grpc libraries (SSL/ALPN property).
- mod: Corrections for HTTP speech processing.
- add: Variable SPYGROUP for ChanSpy using.
- add: Support payload from Dialogflow webhooks (expect_user_response).
- add: Add multicontext support for Google Dialogflow (with xml:lang=“context1:context2”).
- mod: Correction to send the JSGF grammars to the ASR engine with uniMRCP.
- add: Add a new bargein mode called “early speech”. Gets speech and interrupts prompts during the end of the prompts.

- add: Average values for feedback and response time (between recognizes and prompts).
- mod: Correction in the messaging waiting step (when the stream/recognize have a result).
- add: Add match RegExp expressions in the grammars, prefix the item with the '@' character.
- add: Add multilanguage support for Google Speech (with xml:lang="fr-FR:es-ES").
- mod: Correction coredump in assignments with void EcmaScript objects.
- mod: Correction to use the attribut timeout in the <prompt>.
- add: Add <clear namelist=":"/> to clear event and prompt counters.
- mod: Correction of the default memory audio directory (transfer announcement and karaoke features).
- add: Add parameters audiotruncate and audiopollard to cut TTS prompts.
- add: Add property encodingtype to set or modify the default POST encodingtype.
- mod: Do not clean text contents with the TTV (language=text or video).
- add: Add the parameters maxLogFileSize and maxContentDirSize.
- mod: Correction to keep the set "context:" on fields after a noinput/nomatch event.
- mod: Corrections around Google RPC timings and answers.
- mod: Correction to support prompt properties with the Watson API.
- mod: Correction to support DTMF results with score equal to '1' (Nuance ASR).
- mod: Correction to forward DTMF events to the speech API (ASR) if DTMF grammar is used.
- mod: Dynamic grammar now support empty srcexpr with throwing an error event.
- mod: Change the word/grammar parsing to find full words and not substrings.
- add: Add the parameter Speech Provider in the General section in the FreePBX module.
- add: Add Dialogflow shadow variables (hangup, intent, name, property...).
- add: Add Parameters for the interpreter properties (section [interpreter]).
- add: Add parameter minimalspeech, to bufferize the STT streaming before starting the session.
- mod: Correction for the TTS with MRCP configuration.
- add: Add property promptlang (and xml:lang) to force the languauge for all prompts.
- add: Add the option useredirect for SIP redirected calls (with OVH french VOIP provider).
- mod: Disable the option speechhotwordscore by default.
- add: Add "[]" break marks as "{}" for Dialogflow.
- add: Add "unanswered" mode to execute a VoiceXML session without a real call (function "url").
- mod: Improvments with the Dialogflow integration (set request payload, process result parameters, stability).
- add: Add "{}" break marks in prompt texts to add prerecorded audio files or wait time in ms.
- mod: Correction bug with the cutprompt feature.
- add: Add extraheaders property.
- add: Add parameters to generate silence to the audio streaming (with RTP silence).
- add: Add parameters to record audio streaming (GoogleSpeech/DialogFlow)
- mod: Remove too verbose traces during HTTP downloads.
- mod: Continue Playing the same audiotransfer during several sequential transfers.
- add: Add Yandex Synthesis support.
- add: Add Dialogflow integration with RPC (text/event and streaming modes).
- add: Add multiple text/transcribe grammars in a single document.
- mod: Default STT streaming set to ulaw.
- add: Add the minspeech property to start the STT only if we reach a minimal speech duration.
- mod: Change the default mark with the parameter id if set.
- mod: Change the Cereproc Cloud integration (SpeakExtended support).
- add: Add parameter speed (and promptspeed) to change the TTS voice rate with Cereproc and Watson.
- add: Integration of GoogleSpeech V1.1, with the enhanced model.
- add: Add Google Text To Speech support.

- add: Add Google Speech Streaming features to improve the results.
- add: Add transferaudio support.
- mod: Corrections in STT streaming (async thread mode).
- mod: Correction to detect the pause/stop with Google Speech Streaming.
- add: Add codecs ULAW and OGG for Google Speech Streaming.
- add: Add a HTTP client for the <data> tag (to keep SSL connections with chatbots APIs).
- mod: Corrections with STT streaming and bargein.
- mod: Improvements in the STT streaming integration.
- mod: Support maxspeechtimeout with the STT streaming.
- mod: Correction to allows flexible URI in the attribute dest with the <transfer>.
- add: Add bargein support with the Speech API Streaming.
- mod: Correction to remove spaces from the digits/number grammar with STT.
- add: Add Google Speech API with streaming.
- add: File streaming feature to get the speaking audio flow in the interpreter.

