VOICEGENIE

VoiceGenie 7

Media Platform Release Notes

April 2005

VoiceGenie

TECHNOLOGIES INC

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Table of Contents

Overview

This release document includes the following information:

- Resources
- Where to Get Help
- Terminology
- Product Version Identification
- Features
- Known Issues
- Known Limitations

Documentation

Please refer to the VoiceGenie 7 Documentation Overview for the list of the documents shipped with the VoiceGenie 7 release.

Resources

There are many resources for developers available on VoiceGenie's Developer website http://developer.voicegenie.com.

The following lists some of the important items you can find on our website:

Resource	URL
FAQs	http://developer.voicegenie.com/faq.php http://speechgenie.voicegenie.com/faq.php
Tutorials	http://developer.voicegenie.com/tutorials_VoiceGenie.php http://speechgenie.voicegenie.com/tutorials_SpeechGenie.php
VoiceXML 2.0/2.1 Reference	http://developer.voicegenie.com/voicexml2tagref.php

Where To Get Help

VoiceGenie Customer Support

Hours of Operation: 8:30 am - 5:30 pm EST Monday to Friday *Closed on Canadian Statutory Holidays*	
Online/Website:	http://support.voicegenie.com
Email:	support@voicegenie.com
Phone:	(416) 736-4151
For 24/7 support information, please email support@voicegenie.com	

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Terminology

The following table gives definitions of some acronyms that are used throughout this document:

Acronyms	Full Definitions
ASR	Automated Speech Recognition (Engines/Technologies)
CLC	Command Line Console -- A command line interface that can be used to query information and issue commands
MRCP	Media Resource Control Protocol -- Adopted by the VoiceGenie Media Platform to control ASR and TTS resources
SRM	Speech Resource Management -- A component integrated into the VoiceGenie Media Platform to provide Speech Recognition and Synthesis functionalities to the application developers
SMC	System Management Console -- A web based tool for administering clusters of VoiceGenie VoiceXML Platforms
OA&M	Operation, Administration and Management
TTS	Text To Speech (Engines/Technologies)

The following sections may contain references to terminology that has become:

Historical Terms	New Terms
PhoneWeb Software / NeXusPoint 6.4.x Software	VoiceGenie 7 Software
Cluster Management Platform (CMP)	OA&M Framework
Voice Resource Manager (VRM)	Speech Resource Management (SRM)
VoiceGenie Management Console (VMC)	System Management Console (SMC)

Product Version Identification

Product Name	VoiceGenie Media Platform
Version	7.0 GA
Release Date	April, 2005

VoiceGenie Linux 3.0 CD information	
Build #	Build 9

Red Hat EL AS 3.0 Upgrade information	
Setup script	post_as.sh
Kickstart file	ks.cfg

The setup script will install the following packages to upgrade to VoiceGenie Linux 3.0:

- atsar_linux-1.6-1.i386.rpm
- cmp-deployer-1.0.0-10.i386.rpm
- httpd-2.0.52-1.i386.rpm
- j2sdk-1.4.2-fcs.i586.rpm
- mysql-server-3.23.58-1.i386.rpm
- unixODBC-2.2.8-2.3.0.2.i386.rpm
- unixODBC-devel-2.2.8-2.3.0.2.i386.rpm
- vg-scriptmanager-3.0.0-9.i386.rpm
- vg-setup-1.0.0-5.i386.rpm
- vg-squid-2.5.0-8.i386.rpm
- vg-tomcat-4.1.24-5.i386.rpm
- vg-xerces-2.0.0-1.i386.rpm

Media Platform Configurations:

MP Configuration #1	
Phoneweb Package	phoneweb-7.0.0-11.tar.gz
Operating System for Telephony Box	VoiceGenie Linux 3.0 or Red Hat EL AS 3.0 with VG 3.0 Upgrade
Operating System Kernel Version for Telephony Box	2.4.21-20 Enterprise
Protocol	SIP2
Telephony Packages	Nil
MP Configuration #2	
Phoneweb Package	phoneweb-7.0.0-11.zip
Operating System for Telephony Box	Windows 2000 Server or Windows Server 2003
Operating System Kernel Version for Telephony Box	Windows 2000 Server with Service Pack 4 or Windows Server 2003
Protocol	SIP2
Telephony Packages	Nil
MP Configuration #3	
Phoneweb Package	phoneweb-7.0.0-11.tar.gz
Operating System for Telephony Box	VoiceGenie Linux 3.0 or Red Hat EL AS 3.0 with VG 3.0 Upgrade
Operating System Kernel Version for Telephony Box	2.4.21-20 Enterprise
Protocol	H323
Telephony Packages	Nil
MP Configuration #4	
Phoneweb Package	phoneweb-7.0.0-11.zip
Operating System for Telephony Box	Windows 2000 Server or Windows Server 2003
Operating System Kernel Version for Telephony Box	Windows 2000 Server with Service Pack 4 or Windows Server 2003
Protocol	H323
Telephony Packages	Nil
MP Configuration #5	
Phoneweb Package	phoneweb-7.0.0-11.tar.gz
Operating System for Telephony Box	VoiceGenie Linux 3.0 or Red Hat EL AS 3.0 with VG 3.0 Upgrade
Operating System Kernel Version for Telephony Box	2.4.21-20 Enterprise
Protocol	Dialogic JCT T1 ISDN (D/480JCT-1T1)

