VoiceGenie 7.2.1

Media Platform

System Reference Guide

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Document Version: 03-2008

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Chapter

1 Introduction

1.1 Overview

This guide serves as the system reference manual for the VoiceGenie 7.2 Media Platform. It is intended to provide a complete reference for all aspects related to configuration parameters, logging/metrics entries, directory structure as well as SNMP alarms for the VoiceGenie Media Platform.

1.2 The System Management Console (SMC)

The SMC is the web-based interface to the VoiceGenie OA&M Framework. All Media Platform configuration can be performed via this interface. For details regarding how to access the SMC, and its operation, please refer to:

VoiceGenie 7.2.1 OA&M Framework – SMC User's Guide

The "Product Configuration" section of the *SMC User's Guide* will be particularly useful to those whose tasks involve configuring various components of the VoiceGenie 7.2 Media Platform.

Chapter

2 Directory Structures

This section describes the directory layout on the VoiceGenie 7.2 platform.

All software installation takes place within the `/usr/local/vg_install` directory (for linux) or `c:\VoiceGenie\` (for Windows).

The `mp` directory contains versioned package directories as delivered from VoiceGenie.

- `audio` – platform resident audio data – is a link to either `audio_alaw` or `audio_mulaw`, depending on the audio format used where the system is installed – both directories contain the same files, but with different audio encodings.
- `bin` – platform binaries
- `callrec` – used for full call recordings
- `config` – platform configuration information
- `grammar` – built-in grammar support (including a subdirectory for each engine)
- `lib` – Library files used by the platform
- `logs` – log and trace data
- `record` – used as the default directory for recordings on the system
- `samples` – simple test VoiceXML files used for initial configuration
- `tmp` – temporary files
- `utils` – setup scripts and Dialogic tools
- `utterance` – contains utterance files when the system is configured to write these to disk

The following table provides a list of the files within each of the above listed directories (they are located under the `mp` subdirectory on Windows platforms):

audio Directory or file	**vox files used to describe error conditions or that can be used as built-in audio**
backgroundaudio.vox	keyboard clicking sound used by `~config/defaults.vxml` while fetching audio files from the application server
default_audio/*	various error messages used by `~config/defaults.vxml`
dtmf/*	dtmf `.vox` and `.wav` files
effects/*	misc `.vox` and `.wav` files – `endofprompt.vox` is used by `~config/voicexml.cfg` as the audio file played to indicate to the caller when they can speak if ASR barge-in is not used
prompts/*	misc `.vox` and `.mu` files – some of which are used for the sample VXML applications in the `~samples/` directory
value/*	letter and number `.vox` files
bin Directory	**Platform binaries and scripts**
busy	Application to busy out 1 system CPU for a short period of time
get_log	Script to parse performance logs
Ks	Script to kill all processes with names containing a substring. Usage: `ks <name>` (Ex: `ks callmgr`)
proxy-sco	Proxy process
purge_logs	Delete all `pw_metricsfile,` speech and grammar files which are more than 6 days old
pwcallmgr	Call Manager binary
pwproxy	Proxy binary
pwvxmli	VXMLI binary
rdl	Legacy script to launch remdial
config	**Configuration directory**
callmgr.cfg	Call Manager configuration file (This file is overwritten when the platform is started)
cm_provision.dat	Configuration file containing dialing rules and URL-DNIS mappings (This file is overwritten when the platform is started)
defaults.vxml	Default settings that inherit to all VoiceXML pages run on the platform using the Legacy Interpreter.

Defaults-ng.vxml	Default settings that inherit to all VoiceXML pages run on the platform using the Next Generation Interpreter.
iproxy.cfg	Fetching Module configuration file
networkid.cfg	Network ID for CMP components
vglicense.txt	VoiceGenie platform license file
voicexml.cfg	VXMLI configuration file (This file is overwritten when the platform is started)
grammar/	**Grammar files**
bbn/hotkey/*	*see list below for descriptions of files common to each
nuance/hotkey/*	*see list below for descriptions of files common to each
nuance/*	Grammars used for built-in types
speechworks/hotkey/*	*see list below for descriptions of files common to each
speechworks/hotkey.gram	*see list below for descriptions of files common to each
telisma/hotkey/*	*see list below for descriptions of files common to each
watson/hotkey/*	*see list below for descriptions of files common to each
Hotkey Grammars	Grammars that recognize Cancel, Exit and Help
Cancel.en-us.grxml	
CancelExit.en-us.grxml	
Exit.en-us.grxml	
HelpCancel.en-us.grxml	
HelpCancelExit.en-us.grxml	
Help.en-us.grxml	
HelpExit.en-us.grxml	
Cancel.en-us.grxml	
CancelExit.en-us.grxml	
logs/	**Log files**
pw_logfile	Current logfile
pw_logfile.#	Old versions of the `pw_logfile.` Most recent is `.1,` next oldest is `.2,` etc.

pw_metricsfile	Current metrics file
pw_metricsfile.yyyymmdd .hhmm	Old metrics file
TRACE.<process_name>.*	trace files used for debugging
samples/	**Sample VoiceXML pages**
helloasr.vxml	Hello world page that uses ASR.
helloaudio.vxml	Hello word page that uses no ASR or TTS
hellorecord.vxml	Hello word page that uses recording
hellotransfer.vxml	Hello world page that transfers the call to VoiceGenie
hellotts.vxml	Hello world page that uses TTS.
helloworld.vxml	Hello word page that uses no ASR and TTS
utils/	**Utilities and scripts**
I101phoneweb.todo	initial configuration script used by vg-scriptmanager
checkversion	script used to check the verion of the phoneweb binaries – used by `vginfo_linux.sh`
differences.sh	used to compare the old and new versions of `~config/callmgr.cfg` and `~config/voicexml.cfg` – these files may not be copied from the older version of the software when the system is upgraded

The caching proxy resides in a directory of its own – `/usr/local/squid`. This directory includes a number of subdirectories:

- `bin` – the binaries for proxy execution
- `cache` – the cache contents directory
- `etc` – configuration files for squid
- `libexec` – support files for squid execution
- `logs` – squid log information
- `man` – manual pages for squid

Note: It is important to keep the hard disk from filling – like any typical software system, the VoiceGenie 7.2 Media Platform may not properly function once the disk spaces is exhausted.

Chapter

3 Audio/Video Formats

3.1 Supported Audio/Video Formats

The most portable file format remains a raw audio file containing data encoded as in CCITT G.711 u-law format, single-channel, with a sample rate of 8000 samples per second and 8 bits per sample. The European equivalent would use G.711 a-law encoding rather than G.711 u-law.

These files typically have a `.vox` extension. It is important for Web servers to recognize the `.vox` type and send the VoiceGenie platform a file with a MIME content type of `audio/x-vox;` this usually requires some manual configuration on the part of the HTTP server administrator.

In general, all supported audio and video codecs can be stored in raw format. The table below summarizes the expected file extension and required mime type to be returned from the HTTP server.

It is also permissible to refer to files with a `.wav` extension, which can use the same encoding but also include a `RIFF` header. In this case, the web server must deliver a MIME content type of `audio/x-wav.` These files can contain data with G.711 u-law, G.711 a-law, Linear PCM, or G.726 ADPCM encoding, and sample rates of 8000 Hz.

It is also permissible to refer to files of NIST SPHERE format, and Next/Sun AU format. In the case of NIST format files, which include a `NIST` header, the web server must deliver a MIME content type of `audio/x-wav.` These files can have an extension of either `.nist` or `.wav.` In the case of AU format files, which include an `AU` header, the web server must deliver a MIME content type of `audio/basic.` These files must have an extension of `.au.` NIST format files can contain data with G.711 u-law or G.711 a-law encoding. AU format files can contain data with G.711 u-law, G.711 a-law, Linear PCM, or G.726 ADPCM encoding.

For video files, the VoiceGenie Media Platform supports raw video-only files. These files typically have a `.263` extension. It is important to configure the Web server to recognize the `.263` type and send the VoiceGenie platform a file with a MIME content type of `video/h263` or `video/x-h263.`

For playing back audio/video content, the VoiceGenie Media Platform currently supports the avi file container format. These files can contain audio data with G.711 u-law, G.711 a-law, Linear PCM, or G.726 ADPCM encoding mixed with h263 video data.

The table below summarizes the media formats supported by the VoiceGenie Media platform.

Expected File Extension	Mime-type	Sample Rate	File Format/Sample Size/Encoding
.vox	audio/x-vox audio/vox audio/basic	8000 Hz	• Raw audio • 8-bit mono • G.711 u-law, G.711 a-law (depends on platform configuration)
.au	audio/basic	8000 Hz	• Audio with `.au` header • 8-bit mono • G.711 u-law, G.711 a-law, PCM, ADPCM (depends on file header information)
.ulaw	audio/basic	8000 Hz	• Raw audio • 8-bit mono • G.711 u-law
.alaw	audio/x-alaw-basic	8000 Hz	• Raw audio • 8-bit mono • G.711 a-law
.g729	audio/g729	8000 Hz	• Raw audio • G.729
.pcm	audio/pcm audio/x-pcm	8000 Hz	• Raw audio • 8-bit unsigned mono • Linear PCM
.adpcm24	audio/x-g726-24	8000 Hz	• Raw audio • 24kbit/sec • ADPCM (G.726)

Expected File Extension	Mime-type	Sample Rate	File Format/Sample Size/Encoding
.adpcm	audio/x-g726 audio/x-adpcm audio/adpcm audio/x-adpcm8	8000 Hz	• Raw audio • 32kbit/sec • ADPCM (G.726)
.adpcm40	audio/x-g726-40	8000 Hz	• Raw audio • 40kbit/sec • ADPCM (G.726)
.pcm8	audio/L8 audio/pcm8 audio/x-pcm8	8000 Hz	• Raw audio • 8-bit unsigned mono • Linear PCM
.pcm16	audio/L16	8000 Hz	• Raw audio • 16-bit signed mono • Linear PCM
.wav	audio/wav audio/x-wav	8000 Hz	• Audio with `.wav` header • G.711 u-law, G.711 a-law, PCM, ADPCM (depends on file header information)
.nist	audio/wav audio/x-wav	8000 Hz	• Audio with `NIST` header • 8-bit mono • G.711 u-law, G.711 a-law (depends on file header information)
.gsm	audio/x-gsm	8000 Hz	• Raw audio • gsm 6.10
.amr	audio/amr	8000 Hz	• Raw audio • AMR
.3gp	audio/3gpp	8000 Hz	• Audio stored in 3GP container. • AMR
.263	video/h263 video/x-h263	30 fps (recommended)	• Raw video • h263
.263	video/h263-1998	30 fps (recommended)	• Raw video • h263-1998

Expected File Extension	Mime-type	Sample Rate	File Format/Sample Size/Encoding
.avi	video/avi video/x-avi	8000 Hz (audio) 30 fps (recommended for video)	• Audio/video stored in AVI container • Audio: G.711 u-law, G.711 a-law, PCM, ADPCM (depends on file header information) • Video: h263, h263-1998 (depends on file header information)
.3gp	video/3gpp	8000 Hz (audio) 30fps (recommended for video)	• Audio/video stored in 3GP container. • Audio: AMR • Video: h263, h263-1998 (depends on file header information)

Notes: The VoiceGenie platform examines the audio data to determine whether an audio/basic file is actually of `.au` format or is `.vox` format.

Only the 8000 Hz audio sampling rate is supported. If non-8000 Hz audio file is detected, the prompt will not play and a warning message will be issued. In pre-7.2 versions of Media Platform, the prompt would continue to play at the wrong sampling rate.

3.2 Supported Recording Formats

Type	Recorded File Format/Sample Size/Encoding	File Extension
audio/x-vox audio/vox audio/basic	Raw audio 8-bit mono G.711 u-law, G.711 a-law (depends on platform configuration)	.vox
audio/x-alaw-basic	Raw audio 8-bit mono G.711 a-law	.alaw
audio/g729	Raw audio G.729	.g729
audio/x-g726-24	Raw audio 24kbit/sec ADPCM (G.726)	.adpcm24

Type	**Recorded File Format/Sample Size/Encoding**	**File Extension**
audio/x-g726 audio/x-adpcm audio/adpcm audio/x-adpcm8	Raw audio 32kbit/sec ADPCM (G.726)	.adpcm
audio/x-g726-40	Raw audio 40kbit/sec ADPCM (G.726)	.adpcm40
audio/L8	Raw audio 8-bit unsigned mono Linear PCM	.pcm8
audio/L16	Raw audio 16-bit signed mono Linear PCM	.pcm16
audio/x-wav[;codec=<audio_codec>][;rate=<g726_encoding_rate>] audio/wav[;codec=<audio_codec>][;rate=<g726_encoding_rate>]	Audio with `.wav` header audio_codec: ulaw, alaw, pcm, pcm16, g726, gsm. Default is ulaw or alaw depends on platform configuration. g726_encoding_rate: 16kbps, 24kbps, 32kbps, or 40kbps. Default is 32kbps.	.wav
audio/x-gsm	Raw audio gsm 6.10	.gsm
audio/amr	Raw audio AMR	.amr
audio/3gpp	Audio stored in 3GP container AMR	.3gp
video/h263	Raw video h263	.263
video/h263-1998	Raw video h263-1998	.263

Type	Recorded File Format/Sample Size/Encoding	File Extension
video/avi[;codec=<audio_codec>] [;rate=<g726_encoding_rate>] [;videocodec=<video_codec>] video/x-avi[;codec=<audio_codec>] [;rate=<g726_encoding_rate>] [;videocodec=<video_codec>]	Audio/video stored in AVI container audio-codec: u-law, a-law, PCM, ADPCM, none. Default is u-law or a-law depends on platform configuration g726_encoding_rate: 16kbps, 24kbps, 32kbps, or 40kbps. Default is 32kbps. video_codec: h263, h263-1998. Default is h263.	.avi
video/3gpp[;codec=<audio_codec>] [;videocodec=<video_codec>]	Audio/video stored in 3GP container audio-codec: amr, none. Default is amr. video_codec: h263, h263-1998. Default is h263.	.3gp

Notes: AU and NIST file recording are not currently supported.

Only the 8000 Hz audio sampling rate is supported.

Chapter

4 Call Manager Configuration

This section describes the parameters in the Call Manager configuration file which can be accessed through the web-based SMC (System Management Console). The parameters described here may not be in the same order as they appear in the SMC configuration interface.

4.1 Main Program and CMAPI Section

4.1.1 Dynamic Loading of Line Managers

Media Platform can dynamically load line manager and media transport modules at startup. The following configuration parameters enable SIP, H323 line managers, RTP and MPC media transports. The configuration parameters can be modified to load a different combination of line managers and media transports. These parameters should be already set properly by the installation process.

Parameter	Description
callmgr.modules	This specifies the list of .so files to be loaded, in the order defined. Possible values: H323, SIP2, MPC, RTP, SnowShore Default: SIP2 MPC
callmgr.devices	This specifies the list of devices to be initialized on startup. Possible values: SWShoreDevice, none Default: none

Parameter	Description
callmgr.mediatransports	This specifies the list of mediatransports to be initialized on startup. Possible values: MTMPC, MTRTP, MTSnowShore Default: MTMPC
callmgr.linemanagers	This specifies the list of linemanagers to be initialized on startup. Possible values: LMH323, LMSIP2, LMSnowShore Default: LMSIP2
callmgr.clusterid	ID of the cluster that this media platform belongs to Default: Cluster1

4.1.2 Dynamic loading of CMAPI Application Modules

The Media Platform can dynamically load CMAPI application modules at startup. The following configuration parameters enable both the VoiceXML Interpreter (vxmli) module, remdial module along with other modules. The configuration parameters can be modified to load a different combination of application modules. These parameters should be set properly by the installation rpm. Customizations are only recommended for advanced users who require usage of special CMAPI applications for their deployments.

Parameter	Description
sessmgr.modules	This specifies the list of .so files to be loaded, in the order defined. Possible values: `VXML`, `Remdial`, `CCM`, `ContCheck`, `PolicyClient`, `Conference`, `PortCount` Default: `VXML Remdial`
sessmgr.appmodules	This specifies the list of names of app modules to be initialized on startup. The value is made up of :. specifies the module containing Possible values: `VXML:VXML`, `Remdial:RemoteDial`, `CCM:CCM`, `Conference:Conference`, `PolicyClient:PolicyClient`, `ContCheck:ContCheck`, `PortCount:PortCount` Default: `VXML:VXML Remdial:RemoteDial`
sessmgr.VXML.VXML	Name of each app module instances Default: `vxmli1`

Parameter	Description
sessmgr.VXML.VXML.#	This parameter defines the app module instance values Default: `8506 127.0.0.1`
sessmgr.VXML.VXML-NG	Name of each app module instance Default: vxmli-ng1
sessmgr.Remdial.RemoteDial	Name of Remdial Instance Default: `RemoteDial`
sessmgr.CCM.CCM	Name of CCM Instance
sessmgr.CCM.CCM.#	This parameter defines the app module instance values
sessmgr.Conference.Conference	Name of Conference Instance
sessmgr.Conference.Conference.#	This parameter defines the app module instance values
sessmgr.PolicyClient.PolicyClient	Name of PolicyClient Instance
sessmgr.PolicyClient.PolicyClient.#	This parameter defines the app module instance values
sessmgr.ContCheck.ContCheck	Name of ContCheck Instance
sessmgr.ContCheck.ContCheck.#	This parameter defines the app module instance values
sessmgr.PortCount.PortCount	Name of PortCount Instance
sessmgr.PortCount.PortCount.#	This parameter defines the app module instance values
sessmgr.default_vxml_interpreter	Specifies which VoiceXML Interpreter is used to handle calls that do not specify the VoiceXML Interpreter. Possible values: `VXML-NG, VXML` Default: `VXML-NG`
callmgr.billing.version	This controls the billing version. Supported values are `1.0` and `1.1.` Possible values: `1.0, 1.1` Default: `1.0`
sessmgr.ECS_Fallback	This determines whether to fallback on MediaRedirect transfer if CallJoin fails. • `1:` Fallback on MediaRedirect transfer if CallJoin fails. • `0:` Do not fallback on MediaRedirect transfer if CallJoin fails. Possible values: `1, 0` Default: `0`

Parameter	Description
sessmgr.join_fallback	This determines whether to fallback on MediaRedirect/Bridged transfer if CallJoin fails. • `1:` Falls back to MediaRedirect if supported. Otherwise, to Bridged transfer. • `0:` No fall back. Possible values: `1, 0` Default: `0`
sessmgr.licensecheck_interval	This determines how often license is checked during the run-time. Default value is 24 hours (1440 minutes). Default: `1440`

4.2 OA&M Framework Integration

4.2.1 Connection Settings

The following parameters configure settings with the OA&M Framework component:

Parameter	Description
cmp.proxy	This specifies call manager proxy. Default: `localhost`
cmp.proxy_port	The server port of this CMP Proxy, other VoiceGenie software components connect to this port Default: `8700`
cmp.heartbeat	The interval, in seconds, to send a periodic heartbeat message from the component to the CMP Proxy Default: `20`
cmp.reconnect	The interval, in seconds, between reconnection attempts to the CMP Server Default: `5`
cmp.sync	Specifies whether the configuration should be synchronized with the cmp database Possible values: `FALSE, TRUE` Default: `TRUE`

4.2.2 OA&M Framework Log Rotation Parameters

The following parameters control various rotation aspects for the logging options:

Parameter	Description
cmp.log_file	The full path to the log file of the Call Manager Default (Linux/Solaris): `/usr/local/phoneweb/logs/CMP.log.callmgr` Default (Windows): `C:\VoiceGenie\mp\logs\CMP.log.callmgr`
cmp.size_option	Rollover all log files by size or by time Possible values: `FALSE`, `TRUE` Default: `FALSE`
cmp.num_rollover_files	The number of files to roll through before they are overwritten when rolling over by size Default: `5`
cmp.rollover_size	The size limit, in MB, for rollover when rolling over by size Default: `10`
cmp.rollover_time	The time at which the log files are rolled over when rolling over by time Default: `4:00`
cmp.rollover_mins	The interval of time, in minutes, between rollover when rolling over by time Default: `1440`

4.2.3 OA&M Framework Email Parameters

Parameter	Description
cmp.email	If the `EMAIL` sink is specified, the email address be used Default: `name@domain.com`

4.2.4 OA&M Framework Logging Service Parameters

Parameter	Description
cmp.log_sinks	Logging sinks that will be used by this component,possible sinks are: `FILE, UPSTREAM, METRICS, SNMP, SYSLOG, EMAIL,` etc. Default: `FILE|UPSTREAM`
cmp.log_dll.#	This parameter defines the location of the sink DLL Default: N/A
cmp.UTC.#	UTC or Local Time Logging Possible values: `TRUE, FALSE` Default: `FALSE`
cmp.trace_flag	Determines if any logs at level `log_5` (tracing/debugging) should be logged Possible values: `FALSE, TRUE` Default: `FALSE`
cmp.pid_option	Appends PID of the process to the name of the trace file so that they are not overwritten when the process restarts Possible values: `FALSE, TRUE` Default: `FALSE`
cmp.log_queue_limit	The number of logs that can be queued for processing before the calling thread is throttled so that the logging thread does not fall behind indefinitely Default: `5000`
cmp.log_write_buffer_size	The size of the buffer, in bytes, for log event preallocation Default: `2560`
cmp.log_write_buffer	The size of the buffer, in bytes, to be used for block writing to the disk, a value of `0` implies no buffering Default: `65536`
cmp.log_write_buffer_stale_timeout	The longest time that a log can remain in the buffer before being written to disk Default: `2000`
cmp.log_write_buffer_idle_timeout	The amount of time during which no logs are received after which the buffer is written to disk Default: `1000`

Parameter	Description
cmp.log._threadloopproc	The maximum sleep time for the logging thread (in ms). A smaller value would mean the logging thread wakes up more frequently to process log requests. Default:
cmp.metrics	log mask for metrics data Default: 0|1
cmp.log_0	Log mask for data logged at log level 0 Default: 111111111111111111111111111111111111111111111111 111111111111111111111111111111111111111111111111 111111111111111111111111111111111111111111111111 111111111111111111111111111111111111111111111111 1|1111111111111111111111111111111111111111111111 111111111111111111111111111111111111111111111111 111111111111111111111111111111111111111111111111 111111111111111111111111111111111111111111111111 111
cmp.log_1	Log mask for data logged at log level 1 Default: 111111111111111111111111111111111111111111111111 111111111111111111111111111111111111111111111111 111111111111111111111111111111111111111111111111 111111111111111111111111111111111111111111111111 1|1111111111111111111111111111111111111111111111 111111111111111111111111111111111111111111111111 111111111111111111111111111111111111111111111111 111111111111111111111111111111111111111111111111 111
cmp.log_2	Log mask for data logged at log level 2 Default: 111111111111111111111111111111111111111111111111 111111111111111111111111111111111111111111111111 111111111111111111111111111111111111111111111111 111111111111111111111111111111111111111111111111 1|1111111111111111111111111111111111111111111111 111111111111111111111111111111111111111111111111 111111111111111111111111111111111111111111111111 111111111111111111111111111111111111111111111111 111

Parameter	Description
cmp.log_3	Log mask for data logged at log level 3 Default: 111111111111111111111111111111111111111111111111 111111111111111111111111111111111111111111111111 111111111111111111111111111111111111111111111111 111111111111111111111111111111111111111111111111 1|1111111111111111111111111111111111111111111111 111111111111111111111111111111111111111111111111 111111111111111111111111111111111111111111111111 111111111111111111111111111111111111111111111111 111
cmp.log_4	Log mask for data logged at log level 4 Default: 000000000000000000000000000000000000000000000000 000000000000000000000000000000000000000000000000 000000000000000000000000000000000000000000000000 000000000000000000000000000000000000000000000000 0|1111111111111111111111111111111111111111111111 111111111111111111111111111111111111111111111111 111111111111111111111111111111111111111111111111 111111111111111111111111111111111111111111111111 111
cmp.log_5	Log mask for data logged at log level 5 Default: 111111111111111111111111111111111111111111111111 111111111111111111111111111111111111111111111111 111111111111111111111111111111111111111111111111 111111111111111111111111111111111111111111111111 1|0000000000000000000000000000000000000000000000 000000000000000000000000000000000000000000000000 000000000000000000000000000000000000000000000000 000000000000000000000000000000000000000000000000 000

4.2.5 Guaranteed Logs Parameters

Parameter	Description
cmp.guaranteed_logs_to_file	Specify if logs that are guaranteed to be sent upstream should be logged to a temp file Possible values: `FALSE, TRUE` Default: `TRUE`
cmp.unsent_log_file	Specify the name of the temp log file to log to if `cmp.guaranteed_logs_to_file` Default (Linux/Solaris): `/usr/local/phoneweb/logs/guaranteed.log.callmgr` Default (Windows): `C:\VoiceGenie\mp\logs\guaranteed.log.callmgr`

4.2.6 Internal Media Transport

The Internal Media Transport Module is responsible for managing the internal media transmission to and from the ASR/TTS. This data transmission uses RTP. (Note that this is independent of the external RTP connections.) The following parameters control the internal media transport module:

Parameter	Description
mtinternal.rtp_min_port	The minimum port range for RTP sockets in MTInternal Default: `20000`
mtinternal.rtp_max_port	The maximum port range for RTP sockets in MTInternal Default: `30000`
mtinternal.max_sessions	Defines the maximum MTInternal sessions Default: `400`
mtinternal.transmit_interval	Defines a constant transmission interval in milliseconds. If set to `0,` packets will be sent as soon as data arrives. Default: `20`
mtinternal.transmit_rate	When `mtinternal.transmit_interval` is non-zero, this parameter specifies the maximum number of packets to be sent for each transmission interval. Set to `0` to turn off this restriction. Default: `5`

Parameter	Description
mtinternal.transmit_min_size	Defines the minimum data size in bytes that can be sent. Note that this number is applied to all codecs with fixed frame size. It will be rounded down to the nearest multiple of the codec frame size. This parameter will be disabled when variable frame size codec is used. Set to `-1` to disable the limit. Default: `160`
mtinternal.transmit_max_size	Defines the maximum data size in bytes that can be sent. Note that this number is applied to all codecs with fixed frame size. It will be rounded down to the nearest multiple of the codec frame size. This parameter will be disabled when variable frame size codec is used. Set to `-1` to disable the limit. Default: `160`
mtinternal.receive_min_size	Defines the minimum packet sample size that will be notified to the receiver. Note that this number is applied to all codecs with fixed frame size. It will be rounded down to the nearest multiple of the codec frame size. This parameter will be disabled when variable frame size codec is used. Set to `-1` to disable the limit. Default: `-1`
mtinternal.receive_max_size	Defines the maximum packet sample size that will be notified to the receiver. Note that this number is applied to all codecs with fixed frame size. It will be rounded down to the nearest multiple of the codec frame size. This parameter will be disabled when variable frame size codec is used. Set to `-1` to disable the limit. Default: `-1`
mtinternal.jitter_log	Defines the logging period in terms of number of received packets. If less than `1,` Jitter logging is turned off. Jitter logging will be disabled if variable frame size codec is used for received packets. Default: `0`
mtinternal.transmit_rate_alarm	If greater than `0,` minor alarm is generated if the transmission rate of outgoing packets is slower the real time by the specified delay in milliseconds. This alarm will be disabled if variable frame size codec is used for transmitted packets. Default: `500`

Parameter	Description
mtinternal.receive_rate_alarm	If greater than `0,` minor alarm is generated if the transmission rate of incoming packets is slower the real time by the specified delay in milliseconds. This alarm will be disabled if variable frame size codec is used for received packets. Default: `500`
mtinternal.transmit_savedata	If specified, utterance is saved under the directory.
mtinternal.receive_savedata	If specified, received data is saved under the directory.
mtinternal.max_concurrent_savedata	If specified as an integer *n*, and `mtinternal.transmit_savedata` or `mtinternal.receive_savedata` is enabled, then only a maximum of n concurrent files will be open for writing data. Default value is `-1,` which would place no limit. Default: `-1`

4.2.7 Remote Dial Application Module (Remdial) Settings

Parameter	Description
remdial.port	Remdial port Default: `6999`
remdial.maxcalls	Maximum number of concurrent remdial calls Default: `500`
remdial.telnetmode	Remdial telnet mode. If set to `RAW,` remdial will buffer data until it recieves a carriage return. `RAW` is for W2K. `NORMAL` is for linux Possible values: `RAW, NORMAL` Default: `NORMAL`
remdial.maxclientsockets	Max number of remdial clients allowed Default: `64`

