



VoiceGenie VoiceXML Gateway

Release Document

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Summary

Overview

This release document includes the following information:

- Product Version Identification
- New Features
- Changes
- Bug Fixes
- Errata
- Change History

This document applies to the SpeechGenie edition of the VoiceGenie VoiceXML Gateway product, release 5.2.

Documents

In addition to this Release Document, you will get four other documents:

VoiceGenie Platform Pool Manager Release Notes

VoiceGenie Administration Guide - A manual written for those who will install, configure, and administer the VoiceGenie VoiceXML Gateway Platform.

VoiceXML Support Document - contains detailed information on VoiceXML compliance, usage notes, and previous change history.

VoiceXML Developer Guide - A VoiceXML 2.0 reference manual containing description of each supported tags, attributes, properties, and other VoiceGenie extensions.

Other relevant documents may also be included.

Resources

There are many resources for developers available on VoiceGenie Developer website <http://developer.voicegenie.com>. The following lists some of the important items you can find on our website:

Resource	URL
FAQ	http://developer.voicegenie.com/faq.php
Tutorials	http://developer.voicegenie.com/tutorials_VoiceGenie.php
VoiceXML 2.0 Reference	http://developer.voicegenie.com/voicexml2tagref.php
Genie IDE (Integrated	http://developer.voicegenie.com/IDE.php

Development
Environment for
VoiceXML)

Where To Get Help

The VoiceGenie Developer website contains forums for the developer community to post questions and discuss problems. We have forums that discuss Application Development, the SpeechGenie Platform, the VoiceXML Language, etc.

The URL for the list of forums is: <http://developer.voicegenie.com/forums.php>

VoiceGenie Customer Support

Email: sg_support@voicegenie.com

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For 24/7 support, please contact VoiceGenie.

Contacting VoiceGenie

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Product Version Identification

Product Name	VoiceXML Gateway
Version	5.2
Release Date	February 15, 2002
Nuance ASR	Version 7.0.4 SP19
Text-To-Speech (TTS)	AT&T Natural Voices 1.0
Operating System	SCO UnixWare 7.1.1 Departmental Edition

New Features

- Supports prompts queued across document boundaries. Please refer to the tutorial for detailed information:

http://developer.voicegenie.com/tutorials/VoiceGenie.php?tutorial=prompt_queuing

- Support for NIST audio header;
- New error events for TTS and ASR;

error.grammar.asr	Speech grammar error.
error.asr.unknownengine	ASRENGINE is unknown.
error.asr.noresource	No ASR resource is currently available.
error.asr.exceedlicense	ASR resource requested has no more license.
error.tts.unknownengine	TTSENGINE is unknown.
error.tts.noresource	No TTS resource is currently available.
error.tts.badcapability	TTS does not support a requested capability such as a language.
error.tts.exceedlicense	TTS resource requested has no more license.
error.tts.badtext	Bad format in speech markup (Applies to Speechify 1.2 only)

- <log> element can now log to specific destinations. A new attribute "dest" is introduced. The attribute can have one or more of the following space-separated values:
 - metrics - default value; metrics log
 - bill - billing log
 - callmgr - currently ignored
 - syslog - system log
 - file:[path/filename] - user defined log file relative to the log directory
 - none – this <log> is ignored
- New configuration parameters supported in voicexml.cfg:
 - ABNF_GRAMMAR_LOOP_COUNT - Maximum loop count used in the ABNF recognition routine to catch an infinite loop; default is 1000
 - ENABLE_LOGDEST - List of log destinations allowed separated by '|'; destinations could be "metrics" "file" "bill" "callmgr" "syslog"; no default.
 - In previous releases, <log> messages were logged to the metrics file. Set ENABLE_LOGDEST=metrics to keep the same behavior in this version.