Telephony Packages	dlgc-6.1-1.tar.gz
MP Configuration #6	
Phoneweb Package	phoneweb-7.0.0-11.tar.gz
Operating System for Telephony Box	VoiceGenie Linux 3.0 or Red Hat EL AS 3.0 with VG 3.0 Upgrade
Operating System Kernel Version for Telephony Box	2.4.21-20 Enterprise
Protocol	Dialogic JCT T1 RB (D/480JCT-2T1, Does not detect busy)
Telephony Packages	dlgc-6.1-1.tar.gz
MP Configuration #7	
Phoneweb Package	phoneweb-7.0.0-11.tar.gz
Operating System for Telephony Box	VoiceGenie Linux 3.0 or Red Hat EL AS 3.0 with VG 3.0 Upgrade
Operating System Kernel Version for Telephony Box	2.4.21-20 Enterprise
Protocol	Dialogic JCT E1 ISDN (D/600JCT-1E1)
Telephony Packages	dlgc-6.1-1.tar.gz
MP Configuration #8	
Phoneweb Package	phoneweb-7.0.0-11.tar.gz
Operating System for Telephony Box	VoiceGenie Linux 3.0 or Red Hat EL AS 3.0 with VG 3.0 Upgrade
Operating System Kernel Version for Telephony Box	2.4.21-20 Enterprise
Protocol	Dialogic JCT E1 CAS (D/600JCT-1E1, Does not detect busy)
Telephony Packages	dlgc-6.1-1.tar.gz
MP Configuration #9	
Phoneweb Package	phoneweb-7.0.0-11.tar.gz
Operating System for Telephony Box	VoiceGenie Linux 3.0 or Red Hat EL AS 3.0 with VG 3.0 Upgrade
Operating System Kernel Version for Telephony Box	2.4.21-20 Enterprise
Protocol	Dialogic JCT T1 SIP C/C (D/480JCT-2T1)
Telephony Packages	dlgc-6.1-1.tar.gz
MP Configuration #10	
Phoneweb Package	phoneweb-7.0.0-11.zip
Operating System for Telephony Box	Windows 2000 Server or Windows Server 2003
Operating System Kernel Version for Telephony Box	Windows 2000 Server with Service Pack 4 or Windows Server 2003
Protocol	Dialogic DMV T1 ISDN (DMV480-A, 960-A, 1200-B)
Telephony Packages	DialogicSR6.0.zip
MP Configuration #11	
Phoneweb Package	phoneweb-7.0.0-11.zip

Operating System for Telephony Box	Windows 2000 Server or Windows Server 2003
Operating System Kernel Version for Telephony Box	Windows 2000 Server with Service Pack 4 or Windows Server 2003
Protocol	Dialogic DMV T1 RB (DMV480-A, 960-A, 1200-B)
Telephony Packages	DialogicSR6.0.zip
MP Configuration #12	
Phoneweb Package	phoneweb-7.0.0-11.zip
Operating System for Telephony Box	Windows 2000 Server or Windows Server 2003
Operating System Kernel Version for Telephony Box	Windows 2000 Server with Service Pack 4 or Windows Server 2003
Protocol	Dialogic DMV E1 ISDN (DMV1200-B, 600)
Telephony Packages	DialogicSR6.0.zip gc.zip
MP Configuration #13	
Phoneweb Package	phoneweb-7.0.0-11.tar.gz
Operating System for Telephony Box	VoiceGenie Linux 3.0 or Red Hat EL AS 3.0 with VG 3.0 Upgrade
Operating System Kernel Version for Telephony Box	2.4.21-20 Enterprise
Protocol	Brooktrout T1 ISDN (TR1000)
Telephony Packages	brkt-tr1000-3.2.2-1.tar.gz
MP Configuration #14	
Phoneweb Package	phoneweb-7.0.0-11.zip
Operating System for Telephony Box	Windows 2000 Server or Windows Server 2003
Operating System Kernel Version for Telephony Box	Windows 2000 Server with Service Pack 4 or Windows Server 2003
Protocol	Brooktrout T1 ISDN (TR1000)
Telephony Packages	TR1000-v3.2.2-Win.zip
MP Configuration #15	
Phoneweb Package	phoneweb-7.0.0-11.tar.gz
Operating System for Telephony Box	VoiceGenie Linux 3.0 or Red Hat EL AS 3.0 with VG 3.0 Upgrade
Operating System Kernel Version for Telephony Box	2.4.21-20 Enterprise
Protocol	Brooktrout T1 RB (TR1000)
Telephony Packages	brkt-tr1000-3.2.2-1.tar.gz
MP Configuration #16	
Phoneweb Package	phoneweb-7.0.0-11.zip