4.2.8 VXML Application Module

Parameter	Description
calllog.directory	This parameter is used to specify the default full call recording file path if it is not specified on the page. Default (Linux/Solaris): `/usr/local/phoneweb/callrec` Default (Windows): `C:\VoiceGenie\mp\callrec`
vxml.audio_control_bargein_enable	This parameter will make possible Audio Control operations when the Bargein is enabled. When this parameter is `true` the defined Audio Control operations will take precedence over the DTMF grammars. Possible values: `true, false` Default: `false`
vxmli.use_isdn_mapping	This parameter controls whether the disconnected return status of the outbound leg should be derived from ISDN code or internal disconnect reason. • `0:` uses internal disconnect reason. • `1:` uses ISDN code Possible values: `0, 1` Default: `0`

4.2.9 Next Generation Interpreter

Parameter	Description
vxmli.builtin_path	This parameter indicates the main path to search for builtin audio files Default (Linux/Solaris): `/usr/local/phoneweb/audio` Default (Windows): `C:\VoiceGenie\mp\audio`
vxmli.recordutterance.path	This parameter indicates the parent directory where all the recorded utterance files are saved, when the user has specified the sub-directory name using com.voicegenie.utterancedest or vg:utterancedest. Default (Linux/Solaris): `/usr/local/phoneweb/utterance` Default (Windows): `C:\VoiceGenie\mp\utterance`

Parameter	Description
vxmli.default.alternate_uri	The value to use for an alternate URI when the main one can not be fetched.
vxmli.universals.help	This parameter specifies the universal help grammar used by the platform Default: `builtin:grammar/universals/Help`
vxmli.universals.exit	This parameter specifies the universal exit grammar used by the platform Default: `builtin:grammar/universals/Exit`
vxmli.universals.cancel	This parameter specifies the universal cancel grammar used by the platform Default: `builtin:grammar/universals/Cancel`
vxmli.maintainer.log_message.on_error	Controls whether the Interpreter will create a log message for the maintainer package automatically, when an error is thrown. Possible values: `FALSE, TRUE` Default: `true`
vxmli.trace	This parameter enables tracing within the VocieXML Interpreter Possible values: `FALSE, TRUE` Default: `true`
vxmli.oem_namespace	This defines the XML-namespace the applications must use for the non-standard, extension features. Each extension XML attribute/element must be defined in this namespace. Default: `http://www.voicegenie.com/2006/vxml21-extension`
vxmli.oem_property_prefix	The value to use for the prefix of custom/VG specific properties. Default: com.`voicegenie.`
vxmli.default.xmllang	The default value to use for xml:lang when it is not provided in the document. Default: `en-US`

Parameter	Description
vxmli.local.webserver.mimetypes	The interpreter exposes inline grammars as external grammars for an offboard speech engines as a URL reference, by a locally configured web server. This parameters defines the mappings between the media type of the grammars to the file extension of the exposed URL. The web server should be configured with the same mapping so that the media type of the grammar is exposed correctly to the speech engines. Default: `application/srgs+xml|.grxml|application/srgs|.srgs|Media-Type|.grammar|application/x-abnf|.abnf`
vxmli.local.webserver.baseurl	This is the base URL to be used when exposing inline grammars as a URL to be fetched by an offboard speech engine. Default: `http://$eth0-ip$/vggrammarbase/inlinetmp/`
vxmli.conformance.strict_grammar_mode	Indicates whether the interpreter will follow the VoiceXML specification strictly when handling the grammar element. Specifically, when set to false it will NOT ignore the mode attribute for an external grammar. Possible values: `FALSE, TRUE` Default: `false`
vxmli.conformance.supported_builtin_dtmf	Indicates the platform supported dtmf built-in grammars when strict grammar mode is enabled Default: `boolean digits currency date number phone time`
vxmli.conformance.supported_builtin_voice	Indicates the platform supported voice built-in grammars when strict grammar mode is enabled Default: `boolean digits currency date number phone time universals/Cancel universals/Exit universals/Help`
vxmli.supported_grammar_languages	Indicates the grammar languages supported Default: `en-US`
vxmli.conformance.strict_tts_mode	Indicates whether the Interpreter will be strict in conformance of the tts mode. The TTS language will be checked against the list specified in vxmli.conformance.supported_tts_languages. Possible values: `FALSE, TRUE` Default: `false`

Parameter	Description
vxmli.conformance.supported_tts_languages	Indicates the tts languages supported Default: `en-US`
vxmli.conformance.supported_grammar_languages	Indicates the grammar languages supported. Note that this is only meaningful when vxmli.conformance.strict_grammar_mode is enabled. Default: `en-US`
vxmli.conformance.strict_complete_timeout	When set to true, the interpreter will calculate the maximum of the completetimeout and incompletetimeout values as the value for the incompletetimeout. Possible values: `FALSE, TRUE` Default: `true`
vxmli.conformance.disable_application_lastresult_extensions	When set to true, none of the additional extension properties of the application.lastresult$ object are set when a result is exposed. Possible values: `FALSE, TRUE` Default: `false`
vxmli.conformance.disallow_exec_content_within_prompts	When set to true, executable content is not permitted inside foreach, when the foreach is inside a prompt. Possible values: `FALSE, TRUE` Default: `false`
vxmli.defaults_vxml_url	This parameter specifies the defaults.vxml path if a default root page is not specified in the DNIS-URL mapping. Default (Linux/Solaris): `file:///usr/local/phoneweb/config/defaults-ng.vxml` Default (Windows): `file://C:\VoiceGenie\mp\config/defaults-ng.vxml`
vxmli.max_num_documents	This parameter specifies the maxium number of cacheable documents Default: `2000`
vxmli.max_num_sessions	The maximum number of permitted concurrent sessions Default: `10000`
vxmli.grammars.cache_size	The amount of memory to allocate for caching grammars. This is slightly more than 100 bytes per grammar. Default: `50000`

Parameter	Description
vxmli.break.strength.x-weak	Specifies the time in milliseconds that the Interpreter should use when encountering a break with the specified strength. This value will be ignored if the break is rendered by a TTS service. Default: `50`
vxmli.break.strength.extraweak	Specifies the time in milliseconds that the Interpreter should use when encountering a break with the specified strength. This value will be ignored if the break is rendered by a TTS service. Default: `100`
vxmli.break.strength.weak	Specifies the time in milliseconds that the Interpreter should use when encountering a break with the specified strength. This value will be ignored if the break is rendered by a TTS service. Default: `200`
vxmli.break.strength.medium	Specifies the time in milliseconds that the Interpreter should use when encountering a break with the specified strength. This value will be ignored if the break is rendered by a TTS service. Default: `500`
vxmli.break.strength.strong	Specifies the time in milliseconds that the Interpreter should use when encountering a break with the specified strength. This value will be ignored if the break is rendered by a TTS service. Default: `1000`
vxmli.break.strength.x-strong	Specifies the time in milliseconds that the Interpreter should use when encountering a break with the specified strength. This value will be ignored if the break is rendered by a TTS service. Default: `2000`
vxmli.break.strength.extrastrong	Specifies the time in milliseconds that the Interpreter should use when encountering a break with the specified strength. This value will be ignored if the break is rendered by a TTS service. Default: `5000`
vxmli.ac.enabled	Controls support for access-control when using the tag Possible values: `FALSE, TRUE` Default: `true`

Parameter	Description
vxmli.ac.allow_if_missing	Used for . Determines the behaviour when fetched XML data doesn't contain any access-control processing instructions. This parameter only has an effect if vxmli.ac.enabled is set to true. Possible values: `FALSE, TRUE` Default: `false`
vxmli.ac.allow_if_nomatch	Used for . Determines behaviour of access-control when the host machine does not appear in any access-control directive. This parameter only has an effect if vxmli.ac.enabled is set to true. Possible values: `FALSE, TRUE` Default: `false`
vxmli.ac.use_platform_host_for_file_url	Used for the element. It determines the behaviour when the VoiceXML page accessing the XML data is a file URI. When set to true it will force access-control to use the hostname of the platform when verifying access-control instructions. When set to false, access will be allowed if VoiceXML page is a file URI. Possible values: `FALSE, TRUE` Default: `true`
vxmli.transfer.allowed	Indicates whether transfers should be permitted Possible values: `FALSE, TRUE` Default: `true`
vxmli.default.connecttimeout	The default value to use for a transfer's connecttimeout attribute if not provided. Applies to bridge or consultation transfers. Specified in milliseconds. Default: 30000

Parameter	Description
vxmli.session_vars	Each session variable entry is composed of three components. The first component is the session variable name as exposed within VoiceXML. The second component is the variable name sent back from the Call Manager. The third component indicates either whether the session variable will be included in the request for the initial page URL (0 = do not include, 1 = include in GET, 2 = include in POST, 3 = include in GET and POST), or the type of array of the session variable (6 = associative array, 7 = ???). Default: `session.connection.local.uri|LOCALURI|1|session.connection.remote.uri|REMOTEURI|1|session.connection.originator|ORIGIN|1|session.connection.protocol.name|PROTOCOLNAME|0|session.connection.protocol.version|PROTOCOLVERSION|0|session.connection.protocol.sip.headers|Sip.Invite|6|session.connection.redirect|REDIRECTHEADER|7|session.connection.callidref|CALLIDREF|1|session.com.voicegenie.instance.parent|PARENT|1|session.connection.ocn|OCN|1|session.connection.rdnis|RDNIS|1|session.connection.rreason|RREASON|1`
vxmli.initial_request_maxstale	Specifies the maximum amount of time (in ms) past content expiration that the VXML document is willing to accept. -1 if undefined. Default: `-1`
vxmli.initial_request_fetchtimeout	The fetch timeout (in ms) of the initial VXML document. If document fetch is not completed within this time, the fetch is considered to have failed and the call will be rejected. If value is set to 0, the parameter will be be ignored and 60000 will be used instead. Default: `30000`
vxmli.initial_request_method	The HTTP method to use for the initial request Possible values: `POST, GET` Default: `GET`
vxmli.initial_request_enctype	The HTTP encoding type to use for the initial request when the request method is POST Default: `application/x-www-form-urlencoded`

Parameter	Description
vxmli.expose.nlsml.dom	Instructs the interpreter whether to expose the NLSML result from the recognizer as a DOM object in application.lastresult$.xmlresult. Possible values: `FALSE, TRUE` Default: `true`
vxmli.tmpdir	Temp directory that exists on the platform Default (Linux/Solaris): `/usr/local/phoneweb/tmp/` Default (Windows): `C:\VoiceGenie\mp\tmp/`
vxmli.logdir	The directory for logs created from the log element with destination file. Default (Linux/Solaris): `/usr/local/phoneweb/logs/` Default (Windows): `C:\VoiceGenie\mp\logs/`
vxmli.max_application_logfile_size	The maximum size in bytes of an application log file which can be logged by using the log element with dest value set to file. Default: `524288000`
email.smtpAddr	SMTP server address for sending maintainer e-mails Default: `localhost`
email.fromAddr	On Windows, this is the "From" header for maintainer e-mails. On Linux, it appears as the first line of the message body. Default: `nobody@example.com`
vxmli.property.	List of VXML properties for which default values can be configured.
vxmli.property.#	This parameter defines the default value of the specified property.
vxmli.property.__	List of VXML VG extension properties for which default values can be configured.
vxmli.property.__.#	This parameter defines the default value of the specified VG extension property.
vxmli.grammar.builtin:dtmf/time	Builtin time grammar path. Default: `http://$eth0-ip$/vggrammarbase/dtmf/time.grxml`

Parameter	Description
vxmli.grammar.builtin:dtmf/phone	Builtin phone grammar path. Default: `http://$eth0-ip$/vggrammarbase/dtmf/phone.grxml`
vxmli.grammar.builtin:dtmf/number	Builtin number grammar path. Default: `http://$eth0-ip$/vggrammarbase/dtmf/number.grxml`
vxmli.grammar.builtin:dtmf/date	Builtin date grammar path. Default: `http://$eth0-ip$/vggrammarbase/dtmf/date.grxml`
vxmli.grammar.builtin:dtmf/currency	Builtin currency grammar path. Default: `http://$eth0-ip$/vggrammarbase/dtmf/currency.grxml`
vxmli.metrics.level_set0	This list specifies the available metrics levels. Default: `log`
vxmli.metrics.level_set1	This list specifies the available metrics levels. Default: `meta_appl compile_done`
vxmli.metrics.level_set2	This list specifies the available metrics levels. Default: `appl_begin appl_end asr_trace error exec_error warning parse_error prompt_start prompt_play prompt_end`
vxmli.metrics.level_set3	This list specifies the available metrics levels. Default: `catch_enter catch_exit choice_select compile_time dtmf ecma_timing event fetch_end filled_enter filled_exit_legacy filling form_enter menu_enter form_exit menu_exit form_select goto input_end input_modes link_triggered parse_warning record_start record_end return root_appl script_result subdialog_start_param submit transfer_start transfer_end`
vxmli.metrics.level_set4	This list specifies the available metrics levels. Default: `fetch_start`

Parameter	Description
vxmli.metrics.level_set5	This list specifies the available metrics levels. Default: `block_enter block_exit eval_cond eval_expr filled_exit field_enter field_exit if_enter if_exit initial_enter initial_exit link_enter link_exit notify_transition object_enter object_exit record_enter record_exit record_result subdialog_enter subdialog_start subdialog_exit submit_start submit_end throw transfer_enter transfer_exit var_begin var_eval value_begin`
vxmli.metrics.level.#	This specifies the level of an individual metric. In the legacy interpreter, these were hard coded. Refer to VXML3 Metrics. Default: `0`
vxmli.metrics.level.#	This specifies the level of an individual metric. In the legacy interpreter, these were hard coded. Refer to VXML3 Metrics. Default: `1`
vxmli.metrics.level.#	This specifies the level of an individual metric. In the legacy interpreter, these were hard coded. Refer to VXML3 Metrics. Default: `2`
vxmli.metrics.level.#	This specifies the level of an individual metric. In the legacy interpreter, these were hard coded. Refer to VXML3 Metrics. Default: `3`
vxmli.metrics.level.#	This specifies the level of an individual metric. In the legacy interpreter, these were hard coded. Refer to VXML3 Metrics. Default: `4`
vxmli.metrics.level.#	This specifies the level of an individual metric. In the legacy interpreter, these were hard coded. Refer to VXML3 Metrics. Default: `5`
vxmli.maintainer.email_subject	The text to use as the subject for Maintainer Email messages. Default: `Message from VoiceGenie 7.2 to Application Maintainer`
vxmli.directories.save_tempfiles	The directory in which to save tempfiles. Default (Linux/Solaris): `/usr/local/phoneweb/tmp/` Default (Windows): `C:\VoiceGenie\mp\tmp/`

Parameter	Description
vxmli.script_max_loop	Maximum number of loops is allowed in each script or ECMAScript expression execution. The loop counter will be increased by 1 when a script branches backward during execution and when a function returns. Default: `1000000`
vxmli.max_script_time	Maximum duration in millisecond is allowed for each script or ECMAScript expression execution. Default: `2000`
vxmli.max_loop_count	Maximum number of runtime loops is allowed between waiting states in an application execution. The runtime loop count will be increased when any form item, event handler and an iteration of is executed. And the counter will be reset at a waiting state (e.g. waiting for user input, recording and transferring call). Default: `1000`
vxmli.beep.uri	The URI (can be either file:// or http://) of the beep file to be played when beep="true" in the tag. Default (Linux/Solaris): `file:///usr/local/phoneweb/audio/effects/endofprompt.vox` Default (Windows): `file://C:\VoiceGenie\mp\audio\effects\endofprompt.vox`
vxmli.recording.recovery.directory	This parameter indicates the directory path in which to store .recovery files for recordings.
vxmli.legacy.simple_dtmf_grammars	When set to true this parameter tells the interpreter to support certain inline legacy DTMF grammars that do not follow proper ABNF syntax. Possible values: FALSE, TRUE Default: `false`
vxmli.num_session_processing_threads	The total number of VXML page execution threads to create. Default: `8`

4.2.10 Call Control Application Module (CCM) Section

These parameters configure the Call Control Application Module, which is used when integrating with the Call Control Platform for CTI integration:

Parameter	Description
ccm.tcp_port	TCP port on which `libAPPCCM.so` listens for connections from TCP. This value should match the port number known to CCP, which is defined in the `ICM Platform Mapping` section of the VMC Default: `1111`
ccm.on_error_url	VXML script to play if an error (for example a software error, or an unexpected network disconnect from CCP<->CCM or ICM<->CCP) occurs Default (Linux/Solaris): `file:///usr/local/phoneweb/samples/saysorryanddisconnect.vxml` Default (Windows): `file://C:\VoiceGenie\mp\samples\saysorryanddisconnect.vxml`
ccm.on_error_defaults	VXML default property page if an error (for example a software error, or an unexpected network disconnect from CCP<->CCM or ICM<->CCP) occurs Default: `defaults.vxml`

4.2.11 General Parameters

The call manager can define the minimum number of ASR and TTS resources that must be available in order for the Call Manager to accept incoming calls. Existing calls in the platform will be allowed to continue if at any time the number of enabled ASR/TTS resources dip below the required number. The default value is `0;` the Call Manager will accept calls with no enabled ASR and TTS resources available.

Parameter	Description
MIN_ASR_REQUIRED	These are the minimum number of ASR resources required for the callmgr to accept any calls. If such number of ASR resources is not available, then the callmgr will not accept any more calls while allowing the existing calls to continue. This parameter should only be used with MRCP-native ASR, otherwise it should be set to 0. Default: `0`

Parameter	Description
MIN_TTS_REQUIRED	These are the minimum number of TTS resources required for the callmgr to accept any calls. If such number of TTS resources is not available, then the callmgr will not accept any more calls while allowing the existing calls to continue. This parameter should only be used with MRCP-native TTS, otherwise it should be set to 0. Default: `0`

This parameter specifies the location of the file containing the beep tone used by the `<record beep="true">` tag to signal to beginning of record:

Parameter	Description
record.start.beep.filename	This parameter is used to specify the filename for the 'beep' before doing a record. Default (Linux/Solaris): `/usr/local/phoneweb/audio/effects/endofprompt.vox` Default (Windows): `C:\VoiceGenie\mp\audio\effects\endofprompt.vox`

The following parameter controls the network cause code field to be transmitted for the badfetch and decline cases:

Parameter	Description
sessmgr.disconnect_cause.badfetch	This parameter is used to specify the ISDN disconnect cause code if the initial page fetch failed for some reason Default: `17`
sessmgr.disconnect_cause.decline	This parameter is used to specify the ISDN disconnect cause code if the platform has chosen to decline the call Default: `21`

The following parameters enable the use of inband transfer and allow customization of the behaviour. Note that enabling the use of inband transfer will automatically disable the use of any other transfer mechanisms provided by the line managers:

Parameter	Description
sessmgr.inbandxferprefix	This specifies the inband transfer dialing prefix used for inband transfer. The transfer number will be appended to this string. For example, in the case of TBT, or the default provisioning of TC, this would typically be `*8`.

Parameter	Description
sessmgr.inbandxfertimeout	This specifies timeout value to terminate the inband transfer. Time, in milliseconds, after end of dialing until we force a disconnect. `-1` means do not disconnect; if the network side does not disconnect the call within the duration of connecttimeout the transfer is failed with no answer. Default: `0`

The following parameter configures the maximum length of inbound calls to the system:

Parameter	Description
sessmgr.maxincalltime	This specifies the maximum call time for inbound calls in seconds. When the timer expires, the inbound call will be disconnected. Default: `0`

The following parameter controls the initial state of the Call Manager after start up:

Parameter	Description
sessmgr.init_accept_call_mode	This specifies the AcceptCallMode when the platform starts up. • `INBOUND:` Accept only inbound call. • `OUTBOUND:` Accept only outbound call • `DUPLEX:` Accept both inbound and outbound calls • `DISABLE:` Do not accept calls Possible values: `INBOUND, OUTBOUND, DUPLEX, DISABLE` Default: `DUPLEX`

The following parameter allows the alerting (ringing) to happen before the VoiceXML application is completely fetched:

Parameter	Description
sessmgr.alert_before_fetch	This issues alerting message to phone network before the page is successfully fetched. • `1:` Yes • `0:` No Possible values: `1, 0` Default: `0`

The following parameter controls when media switching is performed during an outbound call request:

Parameter	Description
sessmgr.mediaswitch_on_alert	When connectimmediate is true, do media switching on outbound alerting message instead of call-out-requested. • `1:` Yes • `0:` No Possible values: `1, 0` Default: 0

4.3 Conferencing Configuration Parameters

Parameter	Description
conference.limit	Max number of participants allowed for the conference initiated by conferencing app. Default: 32
conference.initial_gain	Gain in db when talking to the conference. Default: 0
conference.dynamic_gain	Whether to set auto gain control on input. Default: 0
conference.confdir	default conference direction of the participant. • `0:` Talk only. • `1:` Listen only. • `2:` Duplex. Possible values: `0, 1, 2` Default: 2
conference.highest_input	Whether to choose N highest input for mixing output, `0` indicates choosing all. Default: 0
conference.suppress_silence	Whether to suppress silence on input. • `0:` No • `1:` Yes Possible values: `0, 1` Default: 0

Parameter	Description
conference.silence_fill	Whether to silence fill the output when no data. • `0:` No • `1:` Yes Possible values: `0, 1` Default: `0`
conference.audio_format	Audio Format, default is `pcmu` Possible values: `pcmu, pcma, g726-16, g726-24, g726-32, g726-40, lpcm16, gsm` Default: `pcmu`
conference.video_output_algorithm	Specifies how the conference chooses the video output. • `fixed` will select the first conference participant. • `loudest` will select the loudest participant. • `none` will disable video. Default is `loudest.` Possible values: `fixed, loudest, none` Default: `loudest`

4.4 Obsolete Transfer Configurations

The following transfer-related configurations have become obsolete within the call manager configuration:

- `sip.blindxfermode` (still supported if the new parameters are not defined)
- `h323.transfermode` (still supported and takes precedence over the new parameters)

These parameters have been replaced with the following new parameters:

- `sip.transfermethods`
- `h323.transfermethods`

The old configuration mechanism restricts that only one transfer method (or transfer mode) can be supported at a time. The new parameters take a list of transfer method names and allow multiple transfer methods to be supported simultaneously. The method names are the same as the ones that can be used on the transfer tag method attribute.

For example, if both SIP REFER transfer and Hookflash transfer can be supported on the same Media Platform, the following configuration can be used:

```
sip.transfermethods=REFER HKF
```

Note that the default configurations assume no transfer capabilities are available from the telephony network (hence, all transfers are using bridge method).

The following transfer configurations have been added to the call manager configuration:

- `h323.defaultblindxfer`
- `h323.defaultconsultxfer`
- `h323.defaultbridgexfer`
- `sip.defaultblindxfer`
- `sip.defaultconsultxfer`
- `sip.defaultbridgexfer`

These parameters define the default transfer method to be used if none is defined by the VoiceXML page. The method name must be supported by default, or defined in the supported transfer method configuration list/bitmap.

4.5 SIP Line Manager

4.5.1 General

Parameter	Description
sip.info.contenttype	Specifies content type of outgoing SIP `INFO` messages that correspond to VoiceXML application events. A VoiceXML application can trigger the sending of a SIP `INFO` message by using tag with `dest="callmgr".` Call manager will then send a SIP `INFO` message to the remote end with content being the content of the tag. The default content type is `application/text.` Default: `application/text`
sip.inboundacktimer	timeout value to terminate SIP inbound call if no ACK is received after sending 200OK final response. The units are in msec Default: `40000`
sip.sendalert	Specifies the SIP response for alerting. • `0:` No SIP response • `1:` Send `180 RINGING` response • `2:` Send `183 Session Progress` response with SDP info Possible values: `0, 1, 2` Default: `1`

Parameter	Description
sip.confserver	Address of the VoiceGenie conferencing server. If a VoiceGenie conferencing server is deployed in the network, the address of that server is configured through this parameter. Either the IP address or hostname of the conferencing server may be specified. If the conferencing server uses a non-default SIP port (other than 5060), it must be specified using `:port` following the IP address/hostname. If this parameter is not specified, no conference server will be configured. `sip.confserver=sipconf.voicegenie.com:5020`
sip.vxmlinvite	Specifies acceptance of VoiceXML URLs in `INVITE` message. It is possible for the originator of a SIP call to specify the initial VoiceXML URL that will be delivered on a session by encoding the Request-URI in the special form `sip:dialog.vxml.@host.com.` The portion of the request URI must be encoded (e.g. : `-> %3A`). If such URLs are received, the normal DNIS mapping procedure will be bypassed, and the specified URL will be fetched. `0` is disable and `1` is enable Possible values: `0, 1` Default: `1`
sip.deferoutalerting	Defer `CallOutAlerting` response to SessionMgr. This is for early media for an outbound call. If set to `1,` it will defer `CallOutAlerting` to Session Manager until the media session is initialized and registered. Hence, the session manager can start performing media operations on the channel after `CallOutAlerting` notification. `0` is off and `1` is on. Possible values: `0, 1` Default: `0`
sip.sipinfoallowedcontenttypes	Content types in a SIP `INFO` messages that are allowed to be passed up to the application level. Only the defined content types would be passed up, others would be ignored. If left empty, the default value is `allowall,` which means the content of all received SIP `INFO` messages would be passed upstream.

Parameter	Description
sip.dnis_correlationid_offset	If this parameter is enabled, correlation ID is extracted from the user-id portion of the DNIS, and the correlation ID portion is stripped from DNIS. Value is an integer that specifies the offset of the correlation ID within the user-id. If it is negative, it specifies the offset from the right. Note the special case where correlation ID is all of user-id; `@` will be stripped away from the DNIS as well since `@<hostname>` does not make sense. Default: `0`
sip.dnis_correlationid_length	If this parameter is enabled, correlation ID is extracted from the user-id portion of the DNIS, and the correlation ID portion is stripped from DNIS. Value is a non-negative integer that specifies the length of the correlation ID within the user-id. Note the special case where correlation ID is all of user-id; `@` will be stripped away from the DNIS as well since `@<hostname>` does not make sense. Default: `0`
sip.threadpoolsize	The size of the thread pool for handling DNS queries. Default: `4`
sip.mtusize	Defines the Maximum Transmission Unit (MTU) of the network interfaces. If a SIP request size is within 200 bytes of this value, the request will be sent on a congestion controlled transport protocol, such as TCP. Default: `1500`
sip.enabletfci	Allow TFCI outbound calls. Default: `0`
sip.maxtcpconnections	Defines the maximum number of TCP connections concurrently established. If the maximum number of TCP connections has been reached, new SIP requests to establish TCP connections will be rejected Default: `100`
sip.sessionexpires	Defines the default session expiry value in seconds. The session timer defines the duration of which a SIP session will expire if no re-`INVITE`s are sent/received within this period. Default: `1800`

Parameter	Description
sip.timer.ci_proceeding	Defines the client `INVITE` proceeding timer in milliseconds, default value is `120000.` The timer starts after a `1xx` response is received for a client `INVITE.` If a final response is not received before the timer expires, the SIP session and dialog will be destroyed without further notice to the UAS. Note that the CI proceeding timer should be configured to be greater than the connect timeout. This ensures that a `CANCEL` will be sent to terminate the SIP session properly when connect timeout occurs. Default: `120000`
sip.min_se	Defines the `Min-SE` parameter in seconds. This is the minimum duration of session expiry this SIP stack will accept from a user agent client. Default: `90`

4.5.2 Customizable Headers and Parameters

Parameter	Description
sip.in.invite.headers	Defines list of headers to expose to the application. This specifies a list of header names from the incoming INVITE requests, whose values will be exposed to the application. For example, sip.in.invite.headers = From To Via. The exposed values' names will be in sip.invite.<headername>=<value> format. If this value is '*', then all headers will be exposed. If this value is 'none', then no headers will be exposed. 'none' will be ignored alongside other values. Default: `*`
sip.in.invite.params	Defines list of parameters to expose to the application. This specifies a list of header names from the incoming INVITE requests, whose parameter values will be exposed to the application. For example, sip.in.invite.params = From To Via. The exposed values' names will be in sip.invite.<headername>.<paramname>=<value> format. If this value is 'none', then no parameters will be exposed. 'none' will be ignored alongside other values. Default: `RequestURI`

Parameter	Description
sip.in.bye.headers	Defines list of headers to expose to the application. This specifies a list of header names from the incoming BYE requests, whose values will be exposed to the application. For example, sip.in.bye.headers = Reason. The exposed values' names will be in sip.invite.<headername>=<value> format. If this value is '*', then all headers will be exposed. If this value is 'none', then no headers will be exposed. Default: `Reason`
sip.out.invite.headers	Defines list of headers to expose to the application. This specifies a list of header names from the outgoing INVITE requests, whose values can be customized by the application. For example, sip.out.invite.headers = From To Via. The customized values' names will be in sip.invite.<headername>=<value> format. If this value is '*', then all headers will be exposed. If this value is 'none', then no headers will be exposed. 'none' will be ignored alongside other values. Default: `none`
sip.out.invite.params	Defines list of parameters to expose to the application. This specifies a list of header names from the outgoing INVITE requests, whose parameter values can be customized by the application. sip.out.invite.params = RequestURI. The customized values' names will be in sip.invite.<headername>.<paramname>=<value> format. If this value is 'none', then no headers will be exposed. 'none' will be ignored alongside other values. Default: `RequestURI`
sip.out.refer.headers	Defines list of headers to expose to the application. This specifies a list of header names from the outgoing REFER requests, whose values can be customized by the application. For example, sip.out.refer.headers = From To Via. The customized values' names will be in sip.refer.<headername>=<value> format.

