VoiceGenie VoiceXML Gateway

Release Document

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Version: 5.1
Release Date: December 7, 2001

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Summary

Overview

This release document includes the following information:

- Product Version Identification
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This document applies to the VoiceGenie VoiceXML Gateway product, release 5.1.

Documents

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

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 Development	http://developer.voicegenie.com/IDE.php

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

For questions and problems, you can send email to the VoiceGenie Customer Support Team: sg_support@voicegenie.com

For 24/7 support, please contact VoiceGenie.

Contacting VoiceGenie

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Product Version Identification

Product Name	VoiceXML Gateway
Version	5.1
Release Date	December 7, 2001
Nuance ASR	Version 7.0.4 SP17
Text-To-Speech (TTS)	AT&T Natural Voices 1.0
Operating System	SCO UnixWare 7.1.1 Departmental Edition

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_engines
- 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_LANGUGAGE - 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

The following are changes from the previous VoiceXML Gateway Version 5.0. Please note that some of the following changes may affect applications written for previous versions of SpeechGenie.

- 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

The following are bug fixes since Version 5.0:

- 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

Restrictions

The following restrictions apply to this release:

1. error.tts.badcapability is not used in this release.

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 known problems exists:

- DTMF input recognized as help event
- Cannot access the "restricted_phone_no" variable with call_blending
- Cannot get "near_end_disconnect" through the log file.we can't get log file after running call."get event of call" is often played within call_blending
- Cannot catch "com.voicegenie.call.answered.Ok" event when calling a modem phone number.connecttimeout does't work with callblending.
- <prosody> tag does not work properly
- No different between <break> size
- <ruleref> tag in XML Grammar does not accept built-in grammars
- Playing .vox or .ulaw files do not work with the content type audio/basic. To workaround it, use the content type audio/vox instead.

Previous and outstanding known problems:

No.	Summary	Type
59	Event catch test	Feature Not Supported
65	Recognition of date values test failed	Incorrect Functionality
76	Speech markup does not work within an <audio> ... </audio> tag pairing.	Incorrect Functionality
86	Health monitor infinite loop	Incorrect Functionality
105	Only second utterance that matches the second grammar is recognized	Incorrect Functionality
123	recognition performance deteriorates very much with "timeout"	Incorrect Functionality
132	User events don't support the count attribute	Incorrect Functionality
139	Digit grammar still has spaces??	Incorrect Functionality
153	Broken Playback controls	Incorrect Functionality
155	Outbound calling on 5ESS doesn't work	Incorrect Functionality
156	Call Manager Crash	Incorrect Functionality
180	Fast caching is not working when going to a subdialog page.	Incorrect Functionality
182	tts failure leaves a ghost channel in call manager	Incorrect Functionality
184	Mis-leading error message	Incorrect Functionality
192	javascript exection causes hang-up	Incorrect Functionality
200	Problems with '.' after hostname	Incorrect Functionality

201	Inconsistent DTMF grammar handling	Incorrect Functionality
203	DNIS missing on SpeechGenie developer node	Incorrect Functionality
204	health monitor d channel error	Incorrect Functionality
210	SpeechGenie fetches each grammar twice	Incorrect Functionality
211	additional audio features	Feature Request
212	systems configured withou asr shouldn't keep trying to start it	Incorrect Functionality
213	VGSTOP audio control drains prompt queue until checkpoint	Incorrect Functionality
220	problems with grammarfetchhint	Incorrect Functionality
224	two <link> grammars do not work properly	Incorrect Functionality
226	outbound call billing/metrics records corrupted	Incorrect Functionality
230	Another Accept header change	Incorrect Functionality
231	ISDN DNIS unavailable causes disconnect	Incorrect Functionality
232	Inconsistency between tts voice ports with speechify	Incorrect Functionality
233	Platform startup sequencing needs some thought	Incorrect Functionality
234	Need to be able to bargein on prompts queued prior to off-page subdialog call	Incorrect Functionality
235	Need to be able to bargein on prompts queued prior to off-page subdialog call	Incorrect Functionality
236	Call Manager should monitor peak channel usage with licensing	Feature Request
237	vxmli shows 24 calls in progress, while callmgr shows none	Incorrect Functionality
241	a problem for undefined variable	Incorrect Functionality
242	Prompt timeouts with 0s don't work	Incorrect Functionality
244	Add bargintype attribute to <prompt>	Incorrect Functionality

Change History

Release 5.0

Changes

- Added accept attribute in <choice> and <menu>
- Added mode and weight attributes in <grammar>
- Added maxspeechtimeout property
- Throws maxspeechtimeout 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 Tag Table (at the end of this document), and 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 $.bargein, $.audiooffset, $.utteranceaudio
- Set field shadow variables $.inputmode, $.bargein, $.bargeinresult, $.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 multpart/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 adverstise 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 UUIDATA 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 ambigous 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>.