Operating System for Telephony Box	Windows 2000 Server or Windows Server 2003
Operating System Kernel Version for Telephony Box	Windows 2000 Server with Service Pack 4 or Windows Server 2003
Protocol	Brooktrout T1 RB (TR1000)
Telephony Packages	TR1000-v3.2.2-Win.zip
MP Configuration #17	
Phoneweb Package	phoneweb-7.0.0-11.tar.gz
Operating System for Telephony Box	VoiceGenie Linux 3.0 or Red Hat EL AS 3.0 with VG 3.0 Upgrade
Operating System Kernel Version for Telephony Box	2.4.21-20 Enterprise
Protocol	Brooktrout E1 ISDN (TR1000)
Telephony Packages	brkt-tr1000-3.2.2-1.tar.gz
MP Configuration #18	
Phoneweb Package	phoneweb-7.0.0-11.tar.gz
Operating System for Telephony Box	VoiceGenie Linux 3.0 or Red Hat EL AS 3.0 with VG 3.0 Upgrade
Operating System Kernel Version for Telephony Box	2.4.21-20 Enterprise
Protocol	Brooktrout E1 ISDN SIP Clear Channel (TR1000)
Telephony Packages	brkt-tr1000-3.2.2-1.tar.gz

Speech Resource Manager Configurations

SRM information	
SRM Packages (Linux)	srm-server-7.0.0-11.tar.gz srm-proxy-7.0.0-11.tar.gz
SRM Packages (Windows)	srm-server-7.0.0-11.zip

SRM Server Configuration	
Operating System	VoiceGenie Linux 3.0 or Red Hat EL AS 3.0 with VG 3.0 Upgrade or Windows 2000 Server with Service Pack 4 or Windows Server 2003

SRM Proxy Configuration	
Operating System	O/S: VoiceGenie Linux 3.0 or Red Hat EL AS 3.0 with VG 3.0 Upgrade

Speech Resource Configurations

ASR Configurations

ASR Configuration #1	
ASR name and version	Phonetic GN 5.8.002x Native
ASR Packages	GN_5.8.01_0022.zip (Server Package (Vendor)) phonetic-base-5.8.0-5.zip (Client Package)
OS for ASR server	Windows 2000 Server or Windows Server 2003
ASR Configuration #2	
ASR name and version	Nuance MRCP 1.0 SP8
ASR Packages	MRCP-1-0-0-SP8-WIN2K.zip (Server Package (Vendor)) nuance-mrcp-8.5.0-2.zip (Windows Client Package) nuance-mrcp-8.5.0-2.tar.gz (Linux Client Package)
OS for ASR server	Windows 2000 Server or Windows Server 2003
ASR Configuration #3	
ASR name and version	OSR 3.0.3 Native
ASR Packages	osr-base-3.0.3-3.tar.gz (Server Package) osr-hotfix-3.0.3-1.tar.gz (Required Patch)
OS for ASR server	VoiceGenie Linux 3.0 or Red Hat EL AS 3.0 with VG 3.0 Upgrade
ASR Configuration #4	
ASR name and version	OSR 3.0.3 MRCP
ASR Packages	OSR-3.0.3-I386-win32.zip (Server Package (Vendor)) SWMS-3.1.3-I386-win32.zip (SWMS Package (Vendor)) osr-mrcp-3.0.3-3.tar.gz (Linux Client Package) osr-mrcp-3.0.3-3.zip (Windows Client Package)
OS for ASR server	Windows 2000 Server or Windows Server 2003