- MAX_REDIRECTION - Maximum number of consecutive fetch redirections allowed; default is 5
- <clear> element can be inside a <menu> element with no namelist attribute defined;
- New audio control feature COM.VOICEGENIE.AUDIO.NEXT (or VGNEXT). When this DTMF keypress is detected, playback will skip to the next audio file;
- Allow <join> to a channel before the call is connected with a <call> tag. This allows listening to the ring tone while connecting;
- Configurable parameters for setting initial channel state; see the administration guide for further information;
- Initial support for hook-flash for blind transfers with analog and Robbed bit T1;
- Supports “bargeintype” attribute in <prompt> element;
- “timeout” attribute in <prompt> allows value of 0s;
- Added new ‘alias’ type for TTS Manager. Multiple symbolic names can be defined for a particular TTS engine;
- Added new properties to define default inline grammar type (without the need to specify in the type attribute):
 - COM.VOICEGENIE.GRAMMARTYPE.SPEECHWORKS
 - COM.VOICEGENIE.GRAMMARTYPE.NUANCE
 - COM.VOICEGENIE.GRAMMARTYPE.WATSON

Changes

The following are changes from the previous Release 5.1. Please note that some of the following changes may affect applications written for previous versions of SpeechGenie.

- Throws “error.semantic” if <enumerate> element is not a child of <field> or <menu> elements. Please refer to the tutorial for detailed information on handling error events:

http://developer.voicegenie.com/tutorials_VoiceGenie.php?tutorial=errors

- Audio Control properties have been renamed (Old properties are still supported in Release 5.2);

New Properties	Old Properties
COM.VOICEGENIE.AUDIOCONTROL	VGAUDIOCONTROL
COM.VOICEGENIE.NOAUDIOCONTROL	VGNOAUDIOCONTROL
COM.VOICEGENIE.AUDIO.SKIPAHEAD	VGSKIPAHEAD
COM.VOICEGENIE.AUDIO.SKIPBACK	VGSKIPBACK
COM.VOICEGENIE.AUDIO.SLOWER	VGSLOWER
COM.VOICEGENIE.AUDIO.FASTER	VGFASTER
COM.VOICEGENIE.AUDIO.SOFTER	VGSOFTER
COM.VOICEGENIE.AUDIO.LOUDER	VGLOUDER
COM.VOICEGENIE.AUDIO.PAUSE	VGPAUSE
COM.VOICEGENIE.AUDIO.NEXT	VGNEXT
COM.VOICEGENIE.AUDIO.STOP	VGSTOP

- If property “universal” is not specified, the default value is “all” for VoiceXML 1.0, and “none” for VoiceXML 2.0;
- "nomatch" event will be thrown for DTMF input if inputmodes="voice" or inputmode="both" without a DTMF grammar.

Bug Fixes

The following are bug fixes since Version 5.1:

- Fix audio control feature VGPAUSE
- Fixed processing status messages from ASR and TTS (as viewed in Health Monitor)
- Fixed with DIAL (<call>) where call drop message to vxmli was missed
- Fixed some issues with RLT
- Return ASR failure if inputmode is "both" instead of ignoring the failure and only wait for DTMF input
- Allow slot names and values including "mode", "root" or other key words for ABNF grammar;
- Fixed mime-type "audio/basic" used for .ulaw and .vox file.
- Check for infinite HTTP redirect requests
- Format string properly for "date" and "time" type for Natural Voices TTS Engine
- Fixed transfer duration if near end disconnect
- Added timestamp for record/utterance file names so they will remain unique when platform is restarted
- Time in <break> element is interpreted properly in Speechify 2.0
- PR 237: Fixed ghost VoiceXML interpreter instance
- PR 270: Report detailed error on "type" attribute in <grammar> element
- PR 271: Corrected problem with the top-level rule tag can't assign a value to SWI_meaning, or the grammar return value becomes meaningless (just a plus sign).
- PR 272: lastresults array is no longer empty in <catch> block when thrown by <link> element grammar.
- PR 273: Fixed invalid Nuance generated when grammars do not contain starting period.
- PR 241: undefined variable used in expression should returns false.

Restrictions

The following restrictions apply to this release:

1. error.tts.badcapability is not used.
2. Unpredictable behaviour if a keyword is used as part of an ABNF grammar if it is not inside the tag {...}. For example:

```
<grammar> jump to root page {rootpage} </grammar>
```

"jump to" will be converted to Nuance grammar, and "root page {rootpage}" will be deleted.

3. If an slot value includes "<" in a Nuance Grammar, then the return value will be wrong. For example:

```
.Rule1  
    Toronto {<city "montreal < toronto" >}
```

The return value will be "montreal <_gram1_toronto" instead of "montreal < toronto".