Parameter	Description
sip.out.refer.params	Defines list of parameters to expose to the application. This specifies a list of header names from the outgoing REFER requests, whose parameter values can be customized by the application. sip.out.refer.params = RequestURI. The customized values' names will be in sip.refer.<headername>.<paramname>=<value> format. Default: `RequestURI`
sip.copyunknownheaders	Copy unknown headers from request to all responses. If this parameter is turned on, all unknown SIP headers found in SIP request will be automatically copied to its responses. 0 is disable and 1 is enable. Possible values: `0, 1` Default: `1`
sip.copyheaders	Copy specified headers from inbound call INVITE to outbound call INVITE for bridged calls and RLT calls. This parameter reads a space delimited list of header names. MP will copy this list of header fields from an inbound call INVITE to outbound call INVITE of the same voicexml session (ie. bridged calls and RLT calls). Note that re-INVITE from the inbound call causes headers re-scan and applies latest changes on any outbound calls made within the call session. sip.copyheaders = VG-SS7-Xfer-Param

4.5.3 Call Routing

Parameter	Description
sip.outcalluseoriggw	If a SIP call is placed via call or transfer, and the destination address does not contain a hostname or IP address, this parameter will determine which gateway to use. If `sip.outcalluseroriggw` is set to `1,` the call will be placed using the gateway of the inbound call. If `sip.outcalluseroriggw` is set to `0,` either sip.defaultgw or sip.defaulthost will be used. Default is `0.` Possible values: `0, 1` Default: `0`

Parameter	Description
sip.defaulthost	Default host/port for SIP calls if none given. If a call is placed (either via transfer, call, or remdial) using SIP, and the destination address does not contain a hostname or IP address, this parameter will supply a default hostname or IP address.
sip.defaultgw	Default host/port for SIP calls if none given. If a call is placed (either via transfer, call, or remdial) using SIP, and the destination address is a telephone address, then the call will be routed to the configured default gateway.
sip.localuser	Configures the user name portion of the `Contact` header generated from the platform Default: `VoiceGenie`
sip.transport.0	Configures the sip stack's transport settings. Default: `transport0 udp:any:5060`
sip.localhostname	sip.localhostname provides configurability of the host address part of Contact, Call-ID and From headers. If this parameter is not specified, then the IP Address of the local system will be used. If this value is not defined, sip.localport will be ignored. This parameter can also be used to provide the fully qualified domain name in SIP requests. Example: sip.localhostname=sip.voicegenie.com
sip.localport	Similar to sip.localhostname, the parameter sip.localport provides configurability of the port part of Contact, Call-ID and From Headers. If this parameter is not specified, the default SIP port number of 5060 is used. Note that if sip.localhostname is not defined, sip.localport will be ignored. Default: `5060`
sip.transport	Defines the transport instance ID to configure. The list of transport instance ID must be defined in consecutive and increasing order starting from 0. If 0, 1, 3 are defined, 3 will be ignored. If this parameter is disabled, the Media Platform will enable both UDP and TCP transports and listen from port 5060 on any network interface. Default: 0

Parameter	Description
sip.transport.#	defines transport layer for SIP stack and the network interfaces that are used to process SIP requests Format: sip.transport.x = transport_name type:ip:port [parameters] where transport_name is any string; type is udp/tcp; ip is the IP address of the network interface that accepts incoming SIP messages; port is the port number where SIP stack accepts incoming SIP messages; [parameters] defines any extra SIP transport parameters. Note that this is for LMSIP2. Example: sip.localport setting is for LMSIP. Default: `transport0 udp:any:5060`
sip.registerexpiryadjustment	Specifies the amount of time (in seconds) that the platform should re-register with the configured registrars before their respective expiration times are reached Default: `10`
sip.routeset	Defines a SIP route set for outbound calls. If defined, this route set will be inserted as the ROUTE header for all outgoing calls. This will force the platform to send the SIP messages via this defined route set. Each element in the routeset should be seperated by a comma. This parameter can be used to define outbound proxies. Note that this routeset does not apply to SIP REGISTER messages. sip.routeset = <sip:ip/host;priority>, ... e.g. sip.routeset=<sip:p1.example.com;lr>,<sip:p2.domain.com;lr> In this example, the VoiceGenie platform will route the request to p1.example.com, which will in turn route the message to p2.domain.com, and finally be redirected to its intended destination.

Parameter	Description
sip.registration	Specifies setting for registration. The system can be configured to register with one or more SIP registration servers on the network. The format of the value for `sip.registration` entries is: `<registration-server> <register-as> <requested-expiry> <username> <passowrd> <routeset>` All parameters except `routeset` are compulsory. • `<registration-server>` – Host/port with which to register. As the domain of the location service (e.g. `voicegenie.com`), the `userinfo` and `@` components *must not* be present. • `<register-as>` – SIP identity to register as (without leading `sip:`) • `<requested-expiry>` – Duration of registration; system will re-register after registration expires • `<username>` – The user name when authentication is required by the server. This may or may not be the same as register-as.A dash (`-`) should be used if no user name is needed.Anonymous will be used if the server request authentication under this setting. • `<password>` – The password assoicated with the authentication user name. To specify an empty string please use the dash (`-`) character. • `<routeset>` – Route set to define the list of server(s) that the `REGISTER` messages should go through. Each entry seperated by a comma and no space in between. If left empty, the `REGISTER` messages will be sent directly to the registration-server. The system will attempt to register with all defined registration entries and will periodically re-register as required by the requested-expiry parameter. The system will unregister when shutting down. e.g. `sip.registration = proxy1.voicegenie.com:5064 vg@10.0.0.101 60 - -|proxy2.voicegenie.com:5064 vg@10.0.0.102 60 user password`
sip.route.dests	A list of space-delimited entries in a routing table. The entry ID starts from `0` and increments by `1` each time. For example, to specify 4 entries in the routing table, the value would be `0 1 2 3`

Parameter	Description
sip.route.dest.#	This is an entry in routing table. Format: sip.route.dest.x=[Destination] [Netmask] [Transport] [Metric] To select an entry in routing table, we mask the outgoing IP Address with [Netmask]; if the result matches with the [Destination], we will accept that route. The [Transport] part determines the transport to use and maps to the index 'x' in one of the transports defined as sip.transport.x. In most of the cases, first accepted route will be used. Unless the protocol is specified or required (for example, when the message size is larger than mtusize, tcp is required to be used), the accepted route in routing table is also required to have matched protocol. If there.s no such route, default transport of that protocol will be used. If all cases failed, sip.transport.0.s protocol will be obtained. The default transport of the obtained protocol will be used. Note that [Metric] entry is needed but not used at this point. For example: sip.route.dest.0=138.120.72.0 255.255.255.0 1 0 When we make a call to the machine 138.120.72.20, outgoing IP is masked with [netmask] using .bitwise AND. operator. In this case: 138.120.72.20 & 255.255.255.0 gives 138.120.72.0. This matches the defined [Destination] in the route. Therefore, transport in sip.transport.1 will be used.
sip.route.default.udp	Default route for UDP. The number denotes the transport defined in `sip.transport.x` where `x` is the value of this parameter and will be used when no UDP routes are found. If this parameter is not set, the first UDP transport found in `sip.transport.x` becomes the default.
sip.route.default.tcp	Default route for TCP. The number denotes the transport defined in `sip.transport.x` where `x` is the value of this parameter and will be used when no TCP routes are found.

4.5.4 Media

Parameter	Description
sip.localrtpaddr	Specifies the IP address to advertise for RTP. With multicast/proxied systems, it may be necessary to specify what IP address to advertise in the SDP that describes a session. By default, the IP address of the local system is retrieved by performing a standard gethostname(). However, with multihomed systems or systems that are behind firewalls, it may be necessary to control the IP address that is advertised. By default, this parameter is undefined which causes the local IP address to be determined automatically. Default: `127.0.0.1`
sip.sipinfodtmf	Generation of DTMF via SIP INFO. If 0 (default), SIP INFO DTMF will not be sent. If 1, SIP INFO DTMF will be sent together with RFC2833 on RTP stream. If 2, SIP INFO DTMF will be sent with RFC2833 transmission disabled. Default: `0`
sip.usenullsdp	When an INVITE/REINVITE without SDP information need to be sent, this parameter specifies whether NULL SDP is used. 0: do not include SDP 1: include NULL SDP Possible values: `0, 1` Default: `0`
sip.warningheaders	sip.warningheaders will set when the platform sends warning headers. If 0, the platform will only send warning headers when the response is an error response. If 1, the platform will always send warning headers (if any). If 2, the platform will never send warning headers. Possible values: `0, 1, 2` Default: `0`

4.5.5 Transfer

Parameter	Description
sip.transfermethods	Transfer Methods for sip. • `HKF:` HookFlash. • `REFER:` `REFER`-based transfer. • `REFERJOIN:` consultative `REFER` transfer. • `MEDIAREDIRECT:` media redirect transfer. • `none:` No Transfer Methods for sip. Possible values: `HKF`, `REFER`, `REFERJOIN`, `MEDIAREDIRECT`, `none` Default: `REFER REFERJOIN`
sip.defaultblindxfer	Transfer Methods for sip. • `HKF:` HookFlash. • `REFER:` `REFER`-based transfer. • `BRIDGE:` `BRIDGE`-based transfer. • `REFERJOIN:` consultative `REFER` transfer. • `INBAND`: inband. • `MEDIAREDIRECT`: media redirect transfer. Possible values: `HKF`, `REFER`, `BRIDGE`, `REFERJOIN`, `INBAND`, `MEDIAREDIRECT` Default: `REFER`
sip.defaultconsultxfer	default consult type transfer method for sip. • `HKF:` HookFlash. • `BRIDGE:` bridge-based transfer. • `REFERJOIN:` consultative `REFER` transfer. • `MEDIAREDIRECT:` media redirect transfer. Possible values: `HKF`, `BRIDGE`, `REFERJOIN`, `MEDIAREDIRECT` Default: `REFERJOIN`
sip.defaultbridgexfer	default bridge type transfer method for sip Possible values: `BRIDGE`, `MEDIAREDIRECT` Default: `BRIDGE`

4.5.6 Hookflash Transfer

Parameter	Description
sip.hftype	hook flash transfer type for sip. • `0:` wait for disconnection. • `1:` force disconnection Possible values: `0, 1` Default: `0`
sip.hfdisctimer	The timeout value to terminate SIP hookflash transfer. For `Hookflash/wait for disconnect` mode, if a `BYE` is not received from remote end before this timeout, then the transfer is treated as failed (otherwise, the transfer is successful). For `Hookflash/initiate disconnect` mode, if a `BYE` is not received from remote end, then a `BYE` will be sent from local end after this timeout and the transfer is treated as successful whether `BYE` is received from remote end or generated from local end Default: `5000`
sip.hfprefix	SIP hookflash transfer dialing prefix. Example: `sip.hfprefix=none` means dial string is exactly as specified in `sip.hfprefix=!` would dial a hookflash, and then the pattern in `sip.hfprefix=*8,,` would dial a `*8` followed by two pause durations Default: !
sip.hfstopdial	digits to dial to stop a hookflash transfer. Character(s) to dial to abort a multi-phase hookflash. It will switch the connection back to original caller. Default: !

4.5.7 Refer Transfer

Parameter	Description
sip.referxferhold	Put originator on hold before refer or referjoin transfer. This specifies whether to put the original caller on hold (Invite hold) before sending the REFER for the transfer. 0 is off and 1 is on. Possible values: `0, 1` Default: `1`

Parameter	Description
sip.referxferwaitbye	Wait for remote to disconnect after `NOTIFY.` This specifies a timeout value to wait for `BYE` message from the remote end before sending `BYE` to disconnect the call. If it is zero, it will send `BYE` right after a `NOTIFY/200` is received. If it is non-zero, it will wait for the configured timeout (in milliseconds) before sending the `BYE.` Values are specified in millisecond. Default: `0`
sip.referxferwaitnotify	Specifies whether to wait for `NOTIFY` message before sending `BYE` to disconnect the call. If it is zero, it will send `BYE` right after a `202 Accepted` is received. If it is one, it will wait for a `NOTIFY` message before sending `BYE`. This parameter applies only to blind refer transfer. Default: `1`
sip.referredby	Specifies the header value of `Referred-By` in `REFER` message. `none` means no `Referred-By` header will be included in the `REFER` request. Empty (default) implies the local platform SIP URI (ie, `To` header for inbound call or `From` header for outbound call) for the dialog will be used.

4.6 H.323 Line Manager

4.6.1 General

Parameter	Description
h323.defaultani	Default H323 ANI Default: `0`
h323.h225portrange	`h323.h225portrange = (base port) (max port).` This value specifies a range of ports to be used for H.225 RAS and call signaling. Starting from the base port, odd port numbers will be used for RAS and even numbers for call signaling until the max port is reached. If the range is not specified, port 1719 and 1720 will be used for H.225 RAS and call signaling respectively. Please use an odd number for the base port and an even number for the max port, and ensure that all ports specified in the range are available. Default: `1719 1720`

Parameter	Description
h323.maxsessions	Specifies the maximum number of sessions each endpoint supports. Therefore, in multiple endpoint mode, the maximum number of H.323 sessions would be the number of endpoints multipled by the value of h323.maxsessions Default: `500`
h323.h225csconntimeout	Timeout value for connection in milliseconds Default: `10000`
h323.localrtpaddr	Local ip addresss for RTP sessions
h323.ras.endpointmode	Some gatekeeper (e.g. Avaya) requires the Media Platform to act like multiple endpoints that accepts one call per endpoint, instead of a single endpoint that accepts multiple calls. In multiple endpoint mode, each H.225 RAS registration request will use a unique set of H.225RAS/H.225CS ports. Therefore, multiple H.225RAS/H.225CS ports will be used if multiple H.225 RAS registration requests. The choice of H.225 ports should be specified using `h323.h225portrange.` • `multiple:` Media Platform as multiple endpoints • `single:` Media Platform as single endpoint Possible values: `multiple, single` Default: `single`
h323.h245.awaitmsdack	This value determines whether the platform is required to wait for MSD acknowledgement before a call can proceed. Possible values: `0, 1` Default: `0`
h323.h245.awaittcsack	This value determines whether the platform is required to wait for TCS acknowledgement before a call can proceed. Possible values: `0, 1` Default: `0`
h323.msdack.timeout	Timeout value for Master Slave Determination request for a new call. Default: `5000`
h323.tcsack.timeout	Timeout value for Terminal Capability Set request for a new call Default: `5000`

Parameter	Description
h323.arjreject_to_q931	When Gate keeper rejects ARJ return, the result should be mapped to Q.931 result and passed to the application. When `1` is chosen, Q.931 value assigned will be from `h323.arjreject_q931_code.` Possible values: `1, 0` Default: `1`
h323.arjreject_q931_code	What Q.931 code should be used when `h323.arjreject_to_q931` is `1. 21` is `CallRejected` Possible values: `21` Default: `21`
h323.usefaststart	Allows the h323 line manager to conduct calls using the `faststart` mechanism, where media negotiations are performed before the call is connected. These procedures are done after the call is connected under regular mode of operation. Possible values: `0, 1` Default: `0`
h323.usetunneling	Allows the h323 line manager to encapsulate (i.e. tunnel) all H.245 messages via the H.225 call control channel. Possible values: `0, 1` Default: `0`
h323.lcwaittimeout	After TCS, logical channels must be established within this timeout period, otherwise the call will be terminated Default: `10000`
h323.inbsetuptimeout	A `SETUP` message must be received within this timeout period once a connection is established, otherwise the connection will be terminated Default: `10000`

4.6.2 Media

Parameter	Description
h323.audio.frame.size	The audio frame size (RTP) expressed in milliseconds. Default: `20`

Parameter	Description
h323.codec	the audio codec used for audio transmission and reception Possible values: `pcmu, pcma, gsm, g726, g729` Default: `pcmu`
h323.supportRTPAudioTelephony Event	RTP Audio Telephony Event If set to `1,` the VoiceGenie platform will negotiate will the endpoints that it supports RTP Audio Telephony Events (DTMF etc). Clients such as Netmeeting, which does not support such events, will reject a call to be established seeing this capability, and thus for such clients the value must be set to `0. 0` is off and `1` is on. Possible values: `0, 1` Default: `1`

4.6.3 Call Routing – IP/PSTN gateway

Parameter	Description
h323.defaultgw	Address of the default H.323 gateway

4.6.4 Call Routing – H.323 Gatekeeper

Parameter	Description
h323.ras.registrationinfo	`h323.ras.registrationinfo = (gatekeeper1) (tech prefix) (extension) (password) (max concurrent calls), (tech prefix) (extension) (password) (max concurrent calls), .| (gatekeeper2) (tech prefix) (extension) (password) (max concurrent calls).....` The values specified in each entry are used for sending GRQ/RRQs to each defined gatekeeper. • `gatekeeper` – The IP address of the gatekeeper to register with • `tech prefix` – accepted values are digits `0`–`9,` up to 10 digits. For specifying `#,` please use `p` for the value if no `tech_prefix` is needed, the dash character (`-`) must be used. Note that when a `tech_prefix` is specified, the platform will be registered as a gateway instead of an endpoint. • `extension` – the alias/extension for this registration • `password` – the password for this registration. The value is ignored when `h323.ras.registrationassociationmode` is set to `0` • `max concurrent calls` – or defines the number of supported concurrent calls for each registration entry. This should be set to `1` under multiple endpoint operation mode.
h323.ras.OID	Algorithm ID for encryption under Registratin Association Mode. • `0: Avaya Algorithm Object ID “1.3.14.3.2.6”` • `1: Avaya Algorithm Object ID “2.16.840.1.114187.1.3”` Possible values: `0, 1` Default: `1`
h323.ras.terminaltype	Specifies the terminal type of the Media Platform used for registration. When Tech Prefix is used, the Media Platform will be registered as a gateway. Possible values: `terminal, gateway` Default: `terminal`

Parameter	Description
h323.ras.inarqmode	Turning ARQ requests on/off of in/outbound calls. The default value will be used when `h323.ras.endpointmode = multiple` or `h323.ras.registrationassociationmode = 1.` • `0:` do not submit ARQ • `1:` submit ARQ and call proceeds only if ACF returns from gatekeeper • `2:` submit ARQ and call proceeds even if ACF does not return from gatekeeper. Possible values: `0, 1, 2` Default: 0
h323.ras.outarqmode	Turning ARQ requests on/off of in/outbound calls. The default value will be used when `h323.ras.endpointmode = multiple` or `h323.ras.registrationassociationmode = 1.` • `0:` do not submit ARQ • `1:` submit ARQ and call proceeds only if ACF returns from gatekeeper • `2:` submit ARQ and call proceeds even if ACF does not return from gatekeeper. Possible values: `0, 1, 2` Default: 0
h323.ras.timeout.grq	Timeout vale for GRQ before next retry Default: 5
h323.ras.retrycount.grq	Number of consecutive retries for GRQ if Gatekeeper does not respond. After retrying such number of times the platform will retry GRQ again after a timeout defined by `h323.ras.timeout.grq.retry` Default: 2
h323.ras.timeout.rrq.urq	Timeout vale for RRQ/URQ before next retry Default: 3
h323.ras.retrycount.rrq	Number of consecutive retries for RRQ if Gatekeeper does not respond. After retrying such number of times the platform will retry GRQ again after a timeout defined by `h323.ras.timeout.rrq.retry` Default: 2

Parameter	Description
h323.ras.retrycount.urq	Number of consecutive retries for URQ if Gatekeeper does not respond. After retrying such number of times, unlike GRQ and RRQ, the platform will not retry URQ anymore and will consider this session as unregistered Default: `1`
h323.ras.timeout.arq.drq	Timeout value for ARQ/DRQ before determining the ARQ/DRQ as failure Default: `5`
h323.ras.timeout.grq.retry	After the number of GRQ failure exceeded the configured amount (`h323.ras.retrycount.grq`), the platform will retry GRQ again after this timeout Default: `40`
h323.ras.timeout.rrq.retry	After the number of RRQ failure exceeded the configured amount (`h323.ras.retrycount.rrq`), the platform will retry GRQ again after this timeout Default: `60`
h323.ras.timeout.keepalive.rrq	After RRQ is successful, the platform will use this timeout value to determine when to send a RRQ-keepalive message to the gatekeeper to maintain its registration. Note that a gatekeeper may return its desired value inside the RCF message. In that case that value will be used instead Default: `3600`
h323.ras.registrationassociationmode	Under Registration Association Mode, the platform will make use of the extension/password pairs provided in `h323.ras.registrationinfo`, by including the extension number as part of the GRQ message, and use the encryption algorithm as specified by `h323.ras.OID` to do proper encryption to the password that will be included in the RRQ message to register with a gatekeeper. `0` is off and `1` is on. Possible values: `0, 1` Default: `0`

4.6.5 Transfer

Parameter	Description
h323.usesamegwfortransfer	While multiple gateways are used in the environment, for certain H.323 transfers (such as H.450.2 transfer with two call legs) both the incoming and outgoing legs must land on the same gateway in order for the transfer to be successful. Setting this parameter to `1` overrides the `h323.defaultgw` config value and all outgoing calls (triggered by a transfer) will be directed to the same gateway where the inbound call is coming from If gatekeeper ARQ/ACF is used however, and if the gatekeeper decides to use a different gateway, the gateway address specified in the ACF will take precedence. `0` is off and `1` is on. Possible values: `0, 1` Default: `0`
h323.hfdisctimer	Timeout value for disconnection after a hookflash transfer in milliseconds Default: `5000`
h323.hfflashtimer	Hookflash timer in milliseconds Default: `500`
h323.hfdigittimer	Hookflash digit timer in milliseconds Default: `100`
h323.hfprefix	prefix for hookflash digit (character) Default: !
h323.transfermethods	Transfer Methods for sip. • `HKF:` HookFlash • `H450:` H.450.2 • `H450JOIN:` H.450 `JOIN` • `MEDIAREDIRECT:` media redirect transfer • `none:` No Transfer Methods for h323 Possible values: `HKF, H450, H450JOIN, MEDIAREDIRECT, none` Default: `none`

Parameter	Description
h323.hftype	hook flash transfer Type for h323. It will not be used if `h323.transfermode` is defined. • `0:` wait for disconnection • `1:` force disconnection Possible values: `0, 1` Default: 0
h323.defaultblindxfer	This specifies the default blind type transfer method for h323 Possible values: `HKF, BRIDGE, INBAND, H450, H450JOIN, MEDIAREDIRECT` Default: `BRIDGE`
h323.defaultconsultxfer	This specifies the default consult type transfer method for h323 Possible values: `BRIDGE, H450, H450JOIN, MEDIAREDIRECT` Default: `BRIDGE`
h323.defaultbridgexfer	This specifies the default bridge type transfer method for h323 Possible values: `BRIDGE, MEDIAREDIRECT` Default: `BRIDGE`
h323.hfmode	This specifies the default bridge type transfer method for h323. • `0:` Digit-by-digit H.245 signal. • `1:` One-shot H.245 alphanueric • `2:` RFC 2833 signalling Possible values: `0, 1, 2` Default: 0

4.6.6 H.450.2 Transfer

Parameter	Description
h323.h4502.timeout.awaitidentify response	Timeout value for protecting against the absence of a response to the a `CTIdentify.request` from the transferred-to party when trying to join two H.323 calls together. Default: 3000
h323.h4502.timeout.awaitinitiate response	Timeout value for protecting against failure to establish a new connection during a H.323 call transfer/call join. Default: 3000

Parameter	Description
h323.h4502.supportcalljoin	Turn off/on H.4502 call join support. • `0:` the platform will not attempt a multi-phase transfer. • `1:` the platform will be able to attempt a multi-phase transfer. Possible values: `0, 1` Default: `0`

4.6.7 Debugging

Parameter	Description
h323.logging	This entry is to turn on tracing message details at each specific level for H.323 communication. By default this entry should not be defined. It should be used only for debugging purposes. Valid values: `q931 h2250cs h245 RAS h4501`

4.7 Media Processing Component

The following parameters control media processing and transport behavior (Note that this component obsoletes the MTRTP implementation and the RTP configuration section):

Parameter	Description
mpc.localrtpaddr	mpc.localrtpaddr provides configurability of the connection part of SDP messages sent by the platform. If this parameter is not specified, then the IP Address of the local system will be used.
mpc.codec	list of codec corresponding to advertised capabilities offered by the platform using SDP. The offered codec list will control the codecs that are offered by the platform to the remote party for media sent from the remote party to VoiceGenie. Acceptable codecs are: pcmu pcma g726 g729 gsm amr aurora tfci h263 h263-1998 Default: `pcmu pcma`

Parameter	Description
mpc.codecpref	specify whether remote or local preferences will be used to interpret the accept codec list. If remote preferences are used, then the effective accept list will be the format list offered by the remote entity, filtered to include only those entries also on the locally configured list. If local preferences are used, then the local accept list will be used, but only including those capabilities offered by the remote entity. The "mpc.codecpref" parameter will be used to control this, and can be set to either 'r' (remote), or 'l' (local); the default value will be remote. Possible values: `r, l` Default: `r`
mpc.transmitmultiplecodec	When media negotiation returns more than one supported codecs, this parameter specifies whether to allow transmission of all supported codecs, or restrict transmission to only one codec. If set to 1 (default), more than one codec can be transmitted. If set to 0, only the codec at the top of the negotiated codec list will be transmitted. Note that for SIP devices that support multiple codecs, this parameter must be set to 0 for full call recording to work. Possible values: `0, 1` Default: `1`
mpc.appendrejcodec	When enabled, the platform will advertise all supported codecs when generating an SDP answer or SDP offer. Even if codecs are rejected or not presented in the caller's SDP, the platform will still support receiving these codecs. The platform will not send for those SDPs unless a payload is presented by the caller. Possible values: `0, 1` Default: `0`
mpc.dtmfpayload	Default DTMF payload to use by the platform if none are specified Default: `101`
mpc.amrpayload	Default payload type number to use for the AMR codec Default: `105`
mpc.tfcipayload	Default payload type number to use for tfci Default: `96`

Parameter	Description
mpc.h263_1998payload	Default payload type number to use for the H263-1998 codec Default: `99`
mpc.dtmf.singlepacket	By default, outgoing DTMF are represented by multiple (depends on `mpc.dtmf.duration`) RFC2833 packets followed by 3 RFC2833 packets with the End bit set. Enabling this parameter would force an outgoing DTMF to be represented by a single RFC2833 packet with the End bit set. Possible values: `0, 1` Default: `0`
mpc.rtp.portlow	Specifies the lowest RTP/RTCP port to be used by MPC. Default value is `1025.` Possible values are `1025` to `65535.` Default: `1025`
mpc.rtp.porthigh	Specifies the highest RTP/RTCP port to be used by MPC. Default value is `65535.` Possible values are `1025` to `65535.` Default: `65535`
mpc.rtp.packetseq	Specifies the sequence number for the first outgoing RTP packet. IF set to `0,` the first sequence number will be randomly generated for each RTP stream. Default value is `0.` Default: `0`
mpc.rtp.enablertcp	Specifies whether to transmit RTCP packets. Default is enable. Possible values: `0, 1` Default: `1`
mpc.rtp.restrictsource	Specifies whether to allow dropping packets from other sources (filtering). Possible values: `0 (disable), 1 (enable)` Default: `0`
mpc.rtp.audiobuffersize	Specifies the size of the RTP Audio Buffer. Default value is `20000` bytes. Default: `20000`
mpc.rtp.videobuffersize	Specifies the size of RTP Video Buffer. Default value is `120000` bytes. Default: `120000`