TTS Configurations

TTS Configuration #1	
TTS name and version	Rhetorical TTS 4.3 Native
TTS Packages	rhetorical-tts-base-4.3.0-4.tar.gz (Server Package)
OS for TTS server	VoiceGenie Linux 3.0 or Red Hat EL AS 3.0 with VG 3.0 Upgrade
TTS Configuration #2	
TTS name and version	Nuance Vocalizer 4.0 MRCP
TTS Packages	VOCALIZER-4-0-1-WIN32-8-5-0-SP040820.zip (Server Package (Vendor)) vocalizer-mrcp-4.0.0-2.tar.gz (Linux Client Package) vocalizer-mrcp-4.0.0-2.zip (Windows Client Package)
OS for TTS server	Windows 2000 Server or Windows Server 2003
TTS Configuration #3	
TTS name and version	ScanSoft RealSpeak 4.0.4 Native
TTS Packages	realspeak-base-4.0.4-5.tar.gz (Server Package)
OS for TTS server	VoiceGenie Linux 3.0 or Red Hat EL AS 3.0 with VG 3.0 Upgrade
TTS Configuration #4	

TTS name and version	ScanSoft RealSpeak 4.0.4 Native
TTS Packages	Vendor Windows Server Package + realspeak-base-4.0.4-5.zip (Server Packages)
OS for TTS server	Windows 2000 Server or Windows Server 2003

Other Packages

TTY/RTSP:

Custom Packages information	
RTSP	rtsp-7.0.0-6.tar.gz
TDD/TTY	tty-7.0.0-5.tar.gz

Windows 2000/Windows 2003 Third Party Software Information

Package information	
Third Party Software	This is shipped as a CD which supports the installation of the following third party packages: • Windows Script Host 5.6 • MySQL 3.23 • MyODBC 3.51.06 • Java 2 SDK 1.4.2 • Apache Tomcat 5.0.18 • Squid 2.5

Features

The following new features are introduced with this release:

Ability to Trap Specified SIP Messages

Description
The VoiceGenie Platform provides the capability to log SIP messages (on demand) to a log file or to the OA&M Framework database. These logs contain the information which allow users to isolate all logs (via Call-ID) related to a particular call.

Additional ISDN Cause Code Support

Description
The following ISDN cause codes are recognized as “bad destination”

Incorrect number	28
No longer in service	27
Number is changed	22
Invalid number	01, 18, 29, 50, 54

The following ISDN cause codes are recognized as “network busy”

Switching equipment congestion	42
Invalid transit network selection	91

The following ISDN cause codes are recognized as “unknown”

Interworking, unspecified	127

Alternate Initial Page on DNIS-basis

Description
Previously, there was a configuration parameter in the VoiceXML Interpreter configuration file that specified an alternate initial page to load when the first page could not be loaded by the VoiceXML Interpreter. This alternate page could only be specified on a per-platform basis (i.e. using same alternate URI).

In this VoiceGenie Media Platform release, administrators are able to choose the alternate initial page to be used on a per-DNIS basis.

Please see the Media Platform System Reference Guide for further details.

Application Count Service

Description
This feature restricts the maximum number of concurrent calls per VoiceXML application.

Please see the Media Platform System Reference Guide for further details.

Audio Control with DTMF barge-in

Description
This feature enables audio control operations to be executed when barge-in is enabled in the application.

In some voicemail architectures this feature will increase flexibility by allowing the user to perform Audio Control operations while being able to choose other options using DTMF or ASR input at the same time.

Please see the Media Platform System Reference Guide for further details.

Authenticated SIP Registration

Description
SIP Registration is supported in the VoiceGenie Media Platform. It is possible that the platform be required to be integrated in an environment which requires HTTP Authentication in the SIP registration message. This feature accommodates that as specified in Section 22 of RFC 3261 and RFC 2617.

Please see the Media Platform System Reference Guide for further details.

Change of meaning to ENGINES_FETCH_GRAMMAR

Description
In the VoiceXML Interpreter, there is a configuration parameter ENGINES_FETCH_GRAMMAR which is a list of ASR engines. For those engines in this list, for the external grammar specified via <grammar src= "…."/>, the VoiceXML Interpreter sends the URI for the grammar to the ASR engine. This configuration parameter was designed at the time when most ASR engines did not support native fetching of grammar. If an ASR engine's name did not appear in this list, then the VoiceXML Interpreter would fetch the URI, and send the grammar content to the ASR engine. On the other hand, if an ASR engine's name was in the list, then VoiceXML Interpreter would simply instruct the ASR to obtain the grammar from the URI.