4. Using <join> before a <call> connects must be uni-directional and only join the talk channel of the outgoing call. Otherwise the call will fail to connect. Here is an example of the proper way of using <join>:

```
<call name="xfer" dest="tel://4167360905" returnwhen="immediate"/>  
<assign name="you" expr="xfer$.chan"/>  
<assign name="me" expr="session.com.voicegenie.telephone.primarychan"/>  
<join name="joinchan" listen="me" talk="you"/>
```

...

Errata

Please refer to the Developer Workshop for complete list of Errata:

http://developer.voicegenie.com/errata_VoiceGenie.php

As of the release date, the following issues exist:

PR#	Summary
59	Cannot catch error.noauthorization event
76	Speech markup does not work within <audio> ... </audio> tag pairing
123	Recognition performance deteriorates when "timeout" is used
132	Count attribute in <catch> does not work with user events
184	Misleading error message - page not found vs. audio/grammar file not found
213	VGSTOP audio control drains prompt queue until checkpoint
224	Two <link>-ed grammars may not work properly
234	Need to be able to bargein on prompts queued prior to off-page subdialog call
237	Health Monitor reporting for vxmli number of calls may not be accurate under certain conditions
243	Durations written as floats are not supported
246	Cannot access the "restricted_phone_no" variable for call blending
248	Cannot get "near_end_disconnect" information in the log file for call blending
249	Cannot catch "com.voicegenie.call.answered.Ok" event when calling a modem
250	<prosody> tag does not take any effect
252	Compile defaults.vxml on initial page fetch
253	defaults.vxml does not set all 2.0 caching defaults
256	<value> mode = recorded has some issues with prerecorded files
274	Grammar slot returned from different grammars is ignored
275	Incorrect evaluation of expr in <audio>
276	When audio file is played, Dialogic returns a DTMF "1" and the callmgr would skip over the rest of the prompts
279	cannot catch "error.tts.badtext" event in Unixware platform
280	callmgr won't pick up the call after executing message_asr_grammarerror testcase
281	<meta> inside <grammar> tag is not supported in Unixware platform
282	RRU is not supported for Unixware platform
283	<grammar> one two three </grammar> has to be written as <grammar> one {one} two {two} three {three} </grammar> in order for bargeintype = recognition to work on Unixware
284	cannot catch "error.badfetch" event if grammar mode doesn't match the actual input
286	Comments ending with "--->" will cause a semantic error
288	Nuance client may crash under sever load on both ISDN & Robbed Bit T1 - approx 5 per 100,000 calls on Unixware and NV TTS

Change History

Release 5.1

New Features

- Support for multiple TTS Engines running on the same platform. Please refer to the Multiple TTS Engines Tutorial:

http://developer.voicegenie.com/tutorials_VoiceGenie.php?tutorial=multiple_tts_engine

- Support for new audio formats: .au format and raw PCM.
- Improved TTS error handling with new error events (error.tts.noresource, error.tts.unknownengine, error.tts.badcapability)
- Improved trace file and alarm logging
- Added support for transferaudio attribute in the <transfer> tag. VoiceGenie now fully support <transfer> element as specified in the VoiceXML 2.0 Specification. Please refer to the <transfer> tutorial:
http://developer.voicegenie.com/tutorials_VoiceGenie.php?tutorial=transfer_v2
- Added attribute detectansweringmachine in the <transfer> tag
- Added basic support for RLT and TBCT (Initial Support)
- Added size-based trace file rotation (using environment variables)
- Added -v and -? command line options for all binaries to show the version information and list of command line options, respectively.
- Supports assignment of <record> variable to another variable
- New configuration parameters in interpreter configuration file (voicexml.cfg)
 - HTTP_EXTRA - extra HTTP header that will be appended at the end of the HTTP request header
 - ASR_DEFAULT_LANGUAGE - sets default ASR language if xml:lang is not specified in <grammar> or <vxml>
 - TTS_DEFAULT_LANGUAGE - sets defaults TTS language if the xml:lang attribute is not specified in <prompt> or <vxml>

Changes

- Updated Health Monitor message format for TTS Manager. For each TTS Engine configured in the TTS Manager, the Health Monitor will show the following string:
WATSON 3(1)/3 free, 0 died, PL 10 SP 5 ER 0.
The above message tells there are 3 TTS clients out of a total of 3 TTS clients are free for processing. At the busiest time there was only 1 available TTS client left for processing. PL means the number of plays requested. SP means the number of stops requested. ER means the number of errors occurred.
- TTS Manager reads from a new configuration file