Parameter	Description
mpc.rtp.maxrtppacketsize	Specifies the maximum size of RTP packet. Default value is `20000` bytes. Default: `20000`
mpc.rtp.inputmode	Specifies the input mode of incoming RTP streams. Possible values: `continuous`, `vad` Default: `vad`
mpc.rtp.activetimeout	MPC will not send the first outgoing RTP packet until it received an incoming RTP packet or the RTP active timeout is reached. Default to `0` milliseconds, in which RTP packets will be transmitted immediately. Default: `0`
mpc.rtp.timeout	Specifies the RTP timeout value in milliseconds. A RTP stream will be considered inactive if there has been no activity for the timeout period. Default value is `60000` ms. Default: `6C0000`
mpc.rtp.rfc2429maxpacketsize	Specifies the maximum RTP packet size for RFC2429 packets in bytes. Any RFC2429 packet that exceeds the limit will be broken down into smaller packets. This parameter is used to prevent the OS from breaking down large RTP packets into multiple UDP packets, which may not be supported by some devices. Default value is 1472 - Linux will break down RTP packets greater than 1472 bytes into multiple UDP packets. Values less than 200 are not allowed. Set to 0 to disable the limit. Default: `1472`
mpc.asr.codec	List of SDP name of the codecs to be transmitted to the ASR. If not specified, `DEFAULT_AUDIO_FORMAT` will be used. Example of supported codecs: `pcmu pcma g726 g72632 aurora tfci qcelp`. The valid codec lists include: `one audio codec`, `one audio codec + tfci`, or `tfci` only. Default: `pcmu`
mpc.playsilencefill	Specifies the amount of silence fill in milliseconds to add at the end of prompt play. Default is 160ms. Setting to zero disables play silence fill. Default: `160`

Parameter	Description
mpc.rru.beginsilence	Specifies the amount of begin silence in milliseconds to insert for RRU. Default is 1000ms. Possible values are 0ms–10000ms. Default: 1000
mpc.rru.endsilence	Specifies the amount of end silence in milliseconds to insert for RRU. Default is 3000ms. Possible values are 0ms–10000ms. Default: 3000
mpc.mediamgr.audiobuffersize	Specifies the audio buffer size for the non-TTS source. Default value is 102400 bytes. Possible values are values >= 1024 bytes. Default: 102400
mpc.mediamgr.videobuffersize	Specifies the video buffer size for non-TTS source. Default value is 256000 bytes. Possible values are values >= 8000 bytes. It is recommended the size reflect the resolution of video. Default: 256000
mpc.mediamgr.rtsppause	Set to 1 if PAUSE request is supported by the RTSP server. Set to 0 otherwise. Default value is 1. This value can be overridden using the RTSP URL parameter "vg-rtspserver-pause". Default: 1
mpc.mediamgr.rtspplayrange	Set to 1 if the Range parameter in PLAY request is supported by the RTSP server. Set to 0 otherwise. Default value is 1. This value can be overridden using the RTSP URL parameter "vg-rtspserver-playrange". Default: 1
mpc.mediamgr.rtspupperbufferthreshold	If mpc.media.rtsppause is set to 1, PAUSE will be sent to stop RTP streaming when the packet buffer size has reached the upper threshold. Default value is 200 packets. This value can be overridden using the RTSP URL parameter "vg-rtsp-upperbufferthreshold". Default: 200
mpc.mediamgr.rtsplowerbufferthreshold	If mpc.media.rtsppause is set to 1, and RTP streaming has been paused. PLAY will be sent to resume RTP streaming if the packet buffer size has reduced to the lower threshold. Default value is 100 packets. The lower threshold must be smaller than the upper threshold. This value can be overridden using the RTSP URL parameter "vg-rtsp-lowerbufferthreshold". Default: 100

Parameter	Description
mpc.mediamgr.recordrtphinttrack	For ISO file container, recording a hint track for a media track into a recording file allows the file to be streamed when placed on a streaming server. Possible values: `0 (disable), 1 (enable)` Default: `0`
mpc.mediamgr.isofilerecordheadersize	The header of ISO file container grows as the content of the file grows. The platform will reserve the header size to harddrive before recording media of an ongoing session. The platform will actually record header at the end of the session and harddrive operations may be required if the reserved header size is not enough to accommodate the actual header size. Default: `55000`
mpc.mediamgr.h263overrideTR	This parameter is for video recording with H263 and H263-1998 video codec. Whenever video/audio out-of-sync happens in recorded files, enabling this parameter may solve the issue. By default, this parameter is enabled and the media platform uses an inherent property available in each H263 video sample called Temporal Reference to determine timing between each video sample in a recording session. Video/audio out-of-sync in recorded files, however, may occur if the Temporal References in the video frames are incorrect. Enabling this parameter will allow the media platform to correct Temporal Reference and try to synchronize video and audio during recording sessions. Disabling this parameter will keep the Temporal Reference intact. Possible values: `0 (disable), 1 (enable)` Default: `1`
mpc.mediamgr.rec_iframe_delay_threshold	This parameter is for video recording with audio and video. When starting a recorder, a few initial video frames may be dropped as the first self-referencing intra frame is not received for some reasons. As a result, audio duration received prior to receiving the next self-referencing video frame may be too long and it makes audio and video get out-of-sync. This parameter limits how long in milliseconds the audio is allowed in this situation without having to do video filling. The value of -1 will disable this feature. Default: `160`

Parameter	Description
mpc.transcoders	Specifies the list of transcoders to be used by MPC. PCM, GSM and G726 transcoders are loaded by default. Set to `none` in order to disable all transcoders. Possible values: `PCM`, `GSM`, `G726`, `G729`, AMR, `none` Default: `PCM GSM G726`
mpc.mediamgr.strictsamplingrate	The sampling rate that is officially supported for audio is 8000Hz and video is 90000Hz. Some media files may indicate a different sampling rate than what supported and tring to play those files may result in bad media quality. If this parameter is enabled, media files indicating any sampling rate other than officially supported will not be played. If this parameter is disabled, media files indicating any sampling rate other than supported will still be attempted to play by media platform but without guarantee quality. Possible values: `0, 1` Default: `0`
mpc.dsp.g729a	Specifies whether to use G.729 Annex A for G.729 transcoding. Possible values: `0, 1` Default: `0`
mpc.dsp.g726littleendian	Specifies whether to output transcoded data in little endian order. Possible values: `0, 1` Default: `0`
mpc.mixer.minvideoswitchtime	Specifies the minimum amount of the time that mixer video output is allowed to switch between different video input sources. Default value is `5000.` Default: `5000`
mpc.mixer.minsilencethreshold	Specifies the min silence threshold (0–32). Default value is `6.` Default: `6`
mpc.mixer.maxsilencethreshold	Specifies the max silence threshold (0–32). Default value is 32. Default: `32`
mpc.mixer.audiodelay_flush_all_threshold	Specifies the maximum difference between the current and expected packet time stamps when mixer flushes all buffered packets. Default value is 500 ms. Setting to zero disables flushing. Default: `500`

Parameter	Description
mpc.mixer.audiodelay_flush_silence_threshold	Specifies the maximum difference between the current and expected silent packet time stamps when mixer flushes silent buffered packets. Default value is 100 ms. Setting to zero disables flushing. Default: 100
mpc.fcr.gain	Gain on FCR input from call participants (-30 to 30 dB) Default: 0
mpc.amr.preferred_mode	Specifies the AMR Preferred Codec Mode. This is the value that the platform sends to the remote end as the preferred mode for AMR data sent to the platform. Set to a value in the range 0 to 7, or 15 to disable codec mode request. Default: 15

Parameter	Description
mpc.amr.fmtp	Specifies the AMR SDP RTP payload configurations offered and accepted by the platform. Set to one or more fmtp text values separated by the '|' character. The fmtp text is the same as would appear in the SDP negotitation (see RFC 3267). Each "|" separated value configures an AMR payload type. There are two fmtp parameters that can be set for each payload type, octet-align and mode-set. Setting octet-align=0 or octet-align=1 disables or enables octet align mode for the payload. Setting mode-set controls the AMR modes enabled for the payload. For example, setting "mode-set=0,1" enables modes 0 and 1. If mode-set is not set, all modes are enabled. Setting "mode-set=*" is a special value which will match any set of modes offered by the remote end. For example, setting this parameter to "octet-align=1; mode-set=*" enables one payload type with octet aligned mode enabled and any mode allowed, and setting it to "octet-align=0; mode-set=*|octet-align=1; mode-set=*" enables two payload types, one with bandwidth efficient mode enabled and any mode allowed, and one with octet aligned enabled and any mode allowed. Note, the mode-set parameter can cause transcoding to be required. For example, if a prompt to be played is in AMR format mode 5, but only mode 0 is enabled in the payload, a trancoder will be invoked to transcode from AMR mode 5 to AMR mode 0. Some AMR implementations may specify a fmtp options that are not actually activated for the payload. To work around this, the mpc.amr.fmtp can be set to "*". For this setting, all fmtp content in an SDP offer will be ignored and "octet-align=0" will be returned in the SDP answer. Similarly, an offer for this configuration will be set to "octet-align=0", and all fmtp content in the answer will be ignored. Default: `octet-align=0; mode-set=*| octet-align=1; mode-set=*`
mpc.amr.enable_dtx	This parameter controls whether the AMR transcoder will generate comfort noise frames when transcoding data to AMR format for which the voice activity detector indicates no speech. Set to 1 to enable or 0 to disable comfort noise frame generation. Possible values: `0, 1` Default: `1`

4.8 SRM Client Configuration Parameters

Parameter Name	Description
vrm.client.provision.file	Path to the SRM Client provision data file. It is used for stand alone test Default (Linux/Solaris): `/usr/local/phoneweb/config/vrmclient.dat` Default (Windows): `C:\VoiceGenie\mp\config\vrmclient.dat`
asr.load_once_per_call	When this parameter is equal to 1, there will be only one VRM session for the entire call which could have multiple recognition sessions. If the parameter value is not equal to 1, a VRM session is opened for each recognition request. The VRM session is closed when the recognition request is completed successfully or unsuccessfully (such as no match). As a result, there could be multiple VRM sessions in a call. Having multiple VRM sessions in a call could make the ASR server license usage more efficient. However, this configuration could have the following consequences: 1. There will be longer delays on speech barge in. 2. The save utterance data could be deleted by some recognizer servers after each VRM session. In that case, the VoiceXML application cannot refer to the saved utterance file after the recognition session. **Default:** `1`
asr.delay_for_dtmf	The amount of delay, in milliseconds, for starting the next ASR recognition after the last DTMF input from the previous field. The default value is `250.` Default: `250`
asr.log_metrics_to_asr	This parameter is only for ScanSoft Open Speech Recognizer. When enabled the Call Manager will log certain call metrics including Call Starts and Call Ends to the OSR server for the purposes of tuning Possible values: `0, 1` Default: `0`
vrm.client.dll	This configuration parameter defines the location of the SRM Client library to be used by the Media Platform.
vrm.client.grammar.path	This specifies the location of the built-in grammars residing on the VoiceGenie platform.

Parameter Name	Description
vrm.client.tmp.path	This specifies the location of the temporary directory used by the media platform. This must match the `PW_TMP` entry in the `voicexml.cfg` file used by the VoiceXML Interpreter.

vrm.client.vggrammarbase	This specifies the base-URL for translation of grammars residing under the subdirectory `vrm.client.tmp.path.` For example, in a Linux platform, if
	`vrm.client.tmp.path = /usr/local/phoneweb/tmp/`
	`vrm.client.vggrammarbase = /vggrammarbase/tmp`
	The file `/usr/local/phoneweb/tmp/index.txt` would be translated by the SRM client into `http://205.150.90.166/vggrammarbase/tmp/index.txt`
	where `205.150.90.166` is the IP address of the MP.
	These two options allow the temporary grammar generated by the media platform to be fetched by an offboard server.
	The web server defined in the “HTTP Access to Grammars” section of the *SRM User’s Guide* is used provide the hotkey grammars, so this configuration item must work together with the configuration defined in the web server.

vrm.client.timeout	This is the timeout value used by the SRM client to wait for a response from the MRCP server, for both the VoiceGenie SRM Server and the native MRCP servers. If a response to an MRCP request has is not received within this timeout period, then the request is deemed to have failed.
vrm.ping.frequency	This parameter defines, in milliseconds, the frequency in which the SRM client pings each of its servers. The MRCP `DESCRIBE` method is used as a ping message for each of the MRCP Servers provisioned.
vrm.ping.timeout	This parameter defines, in milliseconds, the timeout period for which we would be waiting for ping response from a MRCP server. If a ping response is not heard back from the server within this timeout, the SRM Client would consider the MRCP server to have become unavailable, and it would then disconnect from the server and periodically re-try connection to the MRCP server again.
vrm.client.max.noinput.timeout	This sets the value, in milliseconds, for the noinput timeout header that is sent to an MRCP engine. This should be set this to a large value, as the VoiceGenie Media Platform handles the no input timer. Default value is 90 seconds.

Parameter Name	Description
vrm.client.modules	This parameter lists the MRCP client protocol modules installed in the platform. In VG7.2.1, the value of this parameter can be any combinations of MRCPV1 and MRCPV2
vrm.client.MRCPV2.dll	In VG7.2.1 this parameter configures the MRCP v2 client library name and path.
vrm.client.MRCPV1.dll	In VG7.2.1 this parameter configures the MRCP v1 client library name and path
vrm.client.mrcpv2.prefix	In VG7.2.1 this parameter is used by the SIP stack. The specified prefix allows the SIP stack to choose a SIP port for MRCPV2 client.
mrcpv2client.sip.transport.0	In VG7.2.1 this parameter conjointly with the vrm.client.mrcpv2.prefix, specifies the SIP port used by the MRCPV2 Client. Note, the “mrcpv2client” must be the prefix specified by vrm.client.mrcpv2.prefix.
vrm.client.mrcpv2.maxopensocket	In VG7.2.1 this parameter specifies the maximum allowed sockets opened for MRCP sessions.
vrm.client.mrcpv2.earlynomatch	In VG7.2.1 setting to TRUE value of this parameter tells the MRCPv2 server must not wait for the end of speech before processing the collected speech to match active grammars.
vrm.client.ping.disable	In VG7.2.1 this parameter controls the MRCPv2 client sending OPTIONS as ping message to a MRCPv2 server. Setting value of this parameter to true will disable ping to all MRCP v2 servers.
stack.transport.type	The protocol type for the MRCP v1 stack Possible values: `RTSP` Default: `RTSP`
stack.connection.type	The type of the MRCP v1 stack handling. Possible values: `client` Default: `client`
stack.connection.timeout	The connection timeout for MRCP stack to establish a TCP connection to the server. Default: `10000`
stack.trace.debug	Whether to enable the stack debug message Possible values: `TRUE, FALSE` Default: `TRUE`

Parameter Name	Description
stack.socket.onesend	This parameter indicates whether to send a complete TCP message in one send request. Possible values: `TRUE, FALSE` Default: `TRUE`
vrm.client.grammar.path	This specifies the location of the builtin grammars residing on the VoiceGenie platform. Default (Linux/Solaris): `/usr/local/phoneweb/grammar/` Default (Windows): `C:\VoiceGenie\mp\grammar\`
vrm.client.tmp.path	This specifies the location of the temporary directory used by the media platform. This must match the `PW_TMP` entry in the `voicexml.cfg` file used by the interpreter. Default (Linux/Solaris): `/usr/local/phoneweb/tmp/` Default (Windows): `C:\VoiceGenie\mp\tmp\`
vrm.client.universals.uri	This gives the URI convention that the NextGen VXMLI uses to specifiy the universals gramamrs. The default value should be set to: vrm.client.universals.uri = builtin:grammar/universals Default: `builtin:grammar/universals`
vrm.client.logmetrics	This enables collection of MRCP message timing data. Possible values: `FALSE, TRUE` Default: `true`

4.9 Partition Definition (PortCount)

Parameter	Description
PortCount.Multicast.Interval	Interval of the multicast of local port count information Default: `500`
PortCount.Multicast.Timeout	Timeout for multicast message to expire Default: `2000`
PortCount.Multicast.Address	Multicast UDP address and port in `:` format Default: `225.0.0.1:9000`

Chapter

5 VoiceXML Interpreter Configuration

Parameter	Description
vxmli.maxEventsPerLoop	Parameter for tuning internal message processing dynamics Default: `10`
email.fromAddr	On Windows, this is the `From` header for maintainer e-mails. On Linux, it appears as the first line of the message body. Default: `nobody@example.com`
email.smtpAddr	SMTP server address for sending maintainer e-mails Default: `localhost`
VXML_DEFAULT	Default location of `defaults.vxml` Default (Linux/Solaris): `/usr/local/phoneweb/config/defaults.vxml` Default (Windows): `C:\VoiceGenie\mp\config\defaults.vxml`
VXML_VER	vxml version to use if not specified in page Possible values: `2.1, 2.0, 1.0` Default: `2.0`
vxmli.srvPortBase	The base port number of the tcp socket where vxmli listens for the connections. This is used in combination with the instance id of the vxmli to determine the socket port number the interpreter uses to listen for TCP connection from the Call Manager Default: `8506`

Parameter	Description
vxmli.ModuleType	This is the module type for use of communication library. Should always be set to `VXMLI` Default: `VXMLI`
PW_HOME	Home installation directory for Media Platform Default (Linux/Solaris): `/usr/local/phoneweb/` Default (Windows): `C:\VoiceGenie\mp\`
PW_AUDIO	Installation directory for audio files on Media Platform Default (Linux/Solaris): `/usr/local/phoneweb/audio` Default (Windows): `C:\VoiceGenie\mp\audio`
PW_BIN	Installation directory for executables on Media Platform Default (Linux/Solaris): `/usr/local/phoneweb/bin` Default (Windows): `C:\VoiceGenie\mp\bin`
PW_CONFIG	Installation directory for configuration files on Media Platform Default (Linux/Solaris): `/usr/local/phoneweb/config` Default (Windows): `C:\VoiceGenie\mp\config`
PW_GRAMMAR	Installation directory for grammaf files on Media Platform Default (Linux/Solaris): `/usr/local/phoneweb/grammar` Default (Windows): `C:\VoiceGenie\mp\grammar`
PW_LOGS	Directory files where logs are written to on Media Platform, when using the `<log>` tag with the dest attribute set to `file.` Default (Linux/Solaris): `/usr/local/phoneweb/logs` Default (Windows): `C:\VoiceGenie\mp\logs`
PW_TMP	Directory files where temporary files are written to on Media Platform Default (Linux/Solaris): `/usr/local/phoneweb/tmp` Default (Windows): `C:\VoiceGenie\mp\tmp`
SCRIPTDIR	Installation directory for pre-installed EcmaScript files on Media Platform Default (Linux/Solaris): `/usr/local/phoneweb/script` Default (Windows): `C:\VoiceGenie\mp\script`

Parameter	Description
ENGINEDIR	Installation directory for builtin grammar files on Media Platform Default (Linux/Solaris): `/usr/local/phoneweb/engine` Default (Windows): `C:\VoiceGenie\mp\engine`
ALTERNATE_INITIAL_PAGE	When a call is first presented to the interpreter, if the initial page indicated by the Call Manager cannot be fetched, and the Call Manager has not indicated an alternate page, this page will be used as the first page presented to the caller. The alternate page has to be a file on the local disk. Default (Linux/Solaris): `/usr/local/phoneweb/samples/alternatepage.vxml` Default (Windows): `C:\VoiceGenie\mp\samples\alternatepage.vxml`
BEEPAUDIO	The audio file containing the beep that is played at the end of a prompt, because recordings Default (Linux/Solaris): `/usr/local/phoneweb/audio/effects/endofprompt.vox` Default (Windows): `C:\VoiceGenie\mp\audio\effects\endofprompt.vox`
MAX_LOOP_COUNT	This value is used to detect loops in application execution. An error is raised if the number of internal states visited without encountering an input state exceeds this value Default: `50`
MAX_OPEN_LOG_FILES	This value is used to specify the maximum number of user log files that can be opened at one time. The least recently used opened user log file is closed if the number of opened user log files exceeds this value. Default: `5`
SAVE_UTTERANCE_AUDIO	This controls whether the saveutterance/utterance recording features are enabled on the platform Possible values: `1, 0` Default: `1`

Parameter	Description
USEMULTITMPDIR	This controls whether the interpreter will use one temp directory per call session. We should always enable this option Possible values: `1, 0` Default: `1`
USER_AGENT	HTTP request header `User-Agent` field contains information about the user agent originating the request. Expansion variable `$v` can be used to specify the platform version number. Default: `VoiceGenie NXP/$v`
GETINFO_PAIRS	It is used to set up the valid input parameter and its value for function `_VGGetInfo(parameter).` Each item in the list must be in the format `<param>=<value>,` for example `hostaddr=www.voicegenie.com.`
NON_SSML_ENGINES	This lists the set of TTS engines that does not support SSML Default: `ATIP|GVZ|FTTTS|PROFIVOX|RTSPTTS`
SUPPORTED_LANGUAGE	This lists the set of languages supported by VXML Default: `ce-HK|cn-HK|da-DK|de-AT|de-BE|de-CH|de-DE|el-GR|en-AU|en-BE|en-CH|en-GB|en-IN|en-SG|en-UK|en-US|es-ES|es-MX|es-US|eu-ES|fi-FI|fr-BE|fr-CA|fr-CH|fr-FR|it-CH|it-IT|ja-JP|ko-KO|ko-KR|nl-BE|nl-NL|no-NO|pl-PL|pt-BR|pt-PT|ru-RU|sk-SK|sv-SE|wa-BE|wv-SE|zh-CN|zh-TW|he-IL`
HTTP_ACCEPT	The HTTP `Accept:` header sent for HTTP requests. If not set, an internal default will be used
DEFAULT_ASR_LANGUAGE	This is the default language used by ASR engines. This may be overridden in the application with the `xml:lang` parameter in the `<grammar>` or the `<vxml>` tags. It must be one of the langauges in the `SUPPORTED_LANGUAGE` parameter
DEFAULT_TTS_LANGUAGE	This is the default language used by TTS engines. This may be override in the application with the `xml:lang` parameter in the `<prompt>` or the `<vxml>` tags. It must be one of the langauges in the `SUPPORTED_LANGUAGE` parameter
ASR_TIMEOUT	The interpreter will generate a nomatch event if no ASR result is returned within this timeout value after a bargein event is received. This is specified in seconds. Default: 60000

Parameter	Description
ENABLE_LOGDEST	This is a list of valid destinations to put in the dest attribute for the `<log>` tag Possible values: `metrics, file, bill, callmgr, syslog, calllog` Default: `metrics|file|bill|callmgr|syslog|calllog`
HTTP_VERSION	This is the HTTP version to be used by the interpreter to perform HTTP fetches Possible values: `HTTP/1.0, HTTP/1.1` Default: `HTTP/1.0`
VXMLI_FETCH_GRAMMAR_ENGINES	This is a list of ASR engines where the VXMLi will fetch the grammar instead of the ASR Engine Default: `PHONETICS|TTY`
WRITE_LOG_TO_DISK	This controls whether the email log messages are temporarily written to disk while the application is being executed. Possible values: `1, 0` Default: `0`
SPEECH_ASRENGINE_PRIORITY	This is the default priority for Speech Recognition Engine, in relation to DTMF and TDD recognizer priorities Default: `1`
TDD_ENGINE_PRIORITY	This is the default priority for TDD Recognition Engine, in relation to DTMF and Speech recognizer priorities Default: `0`
TDD_ENGINE_NAME	This is the internal name used to represent the TDD recognizer Default: `TTY`
ALLOW_FILE_URI	This parameter controls whether the platform allows `file:///` URI. Possible values: `TRUE, FALSE` Default: `TRUE`
DEFAULT_AUDIO_FORMAT	Choose the default audio format to be used by the interpreter Possible values: `MULAW, ALAW` Default: `MULAW`

Parameter	Description
vxmli.ac.enabled	Controls whether `<data>` access control validation is enabled. Possible values: `TRUE, FALSE` Default: `TRUE`
vxmli.ac.allow_if_missing	For `<data>`, determines the behaviour when fetched XML data doesn't contain any access-control processing instructions. Possible values: `TRUE, FALSE` Default: `FALSE`
vxmli.ac.use_platform_host_for_file_url	For `<data>`, determines the behaviour when the VoiceXML page accessing the XML data is a local file. Possible values: `TRUE, FALSE` Default: `TRUE`
SESSION_VARS	Each session variable entry is composed of three components. The first component is the session variable name as exposed within VoiceXML. The second component is the variable name sent back from the Call Manager. The third component indicates whether the session variable will be included in the request for the initial page URL. Default: `session.connection.answeredby|ANSWEREDBY|0|session.connection.uuiprotocol|UUIPROTOCOL|0|session.connection.redirect|REDIRECT|0|session.connection.aai|UUIDATA|0|session.connection.local.uri|LOCALURI|1|session.connection.remote.uri|REMOTEURI|1|session.connection.originator|ORIGIN|0|session.connection.channelidref|PSTNCHANNELID|1|session.connection.protocol.name|PROTOCOLNAME|0|session.connection.protocol.version|PROTOCOLVERSION|0|session.com.voicegenie.consultdata|consultdata|1|session.com.voicegenie.instance.parent|PARENT|1|session.connection.protocol.isup.natureofconnection.si|NatureOfConnection.SI|0|session.connection.protocol.isup.natureofconnection.cc|NatureOfConnection.CCI|0|session.connection.protocol.isup.natureofconnection.ec|NatureOfConnection.EC|0|session.connection.protocol.isup.originalcallednumber.num|OriginalCalledNumber.num|0|session.connection.protocol.isup.originalcallednumber.nai|OriginalCalledNumber.NAI|0|session.connection.protocol.isup.originalcallednumbe...`

Parameter	Description
vxmli.default_transfer_connect_timeout	For `<transfer>`, determines the default value for the connecttimeout attribute. Default: `30`
vxmli.default_xmllang	This parameter is the language to use if the `XML:LANG` attribute is not specified in the prompt or grammar tag and it is not specified in the VXML tag. Default: `en-US`
vxmli.default_record_type	This parameter is the default record type to use if the type attribute is not specified in the record tag. Default: `audio/x-vox`
RESTART_AFTER_N_CALLS	The vxmli should quiese after this number of calls Default: `0`
RESTART_AT_TIME_OF_DAY	The vxmli should quiese after this number of calls Default: `0`
vxmli.enable_bt	Enables BT Possible values: `TRUE`, `FALSE` Default: `FALSE`
vxmli.enable_lc	Enables LC Possible values: `TRUE`, `FALSE` Default: `FALSE`
vxmli.lc.loglevel	Sets the LC log level Default: `4`
vxmli.use_external_dtmf_recognizer	Enables external DTMF recognizer support Possible values: `TRUE`, `FALSE` Default: `FALSE`
vxmli.use_external_input_timeout	Enables external input timeout support Possible values: `TRUE`, `FALSE` Default: `FALSE`
vxmli.remote_audio_url_prefix	Specifies the URL prefix for remote built-in audio files

Parameter	Description
vxmli.srgs_file_extension	Specifies the file extension to use for implicit grammars that are saved as SRGS Default: `grxml`
vxmli.gsl_file_extension	Specifies the file extension to use for implicit grammars that are saved as GSL Default: `gsl`
vxmli.enable_grammar_caching	Enables XML Grammar Caching Possible values: `TRUE, FALSE` Default: `FALSE`
vxmli.simple_list_use_substrings	Enables pattern substring generation for simple list grammar processing. Possible values: `TRUE, FALSE` Default: `TRUE`
vxmli.strict_caching	Setting this option to true enables the interpreter to perform HTTP/1.1 compliant caching. Possible values: `TRUE, FALSE` Default: `FALSE`
vxmli.start_log_vars	This is a list of call-related fields that will be logged along with an `appl_begin` metrics entry. Default: `ANI|DNIS|INIT_URL|DEFAULTS|UUIDATA|PROTOCOLNAME|PROTOCOLVERSION|CALLIDREF`
cmp.proxy	The IP or hostname of the CMP Proxy that CLC to connect to Default: `127.0.0.1`
cmp.proxy_port	The port number of the CMP proxy to connect to Default: `8700`
cmp.heartbeat	The interval, in seconds, to send a periodic heartbeat message from the component to the CMP Proxy Default: `20`
cmp.reconnect	The interval, in seconds, between reconnection attempts to the CMP Server Default: `5`

Parameter	Description
cmp.sync	Specifies whether the configuration should be synchronized with the cmp database Possible values: `FALSE, TRUE` Default: `TRUE`
cmp.log_file	This full path to the log file of the VXML Interpreter Default (Linux/Solaris): `/usr/local/phoneweb/logs/CMP.log.vxmli` Default (Windows): `C:\VoiceGenie\mp\logs\CMP.log.vxmli`
cmp.size_option	Rollover all log files by size or by time Possible values: `FALSE, TRUE` Default: `TRUE`
cmp.rollover_size	The size limit, in MB, for rollover when rolling over by size Default: `10`
cmp.num_rollover_files	The number of files to roll through before they are overwritten when rolling over by size Default: `5`
cmp.rollover_mins	The interval of time, in minutes, between rollover when rolling over by time Default: `1440`
cmp.rollover_time	The time at which the log files are rolled over when rolling over by time Default: `4:00`
cmp.email	If the `EMAIL` sink is specified, the email address be used Default: `name@domain.com`
cmp.log_sinks	Sinks that will be used by this component,possible sinks are: `FILE, UPSTREAM, SYSLOG, SNMP, EMAIL` Default: `FILE|UPSTREAM`
cmp.trace_flag	Determines if logs at level `log_5` (tracing/debugging) should be logged Possible values: `FALSE, TRUE` Default: `FALSE`