14.1

28/05/2018

- add: Add the server name to the frame Hello with SSL.
- add: Add speechrecordsilence parameter for the Speech recording.
- mod: Correction regression with JSGF grammars with mode=dtmf.
- mod: Add the utterance in the nomatch for the SpeechToText.
- mod: Stop the process with signal TERM and after KILL.
- add: Add option -pid to create the voximald.pid PID file from the interpreter too.
- mod: Correction Voximal cleanup at Asterisk exit.
- add: /var/run/voximal/voximal.pid file.
- mod: Correction of JSON/Ecma conversions.
- mod: Correction of the GUI/General parameters.
- mod: Correction in the install script.
- add: Support for Asterisk 15.

14.0

12/07/2017

- add: Correction for DTMF grammar with white spaces.
- add: Add 'hidden' grammars to set replace word/string in the STT results.
- add: Add 'hidden' grammars to set “Phrases” to Google STT.
- add: Silences record with STT, with speechrecordsilence option.
- mod: Changes for Google Speech API V1 (not beta).
- add: Add the property recognizemodel for Watson STT.
- add: Escape the “ by \” in the JSON string contents.
- mod: Correction freePBX module for the option dialformat.
- mod: Correction to support uniMRCP configuration.
- mod: Correction crashes with JSON/TEXT <data> requests.

- mod: Correction to support HTTPS server reset connection (Keep-Alive with timeouts).
- mod: Correction DEV logs for STT.
- mod: Correction with the speech beeps.
- add: Integration of the TTS Amazon/Polly with CLI commands.
- mod: Correction HTTPS read timeouts (when SSL datas pendings).
- mod: Escape the HTTP parameters characters.
- mod: Corrections in the chunk HTTP download.
- mod: Use the directories files and streams for the log contents.
- add: Add parameter lang for the builtin grammar text (text?lang=x).
- mod: Correction of an issue with the MRCPsynth extra parameters.
- add: Add a mark for the VM and specific Voximal installs.
- add: Use the sensibility end completetimeout to adjust the speech recording for STT.
- add: Add JSON support for <data>
- mod: Correction of a memoryleak with a debug trace in the <assign>.
- mod: Add error messages relative to write disk errors and MSQ read errors.
- mod: Set the PlayListSize to 1 to avoid MSQ lock when the MSQ size is to small.
- add: Support STT with menus/options/grammars using (interpreter filters results).
- add: Add speechprovider parameter for the accouts in the FreePBX module.
- add: Integration of the STT IBM/Watson Cloud API (bluemix).
- add: Integration of the STT Microsoft Cloud API.
- add: Integration of ASR/STT with HTTP interface.
- mod: Correction in MD5 functions for cache managment.
- add: Integration of the TTS IBM/Watson Cloud API (bluemix).
- add: Added speechverbio to support Verbio bultins grammars.
- add: Integration of the TTS iSpeech Cloud API.
- mod: Update for the new TTS Microsoft/Bing Cloud API.
- mod: Disable the default POST/100-continue and add the property fetchcontinuetimeout.
- mod: Disable the unload grammar execution by default.
- add: Add wav16 and sln16 support.
- mod: Allows to use one free port with an invalid key.
- mod: Correction to avoid sending grammar actions without finishing the playlist queue.
- add: Asterisk 14 support.
- mod: Disable the POST continue for the TTS requests by default.
- mod: Correction to fully support the POST continue to pass HTTP1.0 proxies.
- add: Integration of the TTS Microsoft Bing Voice Output API.
- add: Integration of the VoiceRSS Cloud Text-to-Speech API.
- mod: Enable to start without configuration file, with defaults parameters.
- mod: Change log directory to /var/log/voximal.
- mod: Change cache directory to /var/cache/voximal.
- add: Support of NLSML answers from Telisma ASR engine.
- add: Option unimrcp to start unimrcpserver, as voximald.
- add: Option cacheclear to clear the cache directories at startup.
- add: Integration of the CereProc Cloud Text-to-Speech API.
- add: Support ogg format (Vorbis OGG 8kHz).
- add: Add max retries to avoid to disable the license immediately.
- 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 for MRCPsynth using without cache.
- mod: Correction to allow VoiceXML execution after throwing the event disconnect.
- add: Added mrcpsynthparams for accounts too.
- add: Integration of the Voxygen Cloud hosted.
- 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>.
- add: Refund to use functions, names and directories based on voximal.
- add: Integration of the SpeechAAS TTS hosted.
- add: Json HTTP/POST request support (with enctype="application/json")
- mod: Set the Speech-Language property of the uniMRCP (for builtins grammars).
- add: Auto-extend threads in the interpreter if needed.
- add: Start the interpreter from the Asterisk module.
- add: Remote License System integration.
- mod: Correction to support https: *uri as Vxml() parameter.*

From:
https://wiki.voximal.com/ - **Voximal documentation**

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