- With the new supports for assignment of <record> variable to another variable, the content of <record> variable is no longer a path to the temporary recording file name. Use the shadow variable record\$.dest instead.
- When collecting user input, DTMF bargein is only allowed when there is a DTMF grammar defined and the property "inputmodes" includes "dtmf".
- When built-in grammar is explicitly referenced using the special-purpose "built-in:" URI scheme, the field variable result conforms to the VoiceXML 2.0 specification instead of returning ASR-specific result.
- Application variable application.lastresult\$[i].interpretation has been changed from a string to ECMAScript object. Field variable can become an ECMAScript object depending on the grammar. Please refer to the tutorial for detailed information:
http://developer.voicegenie.com/tutorials_VoiceGenie.php?tutorial=lastresults
- The <log> element does not add extra space between text and child elements. For example, <log>Today is <value expr=" '2000' " /></log>
- The output text will be "Today is2000"
- Changed configuration parameters in the interpreter configuration file (voicexml.cfg)
- HTTP_ACCEPT - Replaces (instead of appends) accept field in HTTP request header. Please specify the full accept field including keyword "Accept:".

Bug Fixes

- Stops ASR if busy on a new operation (fixes record bug)
- If answering machine is detected, still route lines together
- Do not disconnect line if DNIS was not available
- Fixes in shutdown and restarting of Call Manager
- Do proper checking during initialization to avoid D Channel error
- Added recovery for some errors that may occur with Robbed Bit T1
- Fixed bug in Speechify client to handle errors properly
- Fixed core dump problem during ASR manager shutdown
- Fixed bug that wrong incorrect log messages during unload for ASR Manager
- Support <audio> and Speech Markup inside the <choice> tag
- Properly set the encoding and xml:lang for grammars and TTS Speech Markup
- Fixes problem with modal attribute in <record>
- Prompt properly according to inputmode for <choice>/<option> for <enumerate>
- Removed double space in HTTP header
- Allow an attribute value to span multiple lines

- Support float type for duration
- Set boolean value instead of string value for an input item with boolean built-in ASR grammar
- Throw an error when there is a space within an audio URI instead of infinite loop
- Document variables are accessible via both application.varName and document.varName in root page
- Fix java script bugs where java objects may be lost when jumping to different scopes

Release 5.0

Changes

- Added accept attribute in <choice> and <menu>
- Added mode and weight attributes in <grammar>
- Added maxspeechevent property
- Throws maxspeechevent event
- Support document transitions between root/leaf/subdialog as defined in VoiceXML 2.0
Supports new languages (French, German, UK English) for ASR.
- Encodes and decodes special characters before sending to an ASR engine that takes XML grammar and TTS engine that supports SSML.
- Added parameter SUPPORTED_LANGUAGE in voicexml.cfg
- Supports boolean type for <value> element
- Added property com.voicegenie.xmlencoding to customize encoding for XML grammars and speech markup for TTS
- Added property com.voicegenie.allowcallblending for Call Control Extension
- Default value for property universal is changed from all to none
- Default value for property com.voicegenie.maxrecordtime is changed from 0 to 10m
- Only sends <log> information to maintainer when property loglevel is greater than 2 or when there is any errors or warnings

Bug Fixes

- <field> shadow variable \$.interpretation will be set
- Fixed transition from leaf to root document by the <submit> element so that root document will be initialized
- Fixed a bug if target is specified on the first page
- Fixed transition from a leaf to root page while root was specified with a target
- Encode ASR return results properly before sending to JavaScript engine
- Fixed decoding special characters in VoiceXML document

- Correct property name from KEEPTMPFILESTILEND to KEEPTMPFILESTILLEND
- Do not strip new line for content inside CDATA
- Disconnect when call transferred and the other side hangs up
- Remove error message for <assign> if variable is not defined
- Clean up temporary files with <subdialog>
- Do not return "error" if # is used for DTMF built-in grammars
- Fixed evaluation of expr attribute of <value> if element is visited more than once in one prompt queue
- Fixed support for hotkey grammar in <initial> element
- Temporary files will be deleted when visiting <menu>
- Fixed speech markup so that <phoneme> tag would not be split into two if more than one attribute are specified.