Parameter	Description
cmp.pid_option	Appends PID of the process to the name of the trace file so that they are not overwritten when the process restarts Possible values: FALSE, TRUE Default: FALSE
cmp.log_queue_limit	The number of logs that can be queued for processing before the calling thread is throttled so that the logging thread does not fall behind indefinitely Default: 5000
cmp.log_write_buffer_size	The size of the buffer, in bytes, for log event preallocation Default: 2560
cmp.log_write_buffer	The size of the buffer, in bytes, to be used for block writing to the disk, a value of 0 implies no buffering Default: 65536
cmp.log_write_buffer_stale_timeout	The longest time that a log can remain in the buffer before being written to disk Default: 2000
cmp.log_write_buffer_idle_timeout	The amount of time during which no logs are received after which the buffer is written to disk Default: 1000
cmp.metrics	log mask for metrics data Default: 0|1
cmp.log_0	Log mask for data logged at log level 0 Default: 1111111111111111111111111111111111111111111111111 1111111111111111111111111111111111111111111111111 1111111111111111111111111111111111111111111111111 111111111111111111111111111111111111111111111111| 1111111111111111111111111111111111111111111111111 1111111111111111111111111111111111111111111111111 1111111111111111111111111111111111111111111111111 111111111111111111111111111111111111111111111111

Parameter	Description
cmp.log_1	Log mask for data logged at log level 1 Default: 111111111111111111111111111111111111111111111 111111111111111111111111111111111111111111111 111111111111111111111111111111111111111111111 11111111111111111111111111111111111111111111| 111111111111111111111111111111111111111111111 111111111111111111111111111111111111111111111 11111111111111xxxxxxxxxxxxxxxxxxxxxxxxxxxxxx1 11111111111111111111111111111111111111111111
cmp.log_1.0x030.UPSTREAM	Allowed specifiers for data logged at log level 1, module 0x030 Default: 100000–299999
cmp.log_1.0x031.UPSTREAM	Allowed specifiers for data logged at log level 1, module 0x031 Default: 100000–299999
cmp.log_1.0x032.UPSTREAM	Allowed specifiers for data logged at log level 1, module 0x032 Default: 100000–299999
cmp.log_1.0x033.UPSTREAM	Allowed specifiers for data logged at log level 1, module 0x033 Default: 100000–299999
cmp.log_1.0x034.UPSTREAM	Allowed specifiers for data logged at log level 1, module 0x034 Default: 100000–299999
cmp.log_1.0x035.UPSTREAM	Allowed specifiers for data logged at log level 1, module 0x035 Default: 100000–299999
cmp.log_1.0x036.UPSTREAM	Allowed specifiers for data logged at log level 1, module 0x036 Default: 100000–299999
cmp.log_1.0x037.UPSTREAM	Allowed specifiers for data logged at log level 1, module 0x037 Default: 100000–299999
cmp.log_1.0x038.UPSTREAM	Allowed specifiers for data logged at log level 1, module 0x038 Default: 100000–299999
cmp.log_1.0x039.UPSTREAM	Allowed specifiers for data logged at log level 1, module 0x039 Default: 100000–299999
cmp.log_1.0x03A.UPSTREAM	Allowed specifiers for data logged at log level 1, module 0x03A Default: 100000–299999

Parameter	Description
cmp.log_1.0x03B.UPSTREAM	Allowed specifiers for data logged at log level 1, module 0x03B Default: 100000–299999
cmp.log_1.0x03C.UPSTREAM	Allowed specifiers for data logged at log level 1, module 0x03C Default: 100000–299999
cmp.log_1.0x03D.UPSTREAM	Allowed specifiers for data logged at log level 1, module 0x03D Default: 100000–299999
cmp.log_1.0x03E.UPSTREAM	Allowed specifiers for data logged at log level 1, module 0x03E Default: 100000–299999
cmp.log_1.0x03F.UPSTREAM	Allowed specifiers for data logged at log level 1, module 0x03F Default: 100000–299999
cmp.log_1.0x040.UPSTREAM	Allowed specifiers for data logged at log level 1, module 0x040 Default: 100000–299999
cmp.log_1.0x041.UPSTREAM	Allowed specifiers for data logged at log level 1, module 0x041 Default: 100000–299999
cmp.log_1.0x042.UPSTREAM	Allowed specifiers for data logged at log level 1, module 0x042 Default: 100000–299999
cmp.log_1.0x043.UPSTREAM	Allowed specifiers for data logged at log level 1, module 0x043 Default: 100000–299999
cmp.log_1.0x044.UPSTREAM	Allowed specifiers for data logged at log level 1, module 0x044 Default: 100000–299999
cmp.log_1.0x045.UPSTREAM	Allowed specifiers for data logged at log level 1, module 0x045 Default: 100000–299999
cmp.log_1.0x046.UPSTREAM	Allowed specifiers for data logged at log level 1, module 0x046 Default: 100000–299999
cmp.log_1.0x047.UPSTREAM	Allowed specifiers for data logged at log level 1, module 0x047 Default: 100000–299999
cmp.log_1.0x048.UPSTREAM	Allowed specifiers for data logged at log level 1, module 0x048 Default: 100000–299999

Parameter	Description
cmp.log_1.0x049.UPSTREAM	Allowed specifiers for data logged at log level 1, module 0x049 Default: 100000–299999
cmp.log_1.0x04A.UPSTREAM	Allowed specifiers for data logged at log level 1, module 0x04A Default: 100000–299999
cmp.log_1.0x04B.UPSTREAM	Allowed specifiers for data logged at log level 1, module 0x04B Default: 100000–299999
cmp.log_1.0x04C.UPSTREAM	Allowed specifiers for data logged at log level 1, module 0x04C Default: 100000–299999
cmp.log_1.0x04D.UPSTREAM	Allowed specifiers for data logged at log level 1, module 0x04D Default: 100000–299999
cmp.log_1.0x04E.UPSTREAM	Allowed specifiers for data logged at log level 1, module 0x04E Default: 100000–299999
cmp.log_1.0x04F.UPSTREAM	Allowed specifiers for data logged at log level 1, module 0x04F Default: 100000–299999
cmp.log_2	Log mask for data logged at log level 2 Default: 11111111111111111111111111111111111111111111111111 11111111111111111111111111111111111111111111111111 11111111111111111111111111111111111111111111111111 1111111111111111111111111111111111111111111111111| 11111111111111111111111111111111111111111111111111 11111111111111111111111111111111111111111111111111 111111111111111xxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxxx1 1111111111111111111111111111111111111111111111111
cmp.log_2.0x030.UPSTREAM	Allowed specifiers for data logged at log level 2, module 0x030 Default: 100000–299999
cmp.log_2.0x031.UPSTREAM	Allowed specifiers for data logged at log level 2, module 0x031 Default: 100000–299999
cmp.log_2.0x032.UPSTREAM	Allowed specifiers for data logged at log level 2, module 0x032 Default: 100000–299999
cmp.log_2.0x033.UPSTREAM	Allowed specifiers for data logged at log level 2, module 0x033 Default: 100000–299999

Parameter	Description
cmp.log_2.0x034.UPSTREAM	Allowed specifiers for data logged at log level 2, module 0x034 Default: 100000–299999
cmp.log_2.0x035.UPSTREAM	Allowed specifiers for data logged at log level 2, module 0x035 Default: 100000–299999
cmp.log_2.0x036.UPSTREAM	Allowed specifiers for data logged at log level 2, module 0x036 Default: 100000–299999
cmp.log_2.0x037.UPSTREAM	Allowed specifiers for data logged at log level 2, module 0x037 Default: 100000–299999
cmp.log_2.0x038.UPSTREAM	Allowed specifiers for data logged at log level 2, module 0x038 Default: 100000–299999
cmp.log_2.0x039.UPSTREAM	Allowed specifiers for data logged at log level 2, module 0x039 Default: 100000–299999
cmp.log_2.0x03A.UPSTREAM	Allowed specifiers for data logged at log level 2, module 0x03A Default: 100000–299999
cmp.log_2.0x03B.UPSTREAM	Allowed specifiers for data logged at log level 2, module 0x03B Default: 100000–299999
cmp.log_2.0x03C.UPSTREAM	Allowed specifiers for data logged at log level 2, module 0x03C Default: 100000–299999
cmp.log_2.0x03D.UPSTREAM	Allowed specifiers for data logged at log level 2, module 0x03D Default: 100000–299999
cmp.log_2.0x03E.UPSTREAM	Allowed specifiers for data logged at log level 2, module 0x03E Default: 100000–299999
cmp.log_2.0x03F.UPSTREAM	Allowed specifiers for data logged at log level 2, module 0x03F Default: 100000–299999
cmp.log_2.0x040.UPSTREAM	Allowed specifiers for data logged at log level 2, module 0x040 Default: 100000–299999
cmp.log_2.0x041.UPSTREAM	Allowed specifiers for data logged at log level 2, module 0x041 Default: 100000–299999

Parameter	Description
cmp.log_2.0x042.UPSTREAM	Allowed specifiers for data logged at log level 2 , module 0x042 Default: 100000–299999
cmp.log_2.0x043.UPSTREAM	Allowed specifiers for data logged at log level 2 , module 0x043 Default: 100000–299999
cmp.log_2.0x044.UPSTREAM	Allowed specifiers for data logged at log level 2 , module 0x044 Default: 100000–299999
cmp.log_2.0x045.UPSTREAM	Allowed specifiers for data logged at log level 2 , module 0x045 Default: 100000–299999
cmp.log_2.0x046.UPSTREAM	Allowed specifiers for data logged at log level 2 , module 0x046 Default: 100000–299999
cmp.log_2.0x047.UPSTREAM	Allowed specifiers for data logged at log level 2 , module 0x047 Default: 100000–299999
cmp.log_2.0x048.UPSTREAM	Allowed specifiers for data logged at log level 2 , module 0x048 Default: 100000–299999
cmp.log_2.0x049.UPSTREAM	Allowed specifiers for data logged at log level 2 , module 0x049 Default: 100000–299999
cmp.log_2.0x04A.UPSTREAM	Allowed specifiers for data logged at log level 2 , module 0x04A Default: 100000–299999
cmp.log_2.0x04B.UPSTREAM	Allowed specifiers for data logged at log level 2 , module 0x04B Default: 100000–299999
cmp.log_2.0x04C.UPSTREAM	Allowed specifiers for data logged at log level 2 , module 0x04C Default: 100000–299999
cmp.log_2.0x04D.UPSTREAM	Allowed specifiers for data logged at log level 2 , module 0x04D Default: 100000–299999
cmp.log_2.0x04E.UPSTREAM	Allowed specifiers for data logged at log level 2 , module 0x04E Default: 100000–299999
cmp.log_2.0x04F.UPSTREAM	Allowed specifiers for data logged at log level 2 , module 0x04F Default: 100000–299999

Parameter	Description

cmp.log_3	Log mask for data logged at log level 3
	Default:
	`111111111111111111111111111111111111111111111111`
	`111111111111111111111111111111111111111111111111`
	`111111111111111111111111111111111111111111111111`
	`11111111111111111111111111111111111111111111111|`
	`000000000000000000000000000000000000000000000000`
	`000000000000000000000000000000000000000000000000`
	`000000000000000000000000000000000000000000000000`
	`00000000000000000000000000000000000000000000000`

cmp.log_4	Log mask for data logged at log level 4
	Default:
	`111111111111111111111111111111111111111111111111`
	`111111111111011111111111111111111111111111111111`
	`11111111111111111111111111111111111111111111|000`
	`000000000000000000000000000000000000000000000000`
	`000000000100000000000000000000000000000000000000`
	`00000000000000000000000000000000000000000`

cmp.log_5	Log mask for data logged at log level 5
	Default:
	`111111111111111111111111111111111111111111111111`
	`111111111111111111111111111111111111111111111111`
	`111111111111111011111111111111111111111111111111`
	`11111111111111111111111111111111111111111111111|`
	`000000000000000000000000000000000000000000000000`
	`000000000000000000000000000000000000000000000000`
	`000000000000000100000000000000000000000000000000`
	`00000000000000000000000000000000000000000000000`

cmp.guaranteed_logs_to_file	Specify if logs that are guaranteed to be sent upstream should be logged to a temp file Possible values: `FALSE`, `TRUE` Default: `TRUE`
cmp.unsent_log_file	Specify the name of the temp log file to log to if `cmp.guaranteed_logs_to_file` Default (Linux/Solaris): `/usr/local/phoneweb/logs/guaranteed.log.vxmli` Default (Windows): `C:\VoiceGenie\mp\logs\guaranteed.log.vxmli`

Parameter	Description
cmp.UTC.#	UTC or Local Time Logging Possible values: `TRUE, FALSE` Default: `FALSE`
vxmli.tmpcleantime	Installation parameter, dictates how many days after which temp files are considered stale and deleted Default: `59`
cleantime	Installation parameter, dictates how many days of metrics files are kept on the system Default: `59`
vxml_style	Installation parameter, dictates how stringent the interpreter is Possible values: `1, 0` Default: `1`
VXMLI_GRAM_SRC_CONFORM	Specifies whether the grammar tag will abide by VoiceXML 2.1 Conformance when the mode attribute is omitted for external grammars Possible values: `FALSE, TRUE` Default: `FALSE`
package_sub_type	A parameter used by installation to determine the OS that this package will be installed on. Default (Linux/Solaris): `##subtype##` Default (Windows): `##subtype##`
vxmli.config_inputmode	A parameter used by installation to determine if this Media Platform will be DTMF input only or DTMF and Voice. Default: `1`

Chapter

6 Fetching Module Configuration

Parameter	Description
iproxy.http_proxy	IP address and port of HTTP proxy to use. If disabled, the pwproxy will not use HTTP proxy. Default: `127.0.0.1:3128`
iproxy.https_proxy	IP address and port of HTTPS proxy to use. If disabled, the pwproxy will not use HTTPS proxy. Default: `127.0.0.1:3128`

Parameter	Description
iproxy.connect_timeout	The maximum time in seconds allowed for connecting to a server. If enabled, the smaller of iproxy.connect_timeout and the fetch response timeout will be used. If disabled, the fetch response timeout will be used. Note that when this parameter is enabled, the minimum timeout interval is 1 second. It is not possible to use milliseconds. In addition, if the SSL connection phase does not finish before time timeout is reached, the fetch will be aborted. (i.e. the fetch request will not be successful (on the server side)) If the SSL connection phase has completed but the rest of the fetch does not finish before the timeout is reached, the fetch request will be successful (on the server side). When this parameter is disabled, the timeout interval has no minimum value. The fetch request will always be successful even if the timeout is reached from the VoiceGenie/VoiceXML side. However, note that the fetch cannot timeout until after the SSL connection phrase is finished, even if that takes longer than the timeout interval. After the SSL connection phase is done, the value of this parameter no longer applies, even if it is smaller than the fetchtimeout value. For example, if the SSL connection phase ends after 2 seconds but the rest of the fetch takes longer to finish, and `iproxy.connect_timeout=5s` and `fetchtimeout=10s`, the fetch would not time out until after 10 seconds. Default: `5`
iproxy.curl_handle_fetchtimeout	If set to `FALSE`, fetch timeouts are handled by the Fetching Module itself; when a timeout occurs, a fetch timeout result will be returned to the host application but the actual HTTP fetch operation won't be aborted and will be carried out to completion. If set to `TRUE`, fetch timeouts are handled by cURL; when a timeout occurs, a fetch timeout result will be returned to the host application and the actual HTTP fetch operation will be aborted by cURL, regardless of being in a connection, request or response phase. Possible values: `TRUE`, `FALSE` Default: `false`
iproxy.max_connections	Max. number of concurrent active connections between iproxy and the HTTP proxy/server Default: `1000`

Parameter	Description
iproxy.health_level	Health status reporting level. With a higher number, more information is displayed in the CLC and the SMC health reporting. Currently only supports two levels (1 and 2) Default: 2
iproxy.max_redirections	Max. number of redirections allowed on a fetch request. Default: 5
iproxy.use_strict_caching_rules	When set to `true,` the Fetching Module will perform strictly HTTP/1.1 conformant caching. Setting this to `false` offers better performance. Possible values: `true, false` Default: `true`
iproxy.cache_max_size	Maximum size of the shared memory cache in MBytes Default: 64
iproxy.cache_max_age	Maximum age for data cached in iproxy in seconds (default is 60). It applies only if data is cacheable. iproxy caching could be turned off by setting this to `0.` Default: 60
iproxy.cache_error_max_age	Maximum age of cache for failed fetches in seconds. Default: 0
iproxy.no_cache_url_substr	If a URL contains any one of the sub-strings in this list, it will not be cached. Default: `cgi-bin`
iproxy.cache_file_format	Format of temp file name for cached entries. These temp files are currently used for grammars when ASR cannot fetch on its own. They can also be used for debugging purposes. Default (Linux/Solaris): `/usr/local/phoneweb/cache/tmp/%x` Default (Windows): `C:\VoiceGenie\mp\cache\tmp\%x`
iproxy.max_shmem_entry	Maximum size (MBytes) of cache entry that is permitted in shared memory. Anything larger will be cached as a memory mapped file. Default: 32

Parameter	Description
iproxy.mem_file_format	The path and format of file name of a memory mapped file for cache. Default (Linux/Solaris): `/usr/local/phoneweb/cache/mem/%x` Default (Windows): `C:\VoiceGenie\mp\cache\mem\%x`
iproxy.user_agent	HTTP request header `User-Agent` field contains information about the user agent originating the request. Default: `PMLI/1.1`
iproxy.http_accept	A list of mime types for the default value of the `Accept` directive in HTTP header. Default: `application/grammar-xml, application/octet-stream, application/x-abnf, application/x-javascript, application/x-jsgf, application/x-ms-xml, application/x-swi-grammar-compiled, application/x-voicegenie-nuance, application/x-voicegenie-watson, application/srgs+xml, application/ccxml+xml, audio/*, video/*, text/html, text/plain, text/vxml, text/x-vxml, text/xml`
iproxy.no_x_session_id	By enabling this parameter, the `X-Session-Id` header will not be set in HTTP requests Possible values: `true` Default: `true`
iproxy.http_debug	If this is set, the debug info will be printed into the trace file. Possible values: `true, false` Default: `true`
iproxy.cached_easy_handles	The number of cURL easy handles to create and place in a pool at startup. If this value is `0,` handles will be created for each fetch (no handles will be reused). Default: `0`
iproxy.use_connection_caching	If this is set, the cURL easy handles will reuse their connections. Possible values: `true, false` Default: `true`

Parameter	Description
iproxy.ssl_cert	The file name of your certificate. The default format is `PEM` and can be changed with the configuration parameter `iproxy.ssl_cert_type`
iproxy.ssl_cert_type	The format of the certificate. Possible values: `PEM, DER` Default: `PEM`
iproxy.ssl_key	The file name of the private key. The default format for the key is `PEM` and may be changed by the parameter `iproxy.ssl_key_type.`
iproxy.ssl_key_type	The format of the private key. Possible values: `PEM, DER, ENG` Default: `PEM`
iproxy.ssl_key_passwd	The password required to use the `iproxy.ssl_key.`
iproxy.ssl_engine	The identifier for the crypto engine you want to use for your private key.
iproxy.ssl_engine_default	Sets the actual crypto engine as the default for (asymetric) crypto operations.
iproxy.ssl_version	Set what version of SSL to attempt to use. By default, the SSL library will try to solve this by itself although some servers make this difficult why you at times may have to use this option. Possible values: `2, 3` Default: `2`
iproxy.ssl_verify_peer	Do you want verify the peer's certificate. When this option is set, you should set one of `iproxy.ssl_ca_info` or `iproxy.ssl_ca_path.` Possible values: `0, 1` Default: `0`
iproxy.ssl_ca_info	The file name holding one or more certificates to verify the peer with.
iproxy.ssl_ca_path	The path holding multiple CA certificates to verify the peer with. The certificate directory must be prepared using the openssl `c_rehash` utility.
iproxy.ssl_random_file	The path to a file which is read from to seed the random engine for SSL.

Parameter	Description
iproxy.ssl_verify_host	Should the Common name from the peer certificate in the SSL handshake be verified? Possible values: `0, 1, 2` Default: `0`
iproxy.ssl_cipher_list	The list of ciphers to use for the SSL connection. The list must be syntactly correct, it consists of one or more cipher strings separated rated by colons. Commas or spaces are also acceptable separators but colons are normally used, `,` `-` and `+` can be used as operators. Valid examples of cipher lists include `RC4-SHA`, `SHA1+DES`, `TLSv1` and `DEFAULT`. You'll find more details about cipher lists on this URL: `http://www.openssl.org/docs/apps/ciphers.html.` Default: `0`
cmp.proxy	The IP or hostname of the CMP Proxy that CLC to connect to Default: `127.0.0.1`
cmp.proxy_port	The port number of the CMP proxy to connect to Default: `8700`
cmp.heartbeat	The interval, in seconds, to send a periodic heartbeat message from the component to the CMP Proxy Default: `20`
cmp.reconnect	The interval, in seconds, between reconnection attempts to the CMP Server Default: `5`
cmp.sync	Specifies whether the configuration should be synchronized with the cmp database Possible values: `FALSE`, `TRUE` Default: `TRUE`
cmp.log_file	This full path to the log file of the iProxy Default (Linux/Solaris): `/usr/local/phoneweb/logs/CMP.log.pwproxy` Default (Windows): `C:\VoiceGenie\mp\logs\CMP.log.pwproxy`

Parameter	Description
cmp.size_option	Rollover all log files by size or by time Possible values: `FALSE`, `TRUE` Default: `TRUE`
cmp.rollover_size	The size limit, in MB, for rollover when rolling over by size Default: `10`
cmp.num_rollover_files	The number of files to roll through before they are overwritten when rolling over by size Default: `5`
cmp.rollover_mins	The interval of time, in minutes, between rollover when rolling over by time Default: `1440`
cmp.rollover_time	The time at which the log files are rolled over when rolling over by time Default: `4:00`
cmp.email	If the `EMAIL` sink is specified, the email address be used Default: `name@domain.com`
cmp.log_sinks	Sinks that will be used by this component,possible sinks are: `FILE`, `UPSTREAM`, `SYSLOG`, `SNMP`, `EMAIL` Default: `FILE|UPSTREAM`
cmp.trace_flag	Determines if logs at level `log_5` (tracing/debugging) should be logged Possible values: `FALSE`, `TRUE` Default: `FALSE`
cmp.pid_option	Appends PID of the process to the name of the trace file so that they are not overwritten when the process restarts Possible values: `FALSE`, `TRUE` Default: `FALSE`
cmp.log_queue_limit	The number of logs that can be queued for processing before the calling thread is throttled so that the logging thread does not fall behind indefinitely Default: `5000`

Parameter	Description
cmp.log_write_buffer_size	The size of the buffer, in bytes, for log event preallocation Default: 2560
cmp.log_write_buffer	The size of the buffer, in bytes, to be used for block writing to the disk, a value of 0 implies no buffering Default: 65536
cmp.log_write_buffer_stale_timeout	The longest time that a log can remain in the buffer before being written to disk Default: 2000
cmp.log_write_buffer_idle_timeout	The amount of time during which no logs are received after which the buffer is written to disk Default: 1000
cmp.metrics	log mask for metrics data Default: 0|1
cmp.log_0	Log mask for data logged at log level 0 Default: 1111111111111111111111111111111111111111111111111111 1111111111111111111111111111111111111111111111111111 1111111111111111111111111111111111111111111111111111 111111111111111111111111111111111111111111111111111| 1111111111111111111111111111111111111111111111111111 1111111111111111111111111111111111111111111111111111 1111111111111111111111111111111111111111111111111111 111111111111111111111111111111111111111111111111111
cmp.log_1	Log mask for data logged at log level 1 Default: 1111111111111111111111111111111111111111111111111111 1111111111111111111111111111111111111111111111111111 1111111111111111111111111111111111111111111111111111 111111111111111111111111111111111111111111111111111| 1111111111111111111111111111111111111111111111111111 1111111111111111111111111111111111111111111111111111 1111111111111111111111111111111111111111111111111111 111111111111111111111111111111111111111111111111111

Parameter	**Description**
cmp.log_2	Log mask for data logged at log level 2 Default: 111111111111111111111111111111111111111111111111 111111111111111111111111111111111111111111111111 111111111111111111111111111111111111111111111111 11111111111111111111111111111111111111111111111| 111111111111111111111111111111111111111111111111 111111111111111111111111111111111111111111111111 111111111111111111111111111111111111111111111111 11111111111111111111111111111111111111111111111
cmp.log_3	Log mask for data logged at log level 3 Default: 111111111111111111111111111111111111111111111111 111111111111111111111111111111111111111111111111 111111111111111111111111111111111111111111111111 11111111111111111111111111111111111111111111111| 000000000000000000000000000000000000000000000000 000000000000000000000000000000000000000000000000 000000000000000000000000000000000000000000000000 00000000000000000000000000000000000000000000000
cmp.log_4	Log mask for data logged at log level 4 Default: 111111111111111111111111111111111111111111111111 111111111111111111111111111111111111111111111111 111111111111111111111111111111111111111111111111 11111111111111111111111111111111111111111111111| 000000000000000000000000000000000000000000000000 000000000000000000000000000000000000000000000000 000000000000000000000000000000000000000000000000 00000000000000000000000000000000000000000000000
cmp.log_5	Log mask for data logged at log level 5 Default: 111111111111111111111111111111111111111111111111 111111111111111111111111111111111111111111111111 111111111111111111111111111111111111111111111111 11111111111111111111111111111111111111111111111| 000000000000000000000000000000000000000000000000 000000000000000000000000000000000000000000000000 000000000000000000000000000000000000000000000000 00000000000000000000000000000000000000000000000

Parameter	Description
cmp.guaranteed_logs_to_file	Specify if logs that are guaranteed to be sent upstream should be logged to a temp file Possible values: `FALSE, TRUE` Default: `TRUE`
cmp.unsent_log_file	Specify the name of the temp log file to log to if `cmp.guaranteed_logs_to_file` Default (Linux/Solaris): `/usr/local/phoneweb/logs/guaranteed.log.pwproxy` Default (Windows): `C:\VoiceGenie\mp\logs\guaranteed.log.pwproxy`
cmp.UTC.#	UTC or Local Time Logging Possible values: `TRUE, FALSE` Default: `FALSE`

Chapter

7 Metrics/Logging Entries

Label	Description
appl_begin Logged by: VXMLI/NGI Level: 2	**Application Begin** VXMLI: Logged when the VoiceXML application begins. NGI: Logged before starting the execution of the first page. The format is: appl_begin [<name>=<value>[|<name>=<value>...]] • <name>: The name of the parameter. This name is either the name of a Call Manager variable that is sent to the Interpreter when a session is created, INIT_URL (The initial URL), or DEFAULTS (the default page that is used). • <value>: The value of the parameter. Example: appl_begin INIT_URL=http://www.voicegenie.com/test.vxml|DEFAULTS=defaults.vxml|ANI=1234|DNIS=1234
appl_end Logged by: VXMLI/NGI Level: 2	**Application End** VXMLI: Logged when the VoiceXML application comes to an end and the session terminates. NGI: Logged after all pages have completed execution. Example: appl_end

Label	Description

mrcp_trace
Logged by: CMGR
Level: all
	ASR Trace
	The RTSP/MRCP session ID and the MRCP server information
	The format in MRCP v1 is:
	`mrcp_trace RTSPSESSION: <session id> IP: <MRCP server IP> Vendor: <vendor name>`
	`in MRCPv2 is:`
	`mrcp_trace ChannelID: <session id> IP: <MRCP server IP> Vendor: <vendor name>`
	• `<session id>:` It is an string representing a MRCP session
	• `<MRCP server IP>:` This is the MRCP server IP address and port number information.
	• `<vendor name>:` The MRCP server vendor.
	Note: This metrics entry is also logged by the VXMLI and SRM. See the *SRM System Reference Guide* for the details.
	Example:
	`mrcp_trace RTSPSESSION: d75a96cd_0000054c_44e20460_0dd6_0000 IP: 205.150.90.215:554 Vendor: NUANCE`
	mrcp_trace ChannelID: 2@speechrecog IP: 10.0.0.120:6076 Vendor: SPEECHWORKS