Currently, most ASR engines supports fetching and this is the default operation mode. Whenever a new ASR engine is added to the VoiceGenie offering, the engine name is required to be added to the ENGINES_FETCH_GRAMMAR list. The VoiceGenie Media Platform has been modified in the following way:

- The ENGINES_FETCH_GRAMMAR parameter has been deprecated
- A new VoiceXML Interpreter configuration parameter, VXMLI_FETCH_GRAMMAR_ENGINES, has been added
- This new parameter has the opposite meaning as the deprecated ENGINES_FETCH_GRAMMAR parameter - that is, if an ASR engine name appears in this list, then VoiceXML Interpreter will fetch the grammar and pass the grammar to the ASR engine, instead of simply sending the grammar URI to the ASR engine.

Please see the Media Platform System Reference Guide for further details.

Change of meaning to SSML_ENGINE configuration parameter

Description
In the VoiceXML Interpreter there is a configuration parameter, SSML_ENGINE, which is a list of engines that support SSML_ENGINE. This configuration parameter was previously designed at the time when most TTS engines did not support SSML. If a TTS engine's name did not appear in this list, then the VoiceXML Interpreter would strip away any SSML tags embedded in the text to be spoken. On the other hand, if a TTS engine's name was in the list, the VoiceXML Interpreter would wrap the top-level SSML tags <speak> and </speak> around the entire text block (including the SSML markup tags) to be spoken and send it to the TTS engine.

Currently, most TTS Engines support SSML. Whenever a new voice which supports SSML is added to the VoiceGenie offering, the new voice is required to be added to the SSML_ENGINE parameter. The VoiceGenie Media Platform has been modified in the following way:

- The SSML_ENGINE parameter has been deprecated
- A new VoiceXML Interpreter configuration parameter, NON_SSML_ENGINES, has been added
- The NON_SSML_ENGINES parameter has the opposite meaning as the deprecated SSML_ENGINE parameter - that is, if a TTS engine name appears in this list, then VoiceXML Interpreter will strip away any SSML tags embedded in the text to be spoken

Please see the Media Platform System Reference Guide for further details.

Changing trace level for ASR/TTS clients on the fly

Description
This feature adds functionality in the SRM server whereby the SRM server can send information to the ASR/TTS clients to change trace levels without re-starting the ASR/TTS clients.

Please see the Media Platform System Reference Guide for further details.

Clearing SRM Health Statistics via the CLC

Description
This feature adds support in each component managed by the OA&M Framework, whereby the health statistics for the specific component can be cleared using a generic Command Line Console (CLC) command. This is currently only available for the SRM component.

Please see the OA&M Framework System Reference Guide for further details.

Correlation ID Extraction from DNIS

Description
This feature allows the correlation ID to be extracted from the DNIS in a SIP message (From: field) and be exposed at the VoiceXML application level. The DNIS will have the correlation ID component stripped out after the extraction. Hence, DNIS from the SIP message will be separated into the correlation ID and the DNIS session variables.

Please see the Media Platform System Reference Guide for further details.

<data> element

Description
The <data> element allows a VoiceXML application to fetch arbitrary XML data from a document server without transitioning to a new VoiceXML document. This will be exposed only if the <vxml version> attribute is set to 2.1 or higher.

Please see the Media Platform System Reference Guide for further details.

Defer Platform Output Until RTP Stream is Active

Description
In the RTP protocol there is no way for one end of a stream to know when the other end of the stream is ready. Hence, some packets which are sent at the beginning of an RTP stream will get lost. This feature alleviates this restriction by having one end wait until it receives a packet from the other end before sending out any information.

Please see the Media Platform System Reference Guide for further details.

Digit Input Buffering Before Call is Connected

Description
This feature provides a means of handling a DTMF input before a call is connected. Usually, the DTMF input is ignored until a call is connected. This feature allows the input to be buffered and sent when the call is connected.

Please see the Media Platform System Reference Guide for further details.

DTMF Support via SIP INFO

Description
This feature provides DTMF input as part of the SIP INFO message body during a SIP session which is particularly useful in situations where a DTMF input can be used for Call Control capabilities even when the media path is not connected to the VoiceGenie Media Platform.

Please see the Media Platform System Reference Guide for further details.

Expose Channel ID to VoiceXML Application for SIP/CC

Description
Under clear channel mode the channel ID either comes into the VoiceGenie Media Platform inside the SIP INVITE message (for an inbound call) or is embedded inside the SDP during media association (for an outbound call). This feature extracts and delivers this information to the VoiceXML application.