Release 4.9.5

Changes

- Support for *Call Blending*; this is an implementation of one of the call control proposals currently before the W3C, and includes a rich set of new tags supporting overall call control. See the Call Control Tutorial on the Developer Workshop site.
- Support for 'offsetexpr' and 'lengthexpr' attributes for the <audio> tag;
- Support for playback of a <record> form item variable using the <audio> tag;
- Support for saved utterance when performing a recognition; this is available as a shadow variable (both Nuance and SpeechWorks);
- Support for maxage and maxstale caching parameters (<vxml> version needs to be 2.0);
- Support for a-law audio with SpeechGenie TTS;
- Added support for field shadow variables \$.bargin, \$.audiooffset, \$.utteranceaudio
- Set field shadow variables \$.inputmode, \$.bargin, \$.barginresult, \$.audiooffset, \$.utteranceaudio even for nomatch if applicable

Bug Fixes

Support resolution of the relative URI in a <catch> against activated document instead of where it's declared;

- Fix a bug for alternative tts larger than 700 bytes.
- Fix a bug for supporting "connecttimeout" for <transfer>;
- Fix problems with WAVE file audio control;
- Fix a bug for filling more than one field with same slot name;
- Fix a bug for slot confidence for SpeechGenie;
- Fix a bug for <submit> to support URI fragment;
- Fix a bug when submitting regular variables after a recorded variable;

- Fix a bug when submitting binary data using multipart/form-data; we now add "filename" directive;
- Resolve URIs with "./" or "../" properly;
- Support post-processing SpeechWorks ASR results for built in grammars;
- Fix a bug when interrupting a transfer before the extension number is dialed;
- Restructure the way we do business with Watson ASR to avoid a memory leak in the ASR server;
- On call transfer without extension, don't wait for analysis on hangup request;
- Check bargein correctly for adding beep audio;
- Handle queued DTMF properly on entry to an input item;

Release 4.9.0

Changes

This release of the VoiceGenie interpreter includes a number of new features:

- Support for the 'dtmf' attribute of the <link> tag;
- Better platform error logging, better error event support;
- Add support for <enumerate> and speech markup inside <audio>; (without SRC attribute, or as an alternate prompt with SRC attribute);
- Support for nested markup tags; we do not support <audio>/<value> inside speech markup any longer;
- Support empty "event" attribute for <catch>;
- Added properties COM.VOICEGENIE.DUMPAUDIO, COM.VOICEGENIE.DUMPPARAMETER (Watson ASR);

Bug Fixes

- Fixed a bug in ABNF -> Nuance grammar conversion;
- Bug fix for <audio> with multiple alternate prompts;
- Some minor bug fixes and more error logging with speech markup;
- Support queuing a prompt [in executable content] more than once before playing it out;
- Fix a bug in <audio> when a <value> is specified inside <audio>;
- Use CR and LF, instead of just LF, in HTTP requests;
- Fix a bug in <subdialog> when "name" attribute is not specified;
- Bug fix for speech markup (Watson and NaturalVoices);

Release 4.8.2

Changes

This release of the VoiceGenie interpreter includes a number of features that have been requested by customers:

- Support "bargeintype", energy, speech and recognition;
- Support for slot level confidence score (both Nuance and SpeechWorks);
- Full a-law support; complete E1 support;
- Much richer event support and status returns for <transfer>;
- DNIS now NANA if unavailable;
- Added dest and destexpr attributes to <record>;
- Property controlling maximum record time, shadow variables for destination and maximum exceeded;
- Support for AT&T NaturalVoices TTS! (New Watson);
- Improved Speech Markup support for Old Watson TTS;
- Preliminary support for new AT&T Watson ASR;
- Complete N-best support for Nuance and SpeechWorks;
- Recognition based bargein for Nuance and SpeechWorks;
- A number of important bug fixes;
- Partial support for 'dest' per RFC 2806;
- <log> information is now added to the maintainer e-mail;
- Version control interface for VoiceXML 2.0;
- Language support;