Label	Description
asr_trace Logged by: VXMLI/NGI Level: all	**ASR Trace** Gives information on ASR events. The format is: asr_trace <event>:<result> • <event>: This specifies what recognition event has occured: bargein: Currently playing prompts were barged-in with speech. ASR_DONE: User utterance was successfully recognized. ASR_NOMATCH_WITH_NBEST: N-best results were enabled but none of the results were of sufficient confidence or there was an ambiguous match. • <result>: Legacy Interpreter - This gives futher information depending on the event. For bargein, it's always _bargein_. For ASR_DONE and ASR_NOMATCH_WITH_NBEST, it will be the raw recognition results. NGI: The raw result that was return from the recognizer **Note:** This metrics entry is also logged by the CMGR and SRM. See the *SRM System Reference Guide* for the details. Legacy Interpreter Examples: asr_trace bargein:_bargein_ asr_trace ASR_DONE:results:+<_gram1>Vancouver|98|<Vancouver> asr_trace ASR_NOMATCH_WITH_NBEST:results:+<_gram1>cinq un|5|<cinq un> NGI Examples: asr_trace ASR_DONE:results:<?xml version='1.0'?><result><interpretation grammar="session:0x0026" confidence="90"><input mode="speech">tea</input><instance/></interpretation></result> asr_trace ASR_DONE:results:<?xml version='1.0'?><result><interpretation grammar="session:0x007b6" confidence="97"><input mode="speech">No</input><instance><SWI_meaning>{SWI_literal:No}</SWI_meaning></instance></interpretation></result>

Label	Description
bridge_begin Logged by: CMGR Level: 0	**Bridge Call Begin** This marks the beginning of a 'bridged' call (resulting from use of the transfer tag), recorded when the outbound call has been connected. The data includes dialed number, ANI, UUI, and the owner session (who initiated the call). The format is: bridge_begin <ANI>|<DNIS>|<parent>|<UU> • <ANI>: Automatic Number Identification (if so provisioned); The number from which the user is calling. • <DNIS>: Dialed Number Identification Service; The number dialed by the user. • <parent>: Call ID of the parent inbound call. • <UU>: The User-to-User information passed with the call. This is intended to be represented in IA5 format. Any non-printable data is encoded with 'percent' encoding as defined in RFC 2068. Non-printable data is presented as the hex-equivalent preceded by a percent sign. It will be set to N/A if it's not available. Example: bridge_begin 4167366493|tel:4167366779;postd=970408|00020023-0C001B58|N/A
bridge_end Logged by: CMGR Level: 0	**Bridge Call End** This record marks the end of a bridged call. The format is: bridge_end <reason> • <reason>: Disconnection reason. It can be one of: usrend: Session ended because of user hangup aplend: Session ended because of application hangup, including <transfer> maxtime expires. syserr: Session ended because of system error lmtexc: Session ended because <transfer> maxtime expires Example: bridge_end usrend

Label	Description

bridge_initiated
Logged by: CMGR
Level: 0
	Bridge Call Initiated
	This marks the initiation of a ‘bridged’ call (resulting from use of the transfer tag).
	The format is:
	`bridge_initiated <board ID>:<channel ID>`
	• `<board ID>:` PSTN board number where incoming call is placed. For VoIP, it is always zero.
	• `<channel ID>:` PSTN channel number where incoming call is placed. For VoIP, it is always zero.
	Example:
	`bridge_initiated 101:6`

Label	Description

bridge_reject
Logged by: CMGR
Level: 0
	Bridge Call Rejected
	This record is used to indicate that two-leg transfer was initiated but rejected for some reason.
	The format is:
	`bridge_reject <ANI>|<DNIS>|<parent>|<UU>|<reason>`
	• `<ANI>:` Automatic Number Identification (if so provisioned); The number from which the user is calling.
	• `<DNIS>:` Dialed Number Identification Service; The number dialed by the user.
	• `<parent>:` Call ID of the parent inbound call.
	• `<UU>:` The User-to-User information passed with the call. This is intended to be represented in IA5 format. Any non-printable data is encoded with ‘percent’ encoding as defined in RFC 2068. Non-printable data is presented as the hex-equivalent preceded by a percent sign. It will be set to `N/A` if it’s not available.
	• `<reason>:` rejection reason. It can be one of:
	`badani:` Bad ANI
	`baddest:` Call destination is invalid
	`badie:` Bad IE element
	`busy:` The called number is busy
	`error:` Some error occurred
	`fax:` The called number was to a fax machine (with call analysis)
	`glare:` Call glare (conflict with an inbound call) occurred
	`hangup:` Associated inbound call hung up
	`ineffectiveother:` SIT tone ineffectiveother detected
	`intercept:` SIT tone intercept detected
	`interrupt:` Call was interrupted by the user or platform
	`networkbusy:` The network is busy
	`noanswer:` Call was not answered or timed out
	`noautho:` No authorization for the outbound call
	`nocircuit:` SIT tone no circuit detected
	`nodialtone:` No dialtone was received (call analysis)
	`noresource:` Minimum required resources are not available
	`noringback:` No ringback error occurred (call analysis)
	`operator:` operator intercept happened (call analysis)
	`reorder:` SIT tone reorder occurred
	`unknown:` The attempt failed for an unknown reason
	`unsupported:` The network replies unsupported for the request
	`vacantcode:` SIT tone vacant code detected
	Example:
	`bridge_reject 4167361234|tel:4167366465|00020023-0C00E41E|N/A|busy`

Label	Description

call_appl
Logged by: VXMLI/NGI
Level: 1
	Application Name
	VXMLI: Logged whenever a `<meta>` tag with `name` attribute set to `application` is encountered.
	NGI: Logged before starting the execution of the first page.
	The format is:
	`call_appl <appl_name>`
	• `<appl_name>:` The arbitrary application name as specified with the `content` attribute of the `<meta>` tag.
	Example:
	`call_appl XML grammar test`

call_begin
Logged by: CMGR
Level: 0
	Call Begin
	This marks an event that an outbound call initiated with a `<call>` is connected.
	The format is:
	`call_begin <ANI>|<DNIS>|<parent>|<UU>`
	• `<ANI>:` Automatic Number Identification (if so provisioned); The number from which the user is calling.
	• `<DNIS>:` Dialed Number Identification Service; The number dialed by the user.
	• `<parent>:` Call ID of the parent inbound call.
	• `<UU>:` The User-to-User information passed with the call. This is intended to be represented in IA5 format. Any non-printable data is encoded with 'percent' encoding as defined in RFC 2068. Non-printable data is presented as the hex-equivalent preceded by a percent sign. It will be set to `N/A` if it's not available.
	Example:
	`call_begin 4167379496|tel:4167379205|00020023-0C00E50B|N/A`

call_end
Logged by: CMGR
Level: 0
	Call End
	This record marks the end of an outbound call initiated by `<call>` tag.
	The format is:
	`call_end <reason>`
	• `<reason>:` Disconnection reason. It can be one of:
	`usrend:` Session ended because of user hangup
	`aplend:` Session ended because of application hangup, including `<call>` maxtime expires.
	`syserr:` Session ended because of system error
	`lmtexc:` Session ended because
	• `<transfer>:` maxtime expires:
	Example:
	`call_end usrend`

Label	Description

call_initiated
Logged by: CMGR
Level: 0
	Call Initiated
	This marks the initiation of an outbound call by `<call>` tag.
	The format is:
	`call_initiated <board ID>:<channel ID>`
	• `<board ID>:` PSTN board number where incoming call is placed. For VoIP, it is always zero.
	• `<channel ID>:` PSTN channel number where incoming call is placed. For VoIP, it is always zero.
	Example:
	`call_initiated 101:6`

call_reference
Logged by: CMGR
Level: 0
	Call Reference
	This maps the association between VoiceGenie Call-ID and PSTN channel or SIP-Call-ID.
	The format is:
	`call_reference <Network ID>`
	• `<Network ID>:` PSTN channel ID or SIP Call-ID
	Examples:
	`call_reference XB01T23`
	`call_reference 00020023-080A1371-5060@10.0.0.211`

Label	Description
call_reject Logged by: CMGR Level: 0	**Call Rejected** This record is used to indicate that outbound call was initiated by <call> tag but rejected for some reason. The format is: call_reject <ANI>|<DNIS>|<parent>|<UU>|<reason> • <ANI>: Automatic Number Identification (if so provisioned); The number from which the user is calling. • <DNIS>: Dialed Number Identification Service; The number dialed by the user. • <parent>: Call ID of the parent inbound call. • <UU>: The User-to-User information passed with the call. This is intended to be represented in IA5 format. Any non-printable data is encoded with 'percent' encoding as defined in RFC 2068. Non-printable data is presented as the hex-equivalent preceded by a percent sign. It will be set to N/A if it's not available. • <reason>: Rejection reason. It can be one of: aplend: Session ended because of application hangup badani: Bad ANI baddest: Call destination is invalid badie: Bad IE element busy: The called number is busy error: Some error occurred fax: The called number was to a fax machine (with call analysis) glare: Call glare (conflict with an inbound call) occurred hangup: Associated inbound call hung up ineffectiveother: SIT tone ineffectiveother detected intercept: SIT tone intercept detected interrupt: Call was interrupted by the user or platform networkbusy: The network is busy noanswer: Call was not answered or timed out noautho: No authorization for the outbound call nocircuit: SIT tone no circuit detected nodialtone: No dialtone was received (call analysis) noresource: Minimum required resources are not available noringback: No ringback error occurred (call analysis) operator: Operator intercept happened (call analysis) reorder: SIT tone reorder occurred unknown: The attempt failed for an unknown reason unsupported: The network replies unsupported for the request vacantcode: SIT tone vacant code detected Example: call_reject 4167361234|tel:4167366465|00020023-0C00E3DE|N/A|busy

<table>
<tr><th>Label</th><th>Description</th></tr>
<tr><td>choice_select

Logged by:
VXMLI/NGI

Level: 3</td><td>Choice Select

When a <choice> in a <menu> is selected, the choice phrase and the choice action are logged. The choice action can be a target URL (and possibly an anchor specifying a dialog on that page) of a page that is being transitioned to or an event that is being thrown.

The format is:
choice_select [<DTMF_digits>]|[<PCDATA>]
[next=<target_URL>][event=<event>]

• <DTMF_digits>: The DTMF sequence associated with this choice.
• <PCDATA>: The PCDATA of the selected choice.
• <target_URL>: The target URL (and possibly an anchor specifying a dialog on that page) of a page that is being transitioned to. It has the following format:
[<URL>][#<dialog_ID>] Where:
<URL>: The URL of the target page.
<dialog_ID>: The ID of a <form> or <menu>.
• <event>: The event that is being thrown.

Examples:
choice_select :3|three
next=http://host.com/page.vxml#address_form

choice_select :1|one event=one_selected_event</td></tr>
<tr><td>compile_done

Logged by:
VXMLI/NGI

Level: 1</td><td>Compile Done

The compilation of a fetched VoiceXML page is complete.

The format is:
compile_done :<URL>

• <URL>: The absolute URL of the page.

Example:
compile_done :http://test.voicegenie.com/test.vxml</td></tr>
</table>

Label	Description
dtmf Logged by: VXMLI/NGI Level: 3	**DTMF Input** DTMF input was received. • <action>: This specifies how this input will be processed: input: The digits will be counted as recognition input. bargein: The digits will barge-in an on-going prompt playback. ignored: The digits will be ignored because the Interpreter is not in a state to process DTMF input. • <digit>: The sequence of DTMF digits received. The format is: VXMLI: dtmf <action>:<digits> NGI: dtmf :<digits> Examples: VXMLI: dtmf input:99 dtmf ignored:1 NGI: dtmf :99 dtmf :1
dtmf_end Logged by: VXMLI Level: 3	**DTMF Collection End** The collection of DTMF digits ends. This is not logged by NGI. The format is: dtmf_end :<reason> • <reason>: The reason DTMF collection ended. It can be one of: MATCHED: The DTMF sequence matched an active DTMF grammar. NO_MATCH: The DTMF sequence didn't match any active grammars. NO_INPUT: There was no DTMF input within the timeout interval. NO_DTMF_GRAMMAR: No active DTMF grammars within scope. GRAMMAR_ERROR: There's a problem with a DTMF grammar. INTERNAL_ERROR: DTMF collection terminated due to an internal error. ABORTED: DTMF collection aborted due to an event being thrown (e.g. hangup). Example: dtmf_end :MATCHED

Label	Description
eval_cond Logged by: VXMLI/NGI Level: 5	**Condition Evaluated** The result of the evaluation of a condition. The format is: eval_cond :{<condition>}=<result> • <condition>: A Boolean ECMAScript expression. • <result>: The evaluated result of the expression (true or false). Example: eval_cond :{reenter == true}=true
eval_expr Logged by: VXMLI/NGI Level: 5	**Expression Attribute Evaluated** An ECMAScript expression specified in an element attribute was evaluated. The format is: eval_expr <<element>>:<attr>={<expression>}=<value> • <element>: The element that has the attribute. This can be any element that has any attribute that is an expression. Examples are: FIELD, AUDIO, GRAMMAR. • <attribute>: The attribute being evaluated. • <expression>: The expression that is specified with the attribute. • <value>: The evaluation result of the expression. Example: eval_expr <SCRIPT>:expr={‘utils’ + version + ‘.js’}=utils2.js

Label	Description
eval_script Logged by: VXMLI/NGI Level: 5	**Script Executed** A <script> element was executed. The format is: eval_script <location>:[<size>][<src>] <outcome> • <location>: Where the script resides. This can be one of: inline: An inline script. extern: An external script specified by the src attribute. • <size>: The size in characters of the inline script. • <src>: The src attribute of the <script> element. Note that this does not have to be the absolute URL. • <outcome>: Whether the execution of the script was successful or not. done: The script was executed without problems. error: An error was encountered while executing the script. Examples: eval_script inline:21 done eval_script extern:domutil.js done eval_script inline:142 error
eval_var Logged by: VXMLI/NGI Level: 5	**Variable Declaration** A variable has been declared or assigned to using the <var> or <assign> element. The format is: eval_var :<variable>[={<expression>}]=<value> • <variable>: The name of the variable. • <expression>: The expression that is specified in the expr attribute. • <value>: The final value assigned to the variable. The evaluation result of the expression if an expression has been specified, or undefined if the variable has just been declared without an expr attribute. Examples: eval_var :globalvar={41 + 1}=42 eval_var :justdeclared=undefined

Label	Description

event
Logged by: VXMLI/NGI
Level: 3
	Event Thrown
	An event has been thrown.
	The format is:
	`event <event_name>:<count>[|<message>]`
	• `<event_name>:` The event that has been thrown.
	• `<count>:` The event count associated with this event.
	• `<message>:` The message associated with this event.
	Examples:
	`event connection.disconnect.hangup:1|Call hang up during FileWaiter`
	`event myevent:2`

event_handler_enter
Logged by: VXMLI/NGI
Level: 3
	Event Handler Enter
	An event was caught by an event handler.
	The format is:
	`event_handler_enter <event_name>|<location>`
	• `<event_name>:` The `event` attribute of the `<catch>` element or the tag name of a `<nomatch>`, `<noinput>`, `<error>` or `<help>` element. This may or may not be the same as the name of the actual event that is caught.
	• `<location>:` Points to the location in the document where the event handler resides. It has the following format: `[<URL>][#<dialog_ID>[.<form_item_name>]]` Where:
	`<URL>:` The URL of the page. This may include a query string.
	`<dialog_ID>:` The ID of the `<form>` or `<menu>`.
	`<form_item_name>:` The name of the form item.
	Examples:
	`event_handler_enter :NOINPUT|http://host.com/root.vxml`
	`event_handler_enter :myevent|http://host.com/page.vxml#dialog2.field3`

Label	Description
event_handler_exit Logged by: VXMLI/NGI Level: 3	**Event Handler Exit** Whenever the execution of an event handler is complete, this entry is logged. This means that the closing </catch>, </noinput>, etc. tag was reached and the event handler was not left because of a <goto>, <submit>, <throw>, <return>, <exit>, <disconnect> or an event. The format is: event_handler_exit <event_name> • <event_name>: The event attribute of the <catch> element or the tag name of a <nomatch>, <noinput>, <error> or <help> element. Example: event_handler_exit :error.connection
exec_error Logged by: VXMLI/NGI Level: 2	**Execution Error** A fatal problem was encountered during the execution of a page. Note that a blank line may follow this metric. The format is: VXMLI: exec_error (<URL>,[<application>]):<desc> • <URL>: The absolute URL of the page. • <application>: The application name specified by this page. • <desc>: Description of the problem. NGI: A free form message is logged. Example: VXMLI: exec_error (http://host.com/non-vxml.xml,):No <vxml> in VXML page NGI: exec_error TypeError: session.transfer has no properties

Label	Description
exec_warning Logged by: VXMLI Level: 3	**Execution Warning** A non-fatal problem was encountered during the execution of a page. This is not logged by the NGI. The format is: exec_warning (<URL>,[<application>]):<desc> • <URL>: The absolute URL of the page. • <application>: The application name specified by this page. • <desc>: Description of the problem. Example: exec_warning (http://host.com/clear.vxml,):Variable “x” listed in <clear>‘s attribute “namelist” is not defined

<table>
<tr><th>Label</th><th>Description</th></tr>
<tr><td>fetch_end

Logged by: VXMLI/NGI

Level: 4</td><td>Resource Fetch Response

Fetch response for a resource (audio, external grammar, external script, XML data). Note that <code>fetch_end</code> is not logged for built-in audio files.

The format is:
<code>fetch_end <outcome> ([<origin>][<failure_reason>]):<URL></code>

• <code><outcome>:</code> The outcome of the fetch. This can be one of:
<code>done:</code> Fetch success.
<code>fail:</code> Fetch failure.
• <code><origin>:</code> For a successful fetch, the origin of the response. It can be one of the following:
<code>memory:</code> The file was served from the shared memory cache of the Fetching Module.
<code>proxy-hit:</code> The file was served from the HTTP proxy (Squid).
<code>proxy-miss:</code> Means that the HTTP proxy didn’t have a fresh enough copy and the file was fetched from the application server by the HTTP proxy.
<code>direct:</code> Indicates a direct fetch from an application server when the platform is configured to bypass the HTTP proxy.
<code>file:</code> Logged for local files (<code>file://...</code>).
• <code><failure_reason>:</code> For a failed fetch, the reason of failure. This may be <code>timeout, connect timeout</code> for https timeout, or <code>http-error-xxx</code> where <code>xxx</code> specifies an HTTP status code.
• <code><URL>:</code> The absolute URL of the fetch request.

Examples:
<code>fetch_end done (proxy-miss):http://host.com/common/grammar/agent.xml</code>

<code>fetch_end done (file):file:///usr/local/samples/hello.vox</code>

<code>fetch_end fail (timeout):http://mars.com/stream.cgi</code>

<code>fetch_end fail (http-error-404):http://host.com/doesnotexist.wav</code></td></tr>
<tr><td>fetch_start

Logged by: VXMLI/NGI

Level: 4</td><td>Resource Fetch Request

Fetch request for a resource (audio, external grammar, external script, XML data). Note that <code>fetch_start</code> is not logged for built-in audio files.

The format is:
<code>fetch_start <fetch type>:<URL></code>

• <code><fetch type>:</code> The resource type of the file to fetch. It can be one of grammar, audio, script, or data.
• <code><URL>:</code> The absolute URL of the page that’s being fetched.

Example:
<code>fetch_start data:http://www.example.com/rss/newsfeed.xml</code></td></tr>
</table>

Label	Description

filled_enter
 	
Logged by: VXMLI/NGI
 	
Level: 3
	Filled Entered
	
	A `<filled>` handler has been entered.
	
	The format is:
	`filled_enter <mode>[:<form items>]`
	
	• `<mode>:` The `mode` attribute of the `<filled>`. This can be `ALL` or `ANY`.
	• `<form items>:` The list of form items that this `<filled>` has been triggered for.
	
	Example:
	`filled_enter ALL:pword`

filled_exit Logged by: VXMLI/NGI Level: 3	**Filled Exit** This entry is logged when the execution of a `<filled>` element is complete. This means that the closing `</filled>` tag was reached and the `<filled>` was not left because of a `<goto>`, `<submit>`, `<throw>`, `<return>`, `<exit>`, `<disconnect>` or an event. Example: `filled_exit`

filling
 	
Logged by: VXMLI/NGI
 	
Level: 3
	Filling Form Item
	
	Logged when a form item gets filled.
	
	The format is:
	`filling :[<Dialog Id>].<Form Item Name>:<Form Item Type>:<Value>`
	
	• `<Dialog Id>:` The id attribute of the menu or form, if specified.
	• `<Form Item Name>:` The name of the form item.
	• `<Form Item Type>:` The type (tag name) of the form item that is being filled. This can be one of `FIELD`, `TRANSFER`, `RECORD`, `SUBDIALOG` or `OBJECT`.
	• `<Value>:` The value that is used to fill the form item.
	
	Example:
	`filling :.field1:FIELD:yes`

Label	Description
form_enter Logged by: VXMLI/NGI Level: 3	**Form Entered** Logged when a <form> has been entered. The format is: form_enter [:<ID>] • <ID>: The ID of the form, if specified. Examples: form_enter form_enter :Welcome
form_exit Logged by: VXMLI/NGI Level: 3	**Form Exited** The current form has been left either because the FIA cannot find any other items to visit or there has been an internal error. The format is: form_exit <reason>[:<error>] • <reason>: How the form has been left. This can be: normal: The FIA can't find a form item to visit and the form is left for "natural reasons". internal_error: There has been an internal error. This error is not logged by NGI. • <error>: The description of the error when reason is internal_error. Example: form_exit normal
form_select Logged by: VXMLI/NGI Level: 3	**Form Item Selected** The FIA has selected a form item to visit. The format is: form_select :<item_name>:<item_type> • <item_name>: The name of the form item. This is an internally generated name if a name hasn't been specified for the item. • <item_type>: The type of the form item. This can be one of FIELD, TRANSFER, RECORD, SUBDIALOG, OBJECT, BLOCK, or INITIAL. Examples: form_select :pword:FIELD form_select :_tempBlock1:BLOCK

<table>
<tr><th>Label</th><th>Description</th></tr>
<tr><td>goto
Logged by: VXMLI/NGI
Level: 3</td><td>Goto Executed
Transition to another page, dialog or form item.
The format is:
goto :[<target_URL>][#<dialog_ID>[.<form_item_name>]]
• <target_URL>: The absolute URL of the page being transitioned to.
• <dialog_ID>: The dialog on the current page or the target page.
• <form_item_name>: The form item on the current dialog.
Examples:
goto :#exit
goto :http://diamond/next.vxml
goto :#address_form.city</td></tr>
<tr><td>incall_begin
Logged by: CMGR
Level: 0</td><td>Inbound Call Begin
This marks the beginning of an inbound call.
The format is:
incall_begin <DNIS>|<ANI>|<TRAN>|<II>|<UU>|<RDNIS>
• <DNIS>: Dialed Number Identification Service; The number dialed by the user.
• <ANI>: Automatic Number Identification (if so provisioned); The number from which the user is calling.
• <TRAN>: The transaction identifier. The format of the transaction identifier is: <yyyy><mm><dd><tttttt><seqno> Where:
<yyyy>: 4 digit year e.g. 2005
<mm>: 2 digit month. e.g. 08
<dd>: 2 digit day e.g. 20
<tttttt>: 6 digit value of time in seconds since 00:00:00 UTC, January 1, 1970, modulo 1000000
<seqno>: 3-digit number generated for each user session; incremented for each new call, modulo 1000
• <II>: The ISDN Information Digits for the call.
• <UU>: The User-to-User information passed with the call. This is intended to be represented in IA5 format. Any non-printable data is encoded with ‘percent’ encoding as defined in RFC 2068. Non-printable data is presented as the hex-equivalent preceded by a percent sign. It will be set to N/A if it’s not available.
• <RDNIS>: Redirected Dialed Number Identification Service; The number dialed by the user before being re-directed.
Example:
incall_begin
sip:2222@diamond:5060|sip:1234@pearl:5060|20050310488528015|
N/A|foo|N/A</td></tr>
</table>

Label	Description

incall_end
Logged by: CMGR
Level: 0
	Inbound Call End
	This record marks the end of an inbound session. The record specific data indicates the reason for the call end.
	The format is:
	`incall_end <reason>`
	• `<reason>:` Disconnection reason. It can be one of:
	`usrend:` Session ended because of user hangup
	`aplend:` Session ended because of application hangup
	`syserr:` Session ended because of system error
	Example:
	`incall_end usrend`

incall_initiated
Logged by: CMGR
Level: 0
	Inbound Call Initiated
	This marks the offering of an inbound call by the network.
	The format is:
	`incall_initiated <board ID>:<channel ID>`
	• `<board ID>:` PSTN board number where incoming call is placed. For VoIP, it is always zero.
	• `<channel ID>:` PSTN channel number where incoming call is placed. For VoIP, it is always zero.
	Example:
	`incall_initiated 101:23`

Label	Description

incall_reject
Logged by: CMGR
Level: 0
	Inbound Call Rejected
	This record is used to indicate that an inbound call has been presented to the platform, but has been rejected for some reason.
	The format is:
	`incall_reject <DNIS>|<ANI>|<TRAN>|<II>|<UU>|<RDNIS>|<reason>`
	• `<DNIS>:` Dialed Number Identification Service; The number dialed by the user.
	• `<ANI>:` Automatic Number Identification (if so provisioned); The number from which the user is calling.
	• `<TRAN>:` The transaction identifier. The format is the same as in `incall_begin.`
	• `<II>:` The ISDN Information Digits for the call.
	• `<UU>:` The User-to-User information passed with the call.
	• `<RDNIS>:` Redirected Dialed Number Identification Service; The number dialed by the user before being re-directed.
	• `<reason>:` Rejection reason. It can be one of:
	`badfetch:` the page could not be fetched
	`decline:` the call is declined based on the page (meta tag)
	`error:` some error occurred
	`hangup:` Associated inbound call hung up
	`noresource:` a resource, such as TTS or ASR, is not available
	`unknown:` the attempt failed for an unknown reason
	`timeout:` the incoming call is accepted, however the network fails to acknowledge or complete the connection within 10 seconds
	Example:
	`incall_reject sip:4167366779@205.150.90.154;user=phone|sip:205.150.90.78|20060621901817666|N/A|N/A|N/A|timeout`

Label	Description

input_end
Logged by: VXMLI/NGI
Level: 3
	Input End
	Recognition has ended.
	The format is:
	VXMLI:
	`input_end [:<phrase>]`
	• `<phrase>:` The interpretation of the input. If no interpretation is present, the value will be the DTMF sequence for DTMF input or the utterance for speech recognition.
	NGI:
	`input_end <reason>|<mode>|<grammar_scope>|<grammar_url>|<phrase>|<confidence>`
	• `<reason>:` can be one of the following: DISCONNECTED, FAILED, NO_INPUT, ERROR, NO_MATCH, MAX_SPEECH_TIMEOUT, ASR_MAXSPEECHTIMEOUT, RECORD_END, or TRANSFER END.
	• <mode>: input mode. Can be voice or dtmf.
	• <grammar_url>: URL of the grammar.
	• <phrase>: The interpretation of the input.
	• <confidence>: Confidence level of the input.
	Examples:
	VXMLI:
	`input_end :10678`
	`input_end :paper`
	NGI:
	`input_end MATCHED|dtmf|Field|inline|991058|1.000000`
	`input_end NO_INPUT|||||`
	`input_end MATCHED|dtmf|Field|file:///usr/local/phoneweb/samples/testapp/test.vxml|2|1.000000`

Label	Description
input_start Logged by: VXMLI/NGI Level: 3	**Input Begin** Marks the start of a recognition session. The format is: input_start [:<modes>] • <modes>: The list of input modes active for this recognition session, delimited by the	(pipe) character. These can be: DTMF: DTMF recognition. VOICE: Speech recognition. TDD: TDD/TTY input. Example: input_start :VOICE|DTMF
link_triggered Logged by: VXMLI/NGI Level: 3	**Link Triggered** Logged when a link is triggered in the VoiceXML application. The format is: link_triggered <type>=<URL or event> • <type>: The type of action taken by the link. It can be one of: event: An event will be thrown next: The link will goto a URL • <URL or event>: The URL that the link is going to or the name of the event thrown due to triggering the link. Examples: link_triggered next=#form2 link_triggered event=myevent

Label	Description

log
 	
Logged by: VXMLI/NGI
 	
Level: 0
	Log Executed
	
	`<log> element data.`
	
	The format is:
	VXMLI:
	`log [:<data>]`
	
	• `<data>:` The contents of the `<log>` element. If the <log> element logs an empty string, the string ‘log’ will be logged to the metrics file.
	
	NGI:
	`log <label>:<message><expression>`
	
	• `<label>:` The value of the label attrib ute
	• `<message>:` The contents of the log tag
	• `<expression>:` The result of evaluating the expr attribute.
	
	Example:
	VXMLI:
	`log :Hello World!`
	
	NGI:
	`log This is the label:this is the messageThis is the expression`

menu_enter
 	
Logged by: VXMLI/NGI
 	
Level: 3
	Menu Entered
	
	A `<menu>` has been entered.
	