Please see the Media Platform System Reference Guide for further details.

Exposure of Platform IP address in shadow variable

Description
A feature allows the Platform IP address for a running VoiceXML page to be exposed to the VoiceXML application, via a builtin JavaScript function.

Please see the Media Platform System Reference Guide for further details.

Fixed Gain Control for Full Call Recordings (FCRs)

Description
Incoming speech in certain environments can have significantly lower amplitude levels than platform output. After mixing with a much louder platform output for Full Call Recording, the signal becomes unintelligible. This feature configures the Full Call Recording mixer so as to apply variable gain to the incoming signal and compress the dynamic range of the mixed signal to protect it from clipping.

Please see the Media Platform System Reference Manual for further details.

Hotword Detection for Telisma

Description
This feature introduces the functionality in the SRM client whereby it can write special vendor-specific parameters to enable Hotword detection for the TeliSpeech MRCP Server. The application would use the standard “bargeintype” property to control this feature.

Please see SRM System Reference Guide for further details.

H.323 Multiple End-point Support

Description
H.323 is a standard that specifies the components, protocols and procedures that provide multimedia communication services over packet networks. The implementation is specified in the ITU-T Recommendation. The VoiceGenie Media Platform follows the ITU-T recommended H.323 implementation. However, some vendors implement proprietary H.323 features which hinder inter-operability. This feature provides multiple end-point inter-operability for Avaya H.323 products.

Please see the Media Platform System Reference Guide for further details.

Introduction of SRM Proxy Capabilities

Description
The SRM Proxy is a new software component that has been added to the existing VoiceGenie Media Platform.

The SRM Proxy manages the routing of speech media server requests from Clients to Servers and forwards media server responses from Servers to Clients.

The SRM Proxy will support the following modes:

- Simple
- RTSP Media Redirect
- Streaming Relay

Simple mode is used when the SRM Proxy simply allocates a resource and passes the MRCP messages along, using standard MRCP.

RTSP Media Redirect mode is used when the MRCP client can support receiving a message from the server asking it to change its SDP information to send/receive its media to a different destination (which is implemented in the VoiceGenie SRM Client).

Streaming Relay mode is when the SRM Proxy receives and forwards the media between the Client and the Server.

If the SRM Proxy is going to be used with VoiceGenie only, then only RTSP Media Redirect mode is used.

The SRM Proxy is also responsible for handling server failure scenarios.

It has the following failover capabilities (if the SRM Proxy detects that the Server has become unavailable):

- If the client supports RTSP Media Redirect or the Stream Relay modes, then the recognition/synthesis request will fail; but not the MRCP session itself.
- Otherwise all the current and future recognition/synthesis will fail, until the client establishes a new MRCP session with the Proxy.

In the case of Proxy failure, the client will lose all the sessions currently handled by the Proxy. The client will need to connect to a back-up Proxy.

The Proxy offers round-robin resource allocation as well as least-used resource allocation. It also supports resource allocation based on Language. This will only be available if the client supports RTSP Media Redirect or Stream Relay modes.

The Proxy reports statistics via the OA&M Framework reporting facilities. It is integrated with the OA&M Framework for process startup and monitoring, configuration update, as well as other general integrated capabilities offered by the OA&M Framework. The Proxy also offers soft-shutdown capabilities.

Please see the SRM System Reference Guide for further details.

Iproxy Logging

Description

Previously, the Iproxy component of the VoiceGenie platform performed some basic caching of HTTP responses. This new feature has been introduced such that, for every HTTP fetch performed, the Iproxy must also log a transaction entry stating whether the HTTP response was generated from squid or from the shared memory cache.

Please see the Media Platform System Reference Guide for further details.

<mark> in SSML

Description
The <mark> tag is used for exposing the barge-in location information to the VXML application, once barge-in occurs. This tag is in compliance on the Media Platform with VoiceXML 2.1 specifications. This will be exposed only if <vxml version> attribute is set to 2.1 or higher.

Note: <mark> tag support is not available for all TTS engines.

Please see the Media Platform System Reference Guide for further details.

Media - Onboard Conferencing

Description
Many interesting voice applications require the use of conferencing capability in order to deliver the intended user experience. Examples of applications that can make use of conferencing include:
Enhanced voice activated dialers that allow multiple people to be called simultaneously;
Voicemail and other applications that allow outbound calls while staying on the line;
Delivery of IP Centrex conferencing services (in conjunction with a softswitch application);
and numerous others

This version of the VoiceGenie Media Platform incorporates conferencing support for VoIP users using RTP.