Bug Fixes

- Support "s", "ms" and "m" units for "COM.VOICEGENIE.MAXRECORDTIME" property (PR 136);
- Fix a problem with the <transfer> attribute "connecttimeout";
- Fix a problem regarding nested <if> elements (PR 151);
- Attribute src now takes precedence over expr in <audio>;
- Ignore <audio> element if "expr" attribute evaluates to null;
- Support new return value "maxtime_disconnect" for <transfer> if bridge transfer;
- Support per-slot confidence for Nuance and SpeechWorks (PR #127);
- Support bargeintype (speech, energy, recognition) for Nuance ASR;
- Add more information in lastresult\$;
- Some internal/shadow variables are initialized as "undefined" instead of empty string;
- Throw "error.grammar.dtmf" event if parsing a dtmf grammar returns an error;
- Add support for xml:lang in <prompt>, <vxml>, <grammar>;
- Throw error.unsupported.language event if "xml:lang" is not "en-US";
- Correct problem with retrieval of xml:lang value for <grammar>;
- Use default version "1.0" for inline xml grammar when version is not specified;
- Add VoiceXML version control interface (1.0, 2.0, etc.);
- Support SSML tags if version="2.0" is specified in <vxml> tag;
- Remove '.' from generated hostnames (PR 138);

- Break long HTTP header for "Accept" over multiple lines (PR 114);
- Fix a problem with a leading "|" in ABNF grammar parsing (PR #129);
- Fix a problem with built-in Nuance digit grammar (PR #149);
- Fix a problem in HTTP header regarding port number. If port number is specified, then it will be added in the header. If it's not specified, 80 is no longer used as the default;
- Encode space properly in HTTP requests. (PR #118);
- Correct MIME type for record file submit;
- Fix problem with double '/' in request URL; breaks certain JSP;
- Make sure ASRINITTIMEOUT >= TIMEOUT, to avoid problems in certain cases;

Release 4.7

Changes

- Support for Nuance Call Logging; using the following properties:
VGASRCALLOG: true or false, to control ASR engine logging on the platform.
VGASRRECORDUTTERANCE: true or false, to control ASR utterance recording, with or without VGASRCALLOG=true.
VGASRCONFIDENTIALUTTERANCE: false or true, to indicate if utterance should be logged or not;
- Full a-law audio format support;
- Extended maximum record time – a historical Dialogic limitation for recording has been removed, so we now support longer record times. The limit is now set by the property com.voicegenie.maxrecordtime. This is only allowed in the defaults.vxml on the platform (to avoid Denial of Service attacks).
- If both <param> and 'expr' are specified , the <param> reference is used to set the variable in a subdialog;
- You can now assign to a variable without declaring it first;
- <script> can now be a child of <form>;
- New property 'fetchaudiominimum' specifies the minimum length of time that an audio file will play during a <submit> or <goto> (this is the same as the old 'fetchaudiotime');
- Support <throw> with attributes 'eventexpr', 'message' and 'messageexpr', per VoiceXML 2.0;
- Support anonymous variables '_event' and '_message' with <catch>, per VoiceXML 2.0;
- Support attribute 'label' for <log>;
- Support shadow object for ASR: application.lastresult\$

Bug Fixes

- Nested ECMAScript objects now correctly submitted as o.f1, o.f2, etc;
- We now advertise acceptance of content type 'text/plain';

- The Content-length: header now has only a single space between the header and the value (two spaces broke some web servers);
- <subdialog> fetchaudio now terminates correctly when the page is available (rather than playing to completion);

Release 4.6

Changes

- Only add "#" to URL when target is specified;
- Changes to support queued prompts for executable content, including barge-in;
- More support for GenieTracer (if you haven't tried it, you should!);
- N-best support for Nuance;
- Better support for 'file:' scheme in <dtmf>, <grammar>, <audio>
- Support for SSML with <value mode=tts class=...>
- Support <say-as> (and) <sayas> for SSML support;
- For class=date or time in <value mode=tts...>, use type=date:ymd or type=time:hm with <say-as> tag in SSML;
- Added attribute UIDATA to <transfer>

Bug Fixes

- Fixed a problem with playing certain types of WAVE files;
- Fixed some bugs related to EXPR support in <goto> and <submit>;
- Fixed a problem where the last prompt timeout was used for input collection;
- Fix for <goto nextitem>; we now always select and queue the prompt even if the nextitem has been visited already;