	The format is:
	`menu_enter [<ID>]`
	
	• `<ID>:` The ID of the menu, if specified.
	
	Example:
	`menu_enter`

<table>
<tr><th>Label</th><th>Description</th></tr>
<tr><td>outcall_begin
Logged by: CMGR
Level: 0</td><td>Outbound Call Begin
This marks an event that an outbound call initiated with remote dial client is connected.
The format is:
<code>outcall_begin <ANI>|<DNIS>|<TRAN>|<UU></code>
• <code><ANI>:</code> Automatic Number Identification (if so provisioned); The number from which the user is calling.
• <code><DNIS>:</code> Dialed Number Identification Service; The number dialed by the user.
• <code><TRAN>:</code> The transaction identifier. The format of the transaction identifier is: <code><yyyy><mm><dd><tttttt><seqno></code> Where:
<code><yyyy>:</code> 4 digit year e.g. <code>2005</code>
<code><mm>:</code> 2 digit month. e.g. <code>08</code>
<code><dd>:</code> 2 digit day e.g. <code>20</code>
<code><tttttt>:</code> 6 digit value of time in seconds since 00:00:00 UTC, January 1, 1970, modulo 1000000
<code><seqno>:</code> 3-digit number generated for each user session; incremented for each new call, modulo 1000
• <code><UU>:</code> The User-to-User information passed with the call. This is intended to be represented in IA5 format. Any non-printable data is encoded with ‘percent’ encoding as defined in RFC 2068. Non-printable data is presented as the hex-equivalent preceded by a percent sign. It will be set to <code>N/A</code> if it’s not available.
Example:
<code>outcall_begin</code>
<code>sip:VoiceGenie@pearl|sip:dialog.vxml.http%3A//host/helloworl</code>
<code>d.vxml@diamond|20050316986819816|N/A</code></td></tr>
<tr><td>outcall_end
Logged by: CMGR
Level: 0</td><td>Outbound Call End
This record marks the end of an outbound call initiated by remote dial client.
The format is:
<code>outcall_end <reason></code>
• <code><reason>:</code> Disconnection reason. It can be one of:
<code>usrend:</code> Session ended because of user hangup
<code>aplend:</code> Session ended because of application hangup
<code>syserr:</code> Session ended because of system error
<code>lmtexc:</code> Session ended because <code><transfer></code> maxtime expires
Example:
<code>outcall_end usrend</code></td></tr>
</table>

Label	Description
outcall_initiated Logged by: CMGR Level: 0	**Outbound Call Initiated** This marks the initiation of an outbound call by remote dial client. In particular, a channel is selected and the outbound calling request is sent to the network. The format is: outcall_initiated <board ID>:<channel ID> • <board ID>: PSTN board number where incoming call is placed. For VoIP, it is always zero. • <channel ID>: PSTN channel number where incoming call is placed. For VoIP, it is always zero. Example: outcall_initiated 101:6
outcall_reject Logged by: CMGR Level: 0	**Outbound Call Rejected** This record is used to indicate that outbound call was initiated by remote dial client but rejected for some reason. The format is: outcall_reject <DNIS>|<ANI>|<TRAN>|<II>|<UU>|<RDNIS>|<reason> • <DNIS>: Dialed Number Identification Service; The number dialed by the user. • <ANI>: Automatic Number Identification (if so provisioned); The number from which the user is calling. • <TRAN>: The transaction identifier. The format is the same as in incall_begin. • <II>: The ISDN Information Digits for the call. • <UU>: The User-to-User information passed with the call. • <RDNIS>: Redirected Dialed Number Identification Service; The number dialed by the user before being re-directed. • <reason>: Rejection reason. It can be one of: badani: Bad ANI baddest: Call destination is invalid badfetch: the page could not be fetched badie: Bad IE element busy: The called number is busy decline: the call is declined based on the page (meta tag) error: Some error occurred fax: The called number was to a fax machine (with call analysis) glare: Call glare (conflict with an inbound call) occurred hangup: Associated inbound call hung up ineffectiveother: SIT tone ineffectiveother detected intercept: SIT tone intercept detected interrupt: Call was interrupted by the user or platform

Label	Description
	`machine:` Called number was to an answering machine (call analysis) `networkbusy:` The network is busy `noanswer:` Call was not answered or timed out `noautho:` No authorization for the outbound call `nocircuit:` SIT tone no circuit detected `nodialtone:` No dialtone was received (call analysis) `noresource:` Minimum required resources are not available `noringback:` No ringback error occurred (call analysis) `operator:` Operator intercept happened (call analysis) `reorder:` SIT tone reorder occurred `unknown:` The attempt failed for an unknown reason `unsupported:` The network replies unsupported for the request `vacantcode:` SIT tone vacant code detected Example: `outcall_reject sip:VoiceGenie@pearl|sip:dialog.vxml.http%3A//host/helloworld.vxml@diamond|20050316956139646|N/A|error`
outcall_requested Logged by: CMGR Level: 0	**Outbound Call Requested** This marks the initiation of an outbound call by remote dial client. The request is not sent to the network and the channel is not selected yet when this entry is logged. Example: `outcall_requested`

parse_error
Logged by: VXMLI/NGI
Level: 2
	Parse Error
	Parse error while compiling a VoiceXML page.
	The format is:
	`parse_error (<URL>,[<application>], line <line>):<desc>`
	• `<URL>:` The absolute URL of the page.
	• `<application>:` The application name specified by this page. This is always empty with NGI.
	• `<line>:` The line number at which the problem was encountered. This is always empty with NGI.
	• `<desc>:` Description of the problem.
	Example:
	VXMLI:
	`parse_error (http://10.0.0.136/cc.vxml,transfer_to_agent, line 33):Exactly one of “aai”(“uuidata”) or “aaiexpr”(“uuidataexpr”) may be specified in <transfer>`
	NGI:
	`parse_error (http://138.120.72.51/testapp/test3.vxml,, line ):Invalid child element Block in element Catch at line 15`

Label	Description
parse_warning Logged by: VXMLI/NGI Level: 3	**Parse Warning** Parse warning while compiling a VoiceXML page. The format is: parse_warning (<URL>,[<application>], line <line>):<desc> • <URL>: The absolute URL of the page. • <application>: The application name specified by this page. This is always empty with NGI. • <line>: The line number at which the problem was encountered. This is always empty with NGI. • <desc>: Description of the problem. Example: VXMLI: parse_warning (http://tester/test.vxml,, line 6):unsupported element FIELD in BLOCK NGI: parse_warning (http://10.0.0.1/property_vgaudiostop.vxml,,):Unknown property name: VGSTOP

<table>
<tr><th>Label</th><th>Description</th></tr>
<tr>
<td>prompt

Logged by:
VXMLI/NGI

Level: 3</td>
<td>Prompt Begin

Marks the begining of prompt playback. There will be a single entry even if a queue of multiple prompts is being played.

When the Next Generation Interpreter is used, this metric has no data. It simply provides an indication that one or more prompts will be played. Each individual prompt is identified from the metric prompt_play.

The format is:
VXMLI:
prompt <filename>[:<type>|<data>[;<type>|<data>...]]

<filename>: The filename for the temporary “prompts file” that is used internally.
<type>: The type of an individual prompt. This can be one of:
audio: An audio file referenced by a URL, a built-in audio or fetch audio.
tts: TTS prompt.
value_tts: The result of a <value> element being played as TTS, or a <record> form item value (the playback of a recording as an audio file).
<data>: The data associated with that prompt. This can mean different things for different prompt types: For audio, it’s the absolute URL of the audio file or the part after builtin: for built-in audio. For tts, it’s the SSML text. For value_tts, it has the following format: <expression>=<value> Where:
<expression>: The expression of a <value> element or the form item name of a <record> element.
<value>: The evaluated value of the expression or the full local path, prefixed by a magic string, of the audio file that contains the recording.

VXMLI:
prompt

Example:
VXMLI:
prompt /usr/local/phoneweb/tmp/00020022-101E7410-0001/...
...:audio|default_audio/error.vox;audio|default_audio/goodbye.vox;

NGI:
prompt</td>
</tr>
</table>

Label	Description
prompt_end Logged by: VXMLI/NGI Level: 2	**Prompt End** Playback of prompts has ended. The format is: prompt_end <reason>[:<input>] • <reason>: Specifies whether the prompt playback was completed or interrupted. It can be one of: done: The prompts were played to completion. hangup: The prompt playback was aborted due to hang up. This is not logged with NGI. error: The prompt playback was aborted due to an error. dtmfbargein: The playback of prompts was interrupted by DTMF barge-in. asrbargein: The playback of prompts was interrupted by speech barge-in. • <input>: The input that lead to the barge-in. This could either be a phrase in case of speech input (asrbargein result) or a sequence of DTMF digits in case of DTMF input (dtmfbargein result). This is always empty with NGI. Examples: VXMLI: prompt_end done prompt_end dtmfbargein:9 prompt_end asrbargein:stop NGI: prompt_end done prompt_end dtmfbargein prompt_end aborted

<table>
<tr><th>Label</th><th>Description</th></tr>
<tr><td>prompt_play

Logged by: NGI

Level: 2</td><td>Prompt Play

This metric has the same data as the prompt metric of the VXMLI, but for only one item of a prompt queue. Also, filename is no longer provided.

The format is:
prompt_play <type>|<data>

• <type>: The type of an individual prompt. This can be one of:
audio: An audio file referenced by a URL, a built-in audio or fetch audio.
tts: TTS prompt.
value_tts: The result of a <value> element being played as TTS, or a <record> form item value (the playback of a recording as an audio file).
• <data>: The data associated with that prompt. This can mean different things for different prompt types: For audio, it’s the absolute URL of the audio file or the part after builtin: for built-in audio. For tts, it’s the SSML text. For value_tts, it has the following format: <expression>=<value> Where:
<expression>: The expression of a <value> element or the form item name of a <record> element.
<value>: The evaluated value of the expression or the full local path, prefixed by a magic string, of the audio file that contains the recording.

Examples:

prompt_play audio|http://127.0.0.1/audio/goodbye.vox

prompt_play tts|<?xml version="1.0" encoding="UTF-8"?><speak version="1.0" xmlns="http://www.w3.org/2001/10/synthesis" xml:lang="en-US">Please press a number between 1 and 5.</speak></td></tr>
<tr><td>record_end

Logged by: VXMLI/NGI

Level: 3</td><td>Record End

A recording has finished or has been aborted.

The format is:
record_end :<outcome>[|<data>]

• <outcome>: The outcome of the record operation. Can be one of:
RECORD SUCCESS: This indicates that there was audio input before record termination and the <record> element is being filled.
NOINPUT: This indicates that there was no audio input within the timeout period.
local grammar match: dtmfterm was set to false and the recording was terminated by a local grammar match with the length of the recording being less than the mintime attribute. This is not logged with NGI, RECORD SUCCESS is logged instead.
global grammar match: dtmfterm was set to false and the recording was terminated by a global grammar match.
MINTIME: This indicates that the recording was terminated and the length of the recording was less than the mintime attribute.
INTERNAL_ERROR: This indicates that when processing the end of a recording</td></tr>
</table>

Label	Description

	an internal error occured.
	• `<data>:` This field provides additional data. The data presented depends on the outcome of the record operation. If the outcome of the record is `RECORD SUCCESS`, data is of the form `|<term reason>|<duration>|<audio_format>|<filename>`.
	• `<term reason>:` This indicates the reason the recording was terminated. Possible values are:
	`DTMF:` The recording was terminated by DTMF input while `dtmfterm` was set to `true`.
	`MAXTIME:` The recording was terminated because the audio input duration exceeded `maxtime`.
	`FINALSILENCE:` This means that the audio input was terminated by a period of silence exceeding `finalsilence`.
	`HANGUP:` This means that the audio input was terminated by the user hanging up.
	`MSG_INTERRUPT:` This means that the audio input was terminated due to an asynchronous message which was handled immediately.
	• `<duration>:` This contains the duration (in seconds) of the recording.
	• `<audio_format>:` This contains the MIME type of the audio format in which the recording was made.
	• `<filename>:` This contains the local filename where the recording has been saved.
	If the outcome of the record is `NOINPUT`, there may be data present. If data is present, data is of the form `MINTIME|dtmf=<dtmf>`.
	• `<dtmf>:` This indicates which dtmf was pressed to terminate the recording or it may be the letter `n` in the case where the recording was terminated because it satisfied the `finalsilence` condition and the length of the recording was less that the `mintime` attribute.
	If the outcome of the record is `MINTIME`, data is of the form `|<reason>`. Reason can be one of:
	• `HANGUP:` This means that the audio input was terminated by the user hanging up.
	• `MSG_INTERRUPT:` This means that the audio input was terminated due to an asynchronous message which was handled immediately. This is not logged with NGI.
	Examples:
	`record_end :RECORD SUCCESS|DTMF|4|audio/vox|/usr/local/tmp/file0001.vox`
	`record_end :RECORD SUCCESS|MAXTIME|20|audio/wave|/usr/local/tmp/file0002.vox`
	`record_end :NOINPUT`
	`record_end :global grammar match`

Label	Description

record_start
 	
Logged by: VXMLI/NGI
 	
Level: 3
	Record Begin
	
	A `<record>` element has been executed and recording has started.
	
	The format is:
	`record_start :<filename>`
	
	• `<filename>:` The local filename where the recording is going to be saved.
	
	Example:
	`record_start :/usr/local/phoneweb/tmp/00020023-101E744C-0001/...`

root_appl
 	
Logged by: VXMLI/NGI
 	
Level: 3
	Root Application
	
	The VoiceXML document specifies a root document.
	
	The format is:
	`root_appl :<URL>`
	
	• `<URL>:` The root application URL as specified in the `application` attribute of the `<vxml>` element.
	
	Example:
	`root_appl :http://darkstar.com/common/root.vxml`

Label	Description
subdialog_return Logged by: VXMLI/NGI Level: 3	**Subdialog End** Returning from a VoiceXML subdialog. The format is: VXMLI: subdialog_return :<name>|<return_type> • <name>: The name of the subdialog. • <return_type>: This field provides additional data. The data presented depends on what is returned from the subdialog. If the subdialog returns a namelist, return_type is of the form namelist|[<param_name>;<param_value>;[<param_name>;<param_value>;...]] • <param_name>: A parameter being returned from the subdialog • <param_value>: The value of the parameter. If the subdialog returns an event, return_type is of the form event|<event_name> • <event_name>: The name of the returned event. NGI: For event, the format is: subdialog_return event|<event_name> • <event_name>: The name of the returned event. For namelist, the format is: subdialog_return namelist|< namelist> • <namelist>: The evaluated namelist. Example: VXMLI: subdialog_return :GetPIN|namelist|status;OK;realname;Joe;

Label	Description

subdialog_start
 	
Logged by: VXMLI/NGI
 	
Level: 3
	Subdialog Begin
	
	Visiting a VoiceXML subdialog.
	
	The format is:
	
	VXMLI:
	`subdialog_start :<name>:<URL>[#<fragment>] |param|[<param_name>;<param_value>;[<param_name>;<param_value>;...]]`
	
	• `<name>:` The name of the subdialog.
	• `<URL>:` The URL of the subdialog.
	• `<fragment>:` The fragment (dialog ID), if specified.
	• `<param_name>:` A parameter being passed to the subdialog.
	• `<param_value>:` The value of the parameter.
	
	NGI:
	`subdialog_start :<name>:<URL>[#<fragment>] |param|[<param_expr>]`
	
	• `<name>:` The name of the subdialog.
	• `<URL>:` The URL of the subdialog.
	• `<fragment>:` The fragment (dialog ID), if specified.
	• `<param_expr>:` Parameter names and values passed to the subdialog., encoded in JavaScript Object Notation. (see `http://json.org/`).
	
	Example:
	
	VXMLI:
	`subdialog_start :GetPIN:http://diamond/collectpin.vxml |param|user;joe;`
	
	NGI:
	`subdialog_start :SDVar3:http://test/scripts/subdialog.vxml|param|inputVar1;111;inputVar2;hello;`

<table>
<tr><th>Label</th><th>Description</th></tr>
<tr><td>submit

Logged by:
VXMLI/NGI

Level: 3</td><td>Submit

When a <submit> is executed, the URL being transitioned is logged along with submitted parameters, if any.

The format is:

VXMLI:
submit :<target>
|namelist|[<name>;<value>;[<name>;<value>...]]

• <target>: The URL, dialog and form item being transitioned to.
• <name>: The name of a variable that is specified in the namelist attribute.
• <value>: The value of the variable.

NGI:
submit :<target> |namelist|[namelist_expr]

• <target>: The URL, dialog and form item being transitioned to.
• <namelist_expr>: The namelist value encoded in JavaScript Object Notation (see http://json.org/).

Examples:

VXMLI:
submit :http://host.com/page.cgi#entry
|namelist|app;45;page;3;

submit :#dialog2 |namelist|

NGI:
submit
:http://host.com/page.cgi|namelist||{"foo":,"result":{"statu
s":"success"}}</td></tr>
</table>

Label	Description

transfer_connected
Logged by: CMGR
Level: 0
	Transfer Connected
	This marks the establishment of transfer where the transfer type with the system provides such information.
	The format is:
	`transfer_connected <ANI>|<DNIS>|<parent>|<UU>`
	• `<ANI>:` Automatic Number Identification (if so provisioned); The number from which the user is calling.
	• `<DNIS>:` Dialed Number Identification Service; The number dialed by the user.
	• `<parent>:` Call ID of the parent inbound call.
	• `<UU>:` The User-to-User information passed with the call. This is intended to be represented in IA5 format. Any non-printable data is encoded with ‘percent’ encoding as defined in RFC 2068. Non-printable data is presented as the hex-equivalent preceded by a percent sign. It will be set to `N/A` if it’s not available.
	Example:
	`transfer_connected 4167366493|tel:4167366779;postd=970408|00020023-0C001B58|N/A`

Label	Description
transfer_end Logged by: VXMLI/NGI Level: 3	**Transfer End** A transfer has ended with either success or failure. The format is: transfer_end :<outcome> • <outcome>: The outcome of the transfer. The possible values are: network_busy: Some intermediate network refused the call. invalid: ... restricted: ... fax: ... machine: ... busy: The endpoint refused the call. noanswer: There was no answer within the time specified by the connecttimeout attribute. rejected: Only possible with multiphase transfers (i.e. when consultexpr is set); this result occurs when the child script exits without setting accepttransfer (or with it set to a value other than true). unknown: The outcome of the transfer is unknown. near_end_disconnect: The call was completed and was terminated by the caller. maxtime_disconnect: The call duration exceeded the value of maxtime attribute and was terminated by the platform. far_end_disconnect: The call was completed and was terminated by the callee. far_end_machine: The remote end was deteced as an answer machine using call analysis. Example: transfer_end :noanswer
transfer_initiated Logged by: CMGR Level: 0	**Transfer Initiated** This marks the initiation of transfer. The format is: transfer_initiated <board ID>:<channel ID> • <board ID>: PSTN board number where incoming call is placed. For VoIP, it is always zero. • <channel ID>: PSTN channel number where incoming call is placed. For VoIP, it is always zero. **Notes:** For redirect type of transfer, <BOARD ID>:<CHANNEL ID> has the information of the inbound, and for transfers involving outbound call this has the information of the outbound. Example: transfer_initiated 0:0

Label	Description
transfer_result Logged by: CMGR Level: 0	**Transfer Result** This record is used to indicate that one-leg transfer has completed. The format is: `transfer_result <ANI>|<DNIS>|<UU>|<method>|<reason>` • `<ANI>:` Automatic Number Identification (if so provisioned); The number from which the user is calling. • `<DNIS>:` Dialed Number Identification Service; The number dialed by the user. • `<UU>:` The User-to-User information passed with the call. This is intended to be represented in IA5 format. Any non-printable data is encoded with 'percent' encoding as defined in RFC 2068. Non-printable data is presented as the hex-equivalent preceded by a percent sign. It will be set to `N/A` if it's not available. • `<method>:` Transfer method applied to the network. It can be one of: `bridge/BRIDGE:` Bridge transfer `h450/H450:` H.450-2 blind transfer `h450join/H450JOIN:` H.450-2 consultative transfer `hkf/HKF:` Hook flash `inband/INBAND:` Inband DTMF transfer `mediaredirect/MEDIAREDIRECT:` Media redirect transfer `refer/REFER:` SIP refer transfer `referjoin/REFERJOIN:` SIP refer transfer with replace header • `<reason>:` Transfer result. It can be one of: `baddest:` Call destination is invalid `badie:` Bad IE element `busy:` The called number is busy `done:` Call is successfully transferred `error:` Some error occurred `fax:` The called number was to a fax machine (with call analysis) `hangup:` Associated inbound call hung up `ineffectiveother:` SIT tone ineffectiveother detected `intercept:` SIT tone intercept detected `interrupt:` Call was interrupted by the user or platform `invalidtrigger:` Invalid trigger for the current call state `lmtexc:` Outbound call has reached maximum call time `machine:` Called number was to an answering machine (call analysis) `maxredirects:` Maximum redirects reached `missingie:` Missing mandatory IE in ISDN transfer message `networkbusy:` The network is busy `noanswer:` Call was not answered or timed out `noautho:` No authorization for the outbound call `nocircuit:` SIT tone no circuit detected `nodialtone:` No dialtone was received `noringback:` No ringback error occurred `operator:` Operator intercept happened (call analysis) `reorder:` SIT tone reorder occurred

Label	Description

	`resourcelimit:` Minimum required resources are not available
	`timeout:` (For Q.Sig Transfer only) Call was disconnected when network fails to respond within the configurated time limit, defined by the parameter `dlgc.qsigfailontimeout`
	`unknown:` The attempt failed for an unknown reason
	`unsupported:` The network replies unsupported for the request
	`vacantcode:` SIT tone vacant code detected
	Example:
	`transfer_result sip:Username@205.150.90.136:5060|sip:XXXX@galahad|N/A|REFER|done`

transfer_start
Logged by: VXMLI/NGI
Level: 3
	Transfer Start
	A transfer element has been executed.
	The format is:
	`transfer_start :<dest>`
	• `<dest>:` The transfer destination.
	Example:
	`transfer_start :3333@wizard`

Label	Description

wf_arrived
 	
Logged by: VXMLI/NGI
 	
Level: 3
	Page Fetch Response
	
	Fetch response for a VoiceXML page
	
	The format is:
	`wf_arrived <outcome> ([<origin>][<failure_reason>]):<URL>`
	
	• `<outcome>:` The outcome of the fetch. This can be one of:
	`s:` Fetch success.
	`f:` Fetch failure.
	• `<origin>:` For a successful fetch, the origin of the response. It can be one of the following:
	`memory:` The file was served from the shared memory cache of the Fetching Module.
	`proxy-hit:` The file was served from the HTTP proxy (Squid).
	`proxy-miss:` Means that the HTTP proxy didn't have a fresh enough copy and the file was fetched from the application server by the HTTP proxy.
	`direct:` Indicates a direct fetch from an application server when the platform is configured to bypass the HTTP proxy.
	`file:` Logged for local files (`file://...`).
	• `<failure_reason>:` For a failed fetch, the reason of failure. This may be `timeout, connect timeout` for https timeout, or `http-error-xxx` where `xxx` specifies an HTTP status code.
	• `<URL>:` The absolute URL of the fetch request.
	
	Examples:
	`wf_arrived s (proxy-miss):http://host.com/test.vxml`
	
	`wf_arrived s (file):file:///usr/local/samples/hello.vxml`
	
	`wf_arrived f (timeout):http://mars.com/slow.cgi`
	
	`wf_arrived f (http-error-404):http://host.com/doesnotexist.vxml`

wf_lookup
 	
Logged by: VXMLI/NGI
 	
Level: 3
	Page Fetch Request
	
	Fetch request for a VoiceXML page
	
	The format is:
	`wf_lookup <URL>`
	
	• `<URL>:` The absolute URL of the page that's being fetched.
	
	Example:
	`wf_lookup http://grass.voicegenie.com/test.vxml`

Note: Metrics entries without any additional fields will still have an extra space character following the metrics label. For instance, the `input_end` metrics entry will be logged as `input_end .` This should be taken into account when programmatically parsing metrics data.

Chapter

8 Health via SNMP

Using SNMP Get, a number of health parameters about the VoiceGenie software are retrievable. This section outlines what health information can be retrieved for the Media Platform Call Manager component. Please refer to the *VoiceGenie 7.2 OA&M – SNMP Guide* for a description of how to perform a SNMP Get.

Note: For all call manager traps, the current count reflects the number of logical call/session objects currently existing in the system. For efficiency, disconnected call/session objects are purged periodically. Hence, even if a call/session is disconnected and the channel is freed for the next call/session, the call/session object will not be destroyed until the next purge and this may cause slight inaccuracy to the current call/session count.