There are two interfaces to initiate a conference:

1. Conferencing interface via SIP

   A conference is invoked from a purely SIP perspective without the use of VoiceXML. DNIS->URL mapping needs to be configured to use the conference CMAPI application module to handle the call. The Request-URI in the SIP INVITE that establishes the conferencing session should contain a 'conference-id' parameter that specifies the conference session to join.

2. Conferencing interface via VoiceXML

   Conferencing capabilities are exposed to VoiceXML applications running on the media platform. Access to conferencing capabilities will be by way of extensions to the <join> and <release> tags supported in the current VoiceGenie Media Platform.

Please see the Media Platform System Reference Guide for further details.

Media Redirect Transfer

Description
This feature extends the capabilities of the VoiceGenie Media Platform to support the redirection of media streams. Two parties which communicate with each other via the VoiceGenie Media Platform can now stream RTP media directly to one another, rather than each maintaining separate RTP sessions with the VoiceGenie Media Platform. This feature is particularly useful in cases where applications running on the VoiceGenie platform may need to manage interactions between multiple platform sessions.

Please see the Media Platform System Reference Guide for further details.

Mozilla Spider Monkey version upgraded to version1.5 RC6a

Description
The VoiceXML Interpreter of the VoicGenie Media Platform has upgraded its Mozilla Spider Monkey version to v1.5 RC6a.

MRCP Vendor Specific Parameter

Description
Most MRCP servers have their own vendor-specific parameters. By setting these parameters, VoiceXML applications can dynamically use vendor specific features. This feature allows the application to set arbitrary vendor-specific parameters, which are sent to the MRCP server.

Please see the SRM System Reference Guide for further details.

New SIP Line Manager

Description
This new proprietary VoiceGenie SIP stack enables support of many enhanced features such as Proxy, CANCEL, redirection transfer, conference, re-INVITE and PRACK in the future. This includes updated compliance to RFC 3261 (SIP-bis), 3264 (offer/answer model based on SDP).

Please see the Media Platform System Reference Guide for further details.

Play/Record Support for G.729

Description
The RTP media transport protocol supports many different types of audio formats. The formats that are transferred in each session are negotiated by session initiation protocols such as SIP and H.323. This feature adds support to the VoiceGenie platform so that it can play and record G.729 files.

Please see the Media Platform System Reference Guide for further details.

QCELP CODEC Integration

The QCELP CODEC compresses each 20 milliseconds of a 8000 Hz, 16-bit sampled input speech into one of four different size output frames: Rate 1 (266 bits), Rate ½ (124 bits), Rate ¼ (54 bits) or Rate 1/8 (20 bits). The QCELP CODEC, because of its variable transcoding rate, is very different from all the other codecs currently supported by the VoiceGenie Media Platform, all of which have a constant sampling rate.

Please see the Media Platform System Reference Guide for further details.

Q.SIG Transfer

Description
Q.SIG is an ISDN based transfer protocol used for signaling between different PBXs. In particular, Q.SIG allows PBX products from different vendors to work together. VoiceGenie utilizes Dialogic for Q.SIG transfer support.

Please see the Media Platform System Reference Guide for further details.

Real Time Streaming Protocol (RTSP) Audio Support

Description
The Real Time Streaming Protocol (RTSP) is a popular protocol for streaming real-time media from a content server to listeners. This feature provides integration between the VoiceGenie Media Platform and some basic RTSP functionality.

Please see the SRM System Reference Guide for further details.

SDP Offer/Answer model

Description
Based on RFC 3264, the VoiceGenie Media Platform will initiate offers and generate answers that contain an “a” SDP attribute set to one of “sendonly”, “recvonly”, “sendrecv” or “inactive”, to indicate the direction(s) of the media stream(s).

Please see the Media Platform System Reference Guide for further details.

Session ID appended to “root document not found” error

Description
Introduces the functionality in the VoiceGenie Media Platform whereby the Session ID is also included in the error message “root document not found”.

Session ID included in email logs

Description
A new feature has been introduced such that, when an email notification is sent to the maintainer of an application, the session ID is also included as part of the email message.

Please see the Media Platform System Reference Guide for further details.

srcexpr attribute in <grammar> and <script>

Description
This new attribute, as defined in VoiceXML 2.1, has been added to the VoiceGenie Media Platform. Its availability is limited to when the <vxml> version is 2.1. An error.badfetch error is thrown if the ‘expr’ attribute is used when <vxml> version not is 2.1.

Please see the Media Platform System Reference Guide for further details.