Release 4.5

Changes

- Support for cookies;
- Added <log> tag;
- Added support for EXPR attribute in <grammar>, <dtmf> and <audio>;
- Added support for fetchaudio with <subdialog>;
- Added attributes ANALYSIS and CONNECTWHEN to <transfer>;
- Throw `error.unsupported.format` event for unknown ASR/DTMF grammar types;
- Remove fragment following '#' from URL during Web fetch;
- Handle grammar rule name specified after '#';
- With <field type=...>, support post processing of ASR or DTMF result with a predefined ECMAScript if it exists, for VoiceXML conformance etc.; DTMF output is processed for conformance even without the ECMAScript file;

- Added support for more literals in `<value mode=recorded...>`, and handle '?' in date string for `class=date`;
- Throw `error.asr` instead of `error.application` on ASR error;
- Added support for ASRENGINE MSSR, incorporated changes done for NT port;
- Added support for ASRENGINE SPEECHWORKS and LSS;
- Added support for inline XML grammar for SPEECHWORKS;
- Added support for n-best with property MAXNBEST, and field shadow variables `<name>$.nbestresult` and `<name>$.nbestconfidence`. This is currently supported only for SpeechWorks and Nuance ASR;
- Subject line in Maintainer e-mail is now 'VoiceGenie VoiceXML Log'; still from PW_TRAP_IP;

Bug Fixes

- Corrected a few issues with outbound calling;
- Tested with more complete range of Dialogic cards;
- Fixes related to answering machine detection;
- Better behavior if a TTS server fails;
- Fixed a bug related to DTMF input processing with `TERMTIMEOUT > 0`;
- Fixed some caching related bugs and changed behavior so safe caching is strictly observed even on same page;
- With `<field type=...>`, allow DTMF or ASR built in grammar to be active if `<grammar>` or `<dtmf>` is inside `<field>`, respectively;
- Fixed bug in form submit: if no fetchaudio to play, do not start audio;
- Fixed bugs related to fetchaudiotime (builtin, submit) and fetchaudiodelay;
- Fix a bug with `<enumerate>` when `inputmodes=dtmf`;
- Fixed a bug where `<return>` in a subdialog did not have `namelist` or `event`.

Release 2.4

Changes

- Support for Speechify TTS and SpeechWorks ASR (internal release only);

Release 2.3

Changes

- A number of logging enhancements are now in place;
- Some issues regarding ASR control and performance have been addressed;
- DTMF bargein behavior has been normalized;
- We have added audio control support (see "Usage Notes");
- There have been some improvements and extensions to our release of VoiceXML;
- There have been some improvements to the platform implementation;
- A number of enhancements supporting future releases have been added;

Bug Fixes

- Handle errors during <value> tag processing;
- Handle null URI properly;
- Handle HTTP error for VoiceXML pages properly;
- Fixed bug in reading HTTP header from a fetched ECMAScript file;
- Fixed problem related to subdialog - if same audio used in root document and the page that uses the root, it would kill vxmli;
- Fixed event in <link> - use context from where link is invoked instead of where it is defined;
- Check and throw error if grammar type is unsupported;
- Fixed bug with MSECS attribute value in <break>, and RANGE attribute in <pros>;
- Use TTSENGINE to add speech markup reset tags as necessary.

Release 2.2

Changes

- Support for Speech markup;
- Default event handlers improved;
- Improved grammar support for <choice> and <menu>;

Bug Fixes

- The parser now supports the entity 'apos';
- Handle small input timeout better;
- If first element is not a <vxml>, throw "error.semantic", instead of simply hanging up;
- ABNF grammar problem with <dtmf>; when there is an ambiguous match (e.g., 1 | 11) it will recognize only the shorter one - the workaround is to set TERMTIMEOUT property to 2s: this has been fixed;
- <choice> entries with no event/expr/next attribute are now handled properly;
- Handle null URI properly;
- Fixed <reprompt> with <menu>;
- Handle illegal/infinite loop in <form>;
- Bugfix related to subdialog - if same audio used in root doc and the page that uses the root, it would kill vxmli;
- Fixed event in <link> - use context from where link is invoked instead of where it is defined;
- Fixed bug in reading http header from a fetched ECMAScript file;
- Fixed bug in <record>: set field item variable on hangup so it could still be submitted in `telephone.disconnect.hangup` event;
- Fixed bug with MSECS attribute value in <break>.