Name	OID	Type	Description
STARTED	.1.3.6.1.4.1.7469.3.9.10.1.1.1	Scalar	Call Manager Start Time
CURRENTSESSION	.1.3.6.1.4.1.7469.3.9.10.1.2.1	Scalar	Number of current CMAPI sessions
PEAKSESSION	.1.3.6.1.4.1.7469.3.9.10.1.3.1	Scalar	Max number of concurrent sessions since the start
TOTALSESSION	.1.3.6.1.4.1.7469.3.9.10.1.4.1	Scalar	Total number of sessions since the start
VXMLIBIND	.1.3.6.1.4.1.7469.3.9.10.1.5.1	Scalar	Total number of times VoiceXML interpreter established connection with Call Manager
SIPPORT	.1.3.6.1.4.1.7469.3.9.10.1.6.1	Scalar	Local SIP port

Name	OID	Type	Description
SIPCURRENTIN	.1.3.6.1.4.1.7469.3.9.10.1.7.1	Scalar	Number of current inbound SIP calls
SIPCURRENTOUT	.1.3.6.1.4.1.7469.3.9.10.1.8.1	Scalar	Number of current outbound SIP calls
SIPPEAKIN	.1.3.6.1.4.1.7469.3.9.10.1.9.1	Scalar	Max number of concurrent inbound SIP calls
SIPPEAKOUT	.1.3.6.1.4.1.7469.3.9.10.1.10.1	Scalar	Max number of concurrent outbound SIP calls
SIPININIT	.1.3.6.1.4.1.7469.3.9.10.1.11.1	Scalar	Total number of inbound SIP calls initiated since the start
SIPOUTINIT	.1.3.6.1.4.1.7469.3.9.10.1.12.1	Scalar	Total number of outbound SIP calls initiated since the start
H323PORT	.1.3.6.1.4.1.7469.3.9.10.1.13.1	Scalar	Local H.323 port
H323CURRENTIN	.1.3.6.1.4.1.7469.3.9.10.1.14.1	Scalar	Number of current inbound H.323 calls
H323CURRENTOUT	.1.3.6.1.4.1.7469.3.9.10.1.15.1	Scalar	Number of current outbound H.323 calls
H323PEAKIN	.1.3.6.1.4.1.7469.3.9.10.1.16.1	Scalar	Max number of concurrent inbound H.323 calls
H323PEAKOUT	.1.3.6.1.4.1.7469.3.9.10.1.17.1	Scalar	Max number of concurrent outbound H.323 calls
H323ININIT	.1.3.6.1.4.1.7469.3.9.10.1.18.1	Scalar	Total number of inbound H.323 calls initiated since the start
H323OUTINIT	.1.3.6.1.4.1.7469.3.9.10.1.19.1	Scalar	Total number of outbound H.323 calls initiated since the start
VXMLIENABLED	.1.3.6.1.4.1.7469.3.9.10.1.60.1	Scalar	Total number of times VoiceXML interpreter established connection with Call Manager
VRMCLIENTLIST	.1.3.6.1.4.1.7469.3.9.10.1.61.1	Scalar	Available VRM Engine list
SIPREGISTRARSTATUS	.1.3.6.1.4.1.7469.3.9.10.1.62.1	Scalar	Registration status of SIP Proxies
PORTCOUNTAVAILABLE	.1.3.6.1.4.1.7469.3.9.10.1.63.1	Scalar	Total number of avaiable ports in the cluster

Name	OID	Type	Description
PORTCOUNTMIN RESERVED	.1.3.6.1.4.1.7469.3.9.10.1.64.1	Scalar	Total number of minimum reserved ports in the cluster
CMGRSTATUS	.1.3.6.1.4.1.7469.3.9.10.1.65.1	Scalar	Operating status of service
SWSHOREVXMLDPORT	.1.3.6.1.4.1.7469.3.9.10.1.66.1	Scalar	Local VXMLD por
SWSHOREUADPORT”	.1.3.6.1.4.1.7469.3.9.10.1.67.1	Scalar	UAD port
SWSHOREMSP	.1.3.6.1.4.1.7469.3.9.10.1.68.1	Scalar	MSP address
SWSHORECURRENTIN	.1.3.6.1.4.1.7469.3.9.10.1.69.1	Scalar	Number of current inbound SnowShore calls
SWSHORECURRENTOUT	.1.3.6.1.4.1.7469.3.9.10.1.70.1	Scalar	Number of current outbound SnowShore calls
SWSHOREPEAKIN	.1.3.6.1.4.1.7469.3.9.10.1.71.1	Scalar	Max number of concurrent inbound SnowShore calls
SWSHOREPEAKOUT	.1.3.6.1.4.1.7469.3.9.10.1.72.1	Scalar	Max number of concurrent outbound SnowShore calls
SWSHOREININIT	.1.3.6.1.4.1.7469.3.9.10.1.73.1	Scalar	Total number of inbound SnowShore calls initiated since the start
SWSHOREOUTINIT	.1.3.6.1.4.1.7469.3.9.10.1.74.1	Scalar	Total number of outbound SnowShore calls initiated since the start

Chapter

9 Call Manager Alarms

Alarm#	Level	Definition and Possible Message/Info	Impacts	Potential Causes	Detailed Recommended Actions
011007D1	EROR	Cannot Open file <filename>	Cannot store a recording. This applies to both `<record>` recording and full call recording using `<log>`	No permission to write the file; insufficient disk space; any other I/O system failure	Check file/directory existence/permission; Check hardware and operational state of the server including disk space and I/O system; Report to VoiceGenie [with logs]
011007D2	EROR	Unsupported audio format g726 ADPCM 2-bit	The audio file cannot be played	Attempting to play a g726 ADPCM 2-bit audio file, which is not supported	Update application to use different prompt audio.
011007D3	EROR	Cannot write header into <filename>	Cannot store a recording. This applies to both `<record>` recording and full call recording using `<log>`	No permission to write the file; insufficient disk space; any other I/O system failure	Check file/directory existence/permission; Check hardware and operational state of the server including disk space and I/O system; Report to VoiceGenie [with logs]

Alarm#	Level	Definition and Possible Message/Info	Impacts	Potential Causes	Detailed Recommended Actions
011007D4	EROR	I/O error when writing file	Cannot store a recording. This applies to both `<record>` recording and full call recording using `<log>`	No permission to write the file; insufficient disk space; any other I/O system failure	Check file/directory existence/permission; Check hardware and operational state of the server including disk space and I/O system; Report to VoiceGenie [with logs]
011007D5	EROR	URLFetch for <filename> is NOT open	Cannot start or continue playing the audio file	Lost the handle to the file; any other I/O system failure	Check file/directory existence/permission; Check hardware and operational state of the server including disk space and I/O system; Report to VoiceGenie [with logs]
011007D6	EROR	strMediaLocation (<URL>) requested from VXMLI is in wrong format	Cannot play the audio file	Mismatched vxml interpreter and call manager software version	Check and correct configuration; Report to VoiceGenie [with logs]
01100BB9	WARN	Cannot remove file <record file> on abort	The aborted recording file may remain on the system	No permission to remove the file; any other I/O system failure	Check file/directory permission; Check hardware and operational state of the server including I/O system; Report to VoiceGenie [with logs]
01100BBA	WARN	Cannot create new URL fetch	Cannot play the audio file	Lost the handle to the file; any other I/O system failure	Check file/directory existence/permission; Check hardware and operational state of the server including disk space and I/O system; Report to VoiceGenie [with logs]

Alarm#	Level	Definition and Possible Message/Info	Impacts	Potential Causes	Detailed Recommended Actions
01100BBB	WARN	Cannot access fetch handle <fetch handle>	Cannot play the audio file	Lost the handle to the file; any other I/O system failure	Check file/directory existence/permission; Check hardware and operational state of the server including disk space and I/O system; Report to VoiceGenie [with logs]
01100BBC	WARN	Cannot retrieve input stream from URLFetch	Cannot play the audio file	Lost the handle to the file; any other I/O system failure	Check file/directory existence/permission; Check hardware and operational state of the server including disk space and I/O system; Report to VoiceGenie [with logs]
01100BBD	WARN	Cannot connect URLFetch	Cannot play the audio file	Lost the handle to the file; any other I/O system failure	Check file/directory existence/permission; Check hardware and operational state of the server including disk space and I/O system; Report to VoiceGenie [with logs]
01100BBE	WARN	Cannot open FetchInput	Cannot play the audio file	Lost the handle to the file; any other I/O system failure	Check file/directory existence/permission; Check hardware and operational state of the server including disk space and I/O system; Report to VoiceGenie [with logs]

Alarm#	Level	Definition and Possible Message/Info	Impacts	Potential Causes	Detailed Recommended Actions
01100BBF	WARN	Offset is greater than or equal to file size (<offset> >= <file size>)	No audio is played for this prompt audio file	Wrong offset specified in the VXML application	Check the size of the audio file and the specified offset expression in the application, and make application adjustments if needed.
01100BC0	WARN	URLFetch read/seek had an error	The seeking operation (e.g., audio control) is not processed successfully	Trying to seek further than fetched data when streaming audio is used. Trying to seek backward from beginning of an audio file.	Check the user experience of the application and make application adjustments if needed.
01100BC1	WARN	Play start offset is invalid	Play the audio file from the beginning, instead of the specified offset.	Wrong offset specified in the VXML application	Check the size of the audio file and the specified offset expression in the application, and make application adjustments if needed.
01100BC2	WARN	Inputstream information not retrievable	May not be able to fulfill the audio playing or seeking requests	Lost the handle to the file; any other I/O system failure	Check file/directory existence/permission; Check hardware and operational state of the server including disk space and I/O system; Report to VoiceGenie [with logs]

Alarm#	Level	Definition and Possible Message/Info	Impacts	Potential Causes	Detailed Recommended Actions
01100BC3	WARN	Cannot Open file <filename>	Cannot play the audio file	No permission to read the file; file does not exist; any other I/O system failure	Check file/directory existence/permission; Check hardware and operational state of the server including disk space and I/O system; Report to VoiceGenie [with logs]
015003E9	CRIT	ID generator directory not accessible	The line manager cannot start	The ID generator directory (defined in `VG_IDGEN_DIR` environment variable) is not accessible.	Check the correctness of installation; check the existence/permission of the specified directory (e.g.: Linux is defaulted to `/usr/local/phoneweb/idgen`)
015003EA	CRIT	system.id file not accessible and not creatable	The line manager cannot start	The system.id file (defined in `VG_SYSTEMID_PATH` environment variable) is not accessible.	Check the correctness of installation; check the existence/permission of the specified file/directory
015003EB	CRIT	System IP Address not retrievable	The line manager cannot start	Misconfigured system	Check the correctness of installation; check the correctness of network card and IP configuration; check the content of `system.id` file and report to VoiceGenie [with logs]

Alarm#	Level	Definition and Possible Message/Info	Impacts	Potential Causes	Detailed Recommended Actions
015003EC	CRIT	0x<orig>.id file cannot be opened for update	The line manager cannot start	Misconfigured system; no permission to write the file; insufficient disk space; any other I/O system failure	Check the correctness of installation; check the existence/permission of the specified file/directory; check hardware and operational state of the server including disk space and I/O system; report to VoiceGenie [with logs]
015003ED	CRIT	Could not update seq number to 0x<orig>.id file	The line manager cannot start	Misconfigured system; no permission to write the file; insufficient disk space; any other I/O system failure	Check the correctness of installation; check the existence/permission of the specified file/directory; check hardware and operational state of the server including disk space and I/O system; report to VoiceGenie [with logs]
015003EE	CRIT	Environment variable ${VG_IDGEN_DIR} or ${VG_SYSTEMID_PATH} undefined	The line manager cannot start	Misconfigured system or incomplete installation	Check the correctness of installation; check the correct definition of the environment variables

Alarm#	Level	Definition and Possible Message/Info	Impacts	Potential Causes	Detailed Recommended Actions
015007D1	EROR	0x<orig>.id file reset due to invalid content	May not guarantee generation of unique call id	Initial system startup after a fresh install; insufficient diskspace; any other I/O system failure	Ignore for initial system startup; otherwise, check hardware and operational state of the server including disk space and I/O system; report to VoiceGenie [with logs]
015007D2	EROR	system.id file reset because content was invalid	May not guarantee generation of unique call id	Initial system startup after a fresh install; insufficient diskspace; any other I/O system failure	Ignore for initial system startup; otherwise, check hardware and operational state of the server including disk space and I/O system; report to VoiceGenie [with logs]
01500BB9	WARN	No Media session playing audio for call	If this message repeats consistently a particular call, it may indicate that the call has entered a bad state	Unexpected system error	Ignore if this is observed occasionally; Report to VoiceGenie [with logs] if the message persists for a particular call
01500BBA	WARN	No Media session playing DTMF for call	If this message repeats consistently a particular call, it may indicate that the call has entered a bad state	Unexpected system error	Ignore if this is observed occasionally; Report to VoiceGenie [with logs] if the message persists for a particular call
01500BBB	WARN	Media operaion on destroyed session for call	If this message repeats consistently a particular call, it may indicate that the call has entered a bad state	Unexpected system error	Ignore if this is observed occasionally; Report to VoiceGenie [with logs] if the message persists for a particular call

Alarm#	Level	Definition and Possible Message/Info	Impacts	Potential Causes	Detailed Recommended Actions
01500BBC	WARN	No Media session recording audio for call	If this message repeats consistently a particular call, it may indicate that the call has entered a bad state	Unexpected system error	Ignore if this is observed occasionally; Report to VoiceGenie [with logs] if the message persists for a particular call
01500BBD	WARN	No Media session media streaming for call	If this message repeats consistently a particular call, it may indicate that the call has entered a bad state	Unexpected system error	Ignore if this is observed occasionally; Report to VoiceGenie [with logs] if the message persists for a particular call
01A007D2	EROR	CallSigAddr Port retrieve error	Some of the configured gatekeepers may not work; H323 line manager may not start	Misconfigurations	Check gatekeeper related configurations; Check the correctness of network card and IP configuration; Report to VoiceGenie [with logs]
01A007D3	EROR	RasAddr Port retrieve error	Some of the configured gatekeepers may not work; H323 line manager may not start	Misconfigurations	Check gatekeeper related configurations; Check the correctness of network card and IP configuration; Report to VoiceGenie [with logs]
01A007D4	EROR	Media Error detected; terminating call	The call will be terminated	RTP connection timeout (RTP/RTCP packets are not received from remote end)	Check the client endpoint behavior; Check hardware and operational state of the server and the client device; Report to VoiceGenie [with logs]

Alarm#	Level	Definition and Possible Message/Info	Impacts	Potential Causes	Detailed Recommended Actions
01A007D5	EROR	Not enough free ports for H.225 CS/RAS	H323 line manager cannot start	Misconfiguration. `h323.h225port range` does not specify enough ports for RAS endpoints configuration	Check gatekeeper related configurations; Report to VoiceGenie [with logs]
01A00BB9	WARN	A port specified in h323.h225portrange is taken by another application	Some of the configured gatekeepers may not work; H323 line manager may not start	Misconfigurations. The ports are used by other applications on the machine	Check gatekeeper related configurations; Check the correctness of network card and IP configuration; Check other processes running on the system; Report to VoiceGenie [with logs]
01A00FA1	EROR	Master Gatekeeper IP Address not specified, platform will not try to perform registration (RRQ)	H323 line manager will not start	Misconfigurations	Check `h323.ras.inarqmode`, `h323.ras.outarqmode`, and `h323.ras.registrationinfo` in Call Manager configuration.
01A00FA2	INFO	No free session available for outbound call	The outbound call will fail	All the ports for the registration endpoints are taken up	Check the load of the system and adjust gatekeeper related configurations accordingly; Report to VoiceGenie [with logs]

Alarm#	Level	Definition and Possible Message/Info	Impacts	Potential Causes	Detailed Recommended Actions
01C003E9	CRIT	ID generator directory not accessible	Session manager cannot start	The ID generator directory (defined in `VG_IDGEN_DIR` environment variable) is not accessible.	Check the correctness of installation; check the existence/permission of the specified directory (e.g.: Linux is defaulted to `/usr/local/phoneweb/idgen`)
01C003EA	CRIT	system.id file not accessible and not creatable	Session manager cannot start	The system.id file (defined in `VG_SYSTEMID_PATH` environment variable) is not accessible.	Check the correctness of installation; check the existence/permission of the specified file/directory
01C003EB	CRIT	System IP Address not retrievable	Session manager cannot start	Misconfigured system	Check the correctness of installation; check the correctness of network card and IP configuration; check the content of `system.id` file and report to VoiceGenie [with logs]
01C003EC	CRIT	0x<orig>.id file cannot be opened for update	Session manager cannot start	Misconfigured system; no permission to write the file; insufficient disk space; any other I/O system failure	Check the correctness of installation; check the existence/permission of the specified file/directory; check hardware and operational state of the server including disk space and I/O system; report to VoiceGenie [with logs]

Alarm#	Level	Definition and Possible Message/Info	Impacts	Potential Causes	Detailed Recommended Actions
01C003ED	CRIT	Could not update seq number to 0x<orig>.id file	Session manager cannot start	Misconfigured system; no permission to write the file; insufficient disk space; any other I/O system failure	Check the correctness of installation; check the existence/permission of the specified file/directory; check hardware and operational state of the server including disk space and I/O system; report to VoiceGenie [with logs]
01C003EE	CRIT	Environment variable ${VG_IDGEN_DIR} or ${VG_SYSTEMID_PATH} undefined	Session manager cannot start	Misconfigured system or incomplete installation	Check the correctness of installation; check the correct definition of the environment variables
01C003EF	CRIT	VRM Initialization Failed (<return value>)	Session manager cannot start	Misconfigured system	Check the correctness of installation; check VRM client configuration; report to VoiceGenie [with logs]
01C007D2	EROR	Badly formatted script file – Something wrong with an audio file on line <request ID>.	The prompt (or transfer audio) will not be played.	Mismatched vxml interpreter and call manager software version; Unexpected system error	Check configuration, hardware, operational state of the server (including both vxmli and callmgr); Report to VoiceGenie [with logs]
01C007D3	EROR	Badly formatted script file – Something wrong with TTS request on line <request ID>	The prompt (or transfer audio) will not be played.	Mismatched vxml interpreter and call manager software version; Unexpected system error	Check configuration, hardware, operational state of the server (including both vxmli and callmgr); Report to VoiceGenie [with logs]

Alarm#	Level	Definition and Possible Message/Info	Impacts	Potential Causes	Detailed Recommended Actions
01C007D4	EROR	Badly formatted script file – Something wrong for streaming on line <request ID>.	The prompt (or transfer audio) will not be played.	Mismatched vxml interpreter and call manager software version; Unexpected system error	Check configuration, hardware, operational state of the server (including both vxmli and callmgr); Report to VoiceGenie [with logs]
01C007D5	EROR	No license for product [<product name>] feature [<feature name>]	Call manager will not start	There is no license for a critical feature or the license has expired.	Acquire an appropriate license from VoiceGenie.
01C007D6	EROR	License for product [<product name>] feature [<feature name>] has expired	Call manager and interpreter may not start (and if already running may not function properly).	The license has expired.	Acquire new VG license.

Alarm#	Level	Definition and Possible Message/Info	Impacts	Potential Causes	Detailed Recommended Actions
01C007D7	WARN	License will expire within [<days>] days for product [<product name>] feature [<feature name>]	Call manager and interpreter will not start (and if already running may not function properly) when the license expires	The license will expire within the specified timeframe. Note: When calculating this value, system assumes license expires at 00:00:00 (beginning of day), ignores "partial" days (number of days is rounded down to the previous day) and will not include the day the warning is received or the day the license will expire. Therefore the number of days specified in the warning may be less than the actual number of days before license expires.	Acquire new VG license.
01C007D8	EROR	Cannot obtain a license for the inbound call <call-id>	The inbound call will be rejected.	There is no license to accept this incoming call.	Contact VoiceGenie for more licenses or more servers.
01C007D9	EROR	Cannot obtain a license for the transfer-initiated outbound call from parent call <call-id>	The outbound call will be rejected.	There is no license to establish this outgoing call.	Contact VoiceGenie for more licenses or more servers.

Alarm#	Level	Definition and Possible Message/Info	Impacts	Potential Causes	Detailed Recommended Actions
01C007DA	EROR	Cannot obtain any TTS license	TTS resource will not be available.	There is no license provisioned for TTS.	Acquire an appropriate license from VoiceGenie, if TTS is required.
01C007DB	EROR	Cannot obtain any ASR license	ASR resource will not be available.	There is no license provisioned for ASR.	Acquire an appropriate license from VoiceGenie, if ASR is required.
01C007DC	EROR	Cannot obtain a license for the remote dial outbound call	The outbound call will be rejected.	There is no license to establish this outgoing call.	Contact VoiceGenie for more licenses or more servers.
01C007DD	EROR	Cannot obtain a license for the outbound call <call-id>	The outbound call will be rejected.	There is no license to establish this outgoing call.	Contact VoiceGenie for more licenses or more servers.
01C007DE	EROR	SMLMHandler is destroyed while stilled registered.	Call manager may not be shutting down properly.	There is an unexpected error during shutdown.	Report the problem to VoiceGenie
01C007DF	EROR	Health Monitor Handler is destroyed while still registered.	Communication link to health monitor will fail	Call manager initialization has failed.	Check for other alarms. Report to VoiceGenie [with logs].
01C007E0	EROR	Outbound call [<call-id>] rejected due to no resource	The outbound call will be rejected.	There is no license, channel, nor media session to establish this outgoing call.	Provision at least 50% more media sessions (through `rtp.maxsessions` for VoIP, `MAX_SESSIONS` for Brooktrout, or `MAX_DIALOGIC_SESSIONS` for Dialogic) than expected maximum load; Report to VoiceGenie [with logs] if problem persists.

Alarm#	Level	Definition and Possible Message/Info	Impacts	Potential Causes	Detailed Recommended Actions
01C007E1	EROR	TTS Manager was lost while doing synthesis.	TTS will not be played.	TTS engine failure; network connection problem	Check operational state of the server including network connection; Report to VoiceGenie [with logs]
01C007E3	EROR	Destination URI is not supported	Transfer request is rejected	The server does not support calling this destination (e.g., fax: destination)	Check application; Report to VoiceGenie [with logs]
01C007E5	EROR	Streaming module returns unexpected event <event> on StreamID <stream ID>	ASR recognition operation cannot be performed	RRU feature is used and the recorded utterance file cannot be opened or the recorded utterance format is not supported; `rtp.input_mode` is misconfigured; invalid RTP information from ASR engine	Check recorded utterance if RRU is used; Check `rtp.input_mode` and ASR expected streaming behavior (e.g., continuous vs. VAD); Report to VoiceGenie [with logs]
01C007E6	EROR	ASR Manager was lost while recognizing.	ASR recognition will not be active and recognition error may be thrown.	ASR engine failure; network connection problem	Check operational state of the server including network connection; Report to VoiceGenie [with logs]
01C007E7	EROR	Initializing CallSession with a NULL LineManager	The call will not be established.	No line manager can be selected to make the outbound call.	Check dialing rules provisioning; Report to VoiceGenie [with logs]

Alarm#	Level	Definition and Possible Message/Info	Impacts	Potential Causes	Detailed Recommended Actions
01C007E8	EROR	0x<orig>.id file reset due to invalid content	May not guarantee generation of unique call id	Initial system startup after a fresh install; insufficient diskspace; any other I/O system failure	Ignore for initial system startup; otherwise, check hardware and operational state of the server including disk space and I/O system; report to VoiceGenie [with logs]
01C007E9	EROR	system.id file reset because content was invalid	May not guarantee generation of unique call id	Initial system startup after a fresh install; insufficient diskspace; any other I/O system failure	Ignore for initial system startup; otherwise, check hardware and operational state of the server including disk space and I/O system; report to VoiceGenie [with logs]
01C007EA	EROR	ISDN cause code (<isdn cause>) is out of range	The specified ISDN cause code will not be used.	Application has specified an invalid cause code.	Check application; Report to VoiceGenie [with logs]
01C00BB9	WARN	Unexpected TTS error event [<event>]	TTS prompts may not work properly.	Unexpected problems with VRM or TTS server	Check VRM or TTS server; Report to VoiceGenie [with logs]
01C00BBA	WARN	Expired ASR/TTS response <event name> ignored	ASR recognitions or TTS prompts may not work properly.	Unexpected problems with VRM, TTS or ASR server	Check VRM, TTS or ASR server; Report to VoiceGenie [with logs]
01C00BBB	WARN	[<call-id>] Unexpected CallBilling event [<call-id>] for session type [<create reason>]	Billing and metrics entries may not be accurate.	Unexpected problems on the VoiceGenie server.	Report to VoiceGenie [with logs and metrics].

Alarm#	Level	Definition and Possible Message/Info	Impacts	Potential Causes	Detailed Recommended Actions
01C00BBD	WARN	LMInterface::MediaStreamResult failed with m_nResult <stream result>	TTS prompts may not work properly.	Invalid RTP information from TTS engine	Report to VoiceGenie [with logs]
01C00FA1	INFO	DTMF: rejected|<dtmf>	Informational logging to show that a DTMF digit is dropped	Application-defined behavior to ignore the DTMF digit at that point of the application	Report to VoiceGenie [with logs] if this is unexpected
01C00FA2	INFO	Obtained <num licenses> TTS Licenses and <num licenses> ASR Licenses	Informational logging to show number of TTS and ASR licenses	N/A	N/A
01C00FA3	INFO	No Inbound/Outbound lines in service	Informational logging when call manager is placed into a state to reject calls	Call manager is placed into suspend state	Report to VoiceGenie [with logs] if this is unexpected
01E007D1	EROR	vxmli process is not connected	Call Manager cannot send message to VoiceXML interpreter.	VoiceXML interpreter maybe down or is re-starting.	Check and correct configuration;Report to VoiceGenie [with logs]
01E00BB9	WARN	Translating unexpected DISCREASON <disc reason>	Interpreter will get general `EROR` for disconnect reason.	Call manager could not map protocol specific disconnect reason to the reason understood by VoiceXML interpreter	Check operational state of the server;Report to VoiceGenie [with logs]

Alarm#	Level	Definition and Possible Message/Info	Impacts	Potential Causes	Detailed Recommended Actions
01F003E9	CRIT	VRM DLL(<vrm client dll>) load failed	ASR/TTS resource will not be available.	Cannot load the libvrmclient.so as a DLL	Check `vrm.client.dll` configuration parameter; Check the existence of the file; Report to VoiceGenie
01F003EA	CRIT	SET_VGLOG() cannot be found in (<vrm client dll>)	ASR/TTS resource will not be available.	`libvrmclient.so` is corrupted or is not a valid VoiceGenie DLL	Check `vrm.client.dll` configuration parameter; Report to VoiceGenie
01F003EB	CRIT	MakeVRMModule() failed for (<vrm client dll>)	ASR/TTS resource will not be available.	`libvrmclient.so` is corrupted or is not a valid VoiceGenie DLL	Check `vrm.client.dll` configuration parameter; Report to VoiceGenie
01F003EC	CRIT	CreateVRMLib() failed for (<vrm client dll>)	ASR/TTS resource will not be available.	`libvrmclient.so` is corrupted or is not a valid VoiceGenie DLL	Check `vrm.client.dll` configuration parameter; Report to VoiceGenie
01F003EE	CRIT	Configurtion parameter <config parameter name> is not set properly.	Call manager will not start	The configuration parameter is not set in call manager configuration	Check proper installation of the system; Check for diskspace; Check the value for the specified configuration parameter; Report to VoiceGenie
01F003EF	CRIT	Trying to load more than <module count> callmgr.modules	Call manager will not start	Too many `callmgr.modules` are configured	Check proper installation of the system; Check the value for the specified configuration parameter; Report to VoiceGenie

Alarm#	Level	Definition and Possible Message/Info	Impacts	Potential Causes	Detailed Recommended Actions
01F003F0	CRIT	Failed to create callmgr module from <library name>	Call manager will not start	The library is corrupted or is not a valid VoiceGenie DLL	Check proper installation of the system; Report to VoiceGenie
01F003F1	CRIT	Trying to load more than <device module name> callmgr.devices	Call manager will not start	Too many `callmgr.devices` are configured	Check proper installation of the system; Check the value for the specified configuration parameter; Report to VoiceGenie
01F003F2	CRIT	Failed to create <device module name>	Call manager will not start	Misconfiguration; The corresponding library is corrupted or is not a valid VoiceGenie DLL	Check proper installation of the system; Check `callmgr.modules` and `callmgr.devices` configuration; Report to VoiceGenie
01F003F3	CRIT	Failed to initialize <device module name>	Call manager will not start	Initialization failure of a lower-level component; Misconfiguration	Check for other alarms; Check proper installation/configuration of the system; Report to VoiceGenie
01F003F4	CRIT	Trying to load more than <media transport name> callmgr.mediatransports	Call manager will not start	Too many `callmgr.mediatransports` are configured	Check proper installation of the system; Check the value for the specified configuration parameter; Report to VoiceGenie
01F003F5	CRIT	Failed to create <media transport name>	Call manager will not start	Misconfiguration; The corresponding library is corrupted or is not a valid VoiceGenie DLL	Check proper installation of the system; Check callmgr.modules and `callmgr.mediatransports` configuration; Report to VoiceGenie

Alarm#	Level	Definition and Possible Message/Info	Impacts	Potential Causes	Detailed Recommended Actions
01F003F6	CRIT	Failed to initialize <media transport name>	Call manager will not start	Initialization failure of a lower-level media transport component; Misconfiguration	Check for other alarms; Check proper installation/configuration of the system; Report to VoiceGenie
01F003F7	CRIT	Trying to load more than <module count> callmgr.linemanagers	Call manager will not start	Too many `callmgr.linemanagers` are configured	Check proper installation of the system; Check the value for the specified configuration parameter; Report to VoiceGenie
01F003F8	CRIT	Failed to create <line manager module name>	Call manager will not start	Misconfiguration; The corresponding library is corrupted or is not a valid VoiceGenie DLL	Check proper installation of the system; Check `callmgr.modules` and `callmgr.linemanagers` configuration; Report to VoiceGenie
01F003F9	CRIT	Failed to initialize <line manager module name>	Call manager will not start	Initialization failure of a lower-level line manager component; Misconfiguration	Check for other alarms; Check proper installation/configuration of the system; Report to VoiceGenie
01F003FA	CRIT	Configuration parameter sessmgr.appmodules is not set properly.	Call manager will not start	The configuration parameter is not set in call manager configuration	Check proper installation of the system; Check for diskspace; Check the value for the specified configuration parameter; Report to VoiceGenie

Alarm#	Level	Definition and Possible Message/Info	Impacts	Potential Causes	Detailed Recommended Actions
01F003FB	CRIT	Trying to load more than <module count> sessmgr.appmodules	Call manager will not start	Too many `callmgr.appmodules` are configured	Check proper installation of the system; Check the value for the specified configuration parameter; Report to VoiceGenie
01F003FC	CRIT	Configuration parameter sessmgr.modules is not set properly.	Call manager will not start	The configuration parameter is not set in call manager configuration	Check proper installation of the system; Check for diskspace; Check the value for the specified configuration parameter; Report to VoiceGenie
01F003FD	CRIT	Trying to load more than <module count> sessmgr.modules	Call manager will not start	Too many `callmgr.modules` are configured	Check proper installation of the system; Check the value for the specified configuration parameter; Report to VoiceGenie
01F003FE	CRIT	Cannot initialize license manager with license file <filename>. File open error.	Call manager will not start	Cannot open `vglicense.txt` file (under `config` directory)	Acquire an appropriate license from VoiceGenie.
01F003FF	CRIT	Cannot initialize license manager with license file <filename>. File parse error.	Call manager will not start	Cannot parse `vglicense.txt` file (under `config` directory)	Acquire an appropriate license from VoiceGenie.
01F00400	CRIT	Cannot initialize license manager with license file <filename>. MAC validation error.	Call manager will not start	Fail to validate the `vglicense.txt` file; the license file was for a different machine; the network card has been swapped	Provide correct MAC address to VoiceGenie to acquire an appropriate license.

Alarm#	Level	Definition and Possible Message/Info	Impacts	Potential Causes	Detailed Recommended Actions
01F00401	CRIT	Cannot initialize license manager with license file <filename>. General error.	Call manager will not start	Cannot initialize `vglicense.txt` file (under `config` directory)	Acquire an appropriate license from VoiceGenie.
01F00402	CRIT	Cannot create VGNetLib	Call manager will not start	Fetching module (iproxy) is not operating normally; Shared memory is corrupted	Check operational state of the iproxy; Perform `clc stop/start;` Report to VoiceGenie [with logs] if problem persists
01F00403	CRIT	Cannot initialize VGNetLib	Call manager will not start	Fetching module (iproxy) is not operating normally; Shared memory is corrupted	