SRM Client configuration to send configuration parameters

Description

This feature adds the capability in the SRM client configuration whereby it can send vendor specific parameters and values for each ASR/TTS Engine. These vendor-specific parameters are sent in the SET-PARAM method after the session is SETUP.

Please see the SRM System Reference Guide for further details.

SRM Client provisioning for server configuration

Description

Previously, the SRM Client configuration was embedded in the Call Manager configuration. This feature allows the SRM Client, using the provisioning service provided by the OA&M Framework, to configure servers.

Please see the SRM System Reference Guide for further details.

TDD/TTY Support

Description

In making telecommunications facilities available to those who are hearing impaired and cannot use normal telephone instruments, a standard was created to allow two suitably equipped parties to communicate over a standard phone line using specialized, text-based terminals on either side that provided a visual display instead of an audio-only interface. These terminals, referred to as Telecommunications Devices for the Deaf (TDD) and also known as TTYs, enable communication without the use of audio over standard telephone lines.

TDDs act as small computers with built-in modems. They have a visual indicator to present incoming calls (since an audible ring tone is not usable), a keyboard to allow typed input from the user, and a screen to display typed input received from the remote end.

This feature integrates TDD support on the VoiceGenie Media Platform.

URI (with parameters) for running page, exposed to the application

Description

A new feature has been introduced such that the URI for a running VoiceXML page, including parameters, is now exposed to the VoiceXML application via a builtin JavaScript function.

Please see the Media Platform System Reference Guide for further details.

UUI Protocol and Codeset support

Description

According to ISDN specifications the protocol field is a mandatory part of UUI IE. Hence support for protocol field is extended for all supported types of outbound calls and transfers that can pass UUI.

This feature can be disabled using configuration parameters to maintain compatibility with applications that send UUI along with the protocol field or with switches that do not require the protocol field.

The following capabilities are supported:
- Allow user to specify UUI protocol to be used for outgoing UUI IE with either configuration parameter or signal variable (with the latter taking precedence)

- Extract and remove protocol field from incoming UUI IE and pass it to an application as a parameter

- Allow user to turn off support for protocol field. In this case no protocol field will be set in outgoing UUI or extracted from incoming UUI IEs

- Allow user to turn off encoding of non-printable characters for incoming and outgoing UUI

- Add an optional Codeset parameter and signal variable with only acceptable value of '7'. With this parameter set outgoing UUI will be placed after locking shift to Codeset 7.

- When using user-defined protocol (0x00) it is the user's responsibility to format UUI content according to application needs. UUI content will be placed into a UUI IE starting from octet 4, unless support for protocol field is disabled. In this case user-provided UUI content will start from octet 3.

Potential compatibility issues with previous versions of the VoiceGenie Media Platform:

- TC_UUI_PROTOCOL configuration parameter will not be supported. pstn.uuiprotocol will replace it.

- If user-defined protocol (0x00) is selected, current implementation pre-appends user-supplied UUI data with data forwarding information (text tag). [TR50075] defines this format as a "suggested encoding" and not a requirement. This feature will be removed; an application must format UUI data as required.

Please see the Media Platform System Reference Guide for further details.

VoiceXML event generation using SIP INFO

Description
This feature provides SIP INFO messages with a certain format to the VoiceXMLI application module so that the interpreter instance may receive catchable events. Since there is no standardized format for the message body of a SIP INFO message, it will be passed in its raw text message form, and the receiver will be responsible to parse and extract the required information.

Please see the Media Platform System Reference Guide for further details.

VoiceXML 2.1 <foreach> tag

Description

As defined in VoiceXML 2.1, the <foreach> element allows a VoiceXML application to iterate through an ECMAScript array and to execute the content contained within the <foreach> element for each item in the array.

Please see the Media Platform System Reference Guide for further details.

VoiceXML 2.1 properties/variables for saveutterance

Description

Recording user utterances, while attempting recognition as specified by VoiceXML 2.1, has been added to the VoiceGenie Media Platform. This feature is supported in the <field>, <initial>, and <menu> elements.

Please see the Media Platform System Reference Guide for further details.

Known Issues

The list of issues, current as of the publication date of this document, is available to supported customers on our Support Website at http://support.voicegenie.com. This list is also updated with a list of issues that have been discovered since the publication date.

Known Limitations

The following are know limitations:

1) Audio control accuracy has been reduced to a maximum of 2s due to a limitation of the Dialogic driver
2) With MRCP ASR server error.asr.grammar.builtin will not be thrown when builtin grammar has an error. error.asr.grammar will be thrown in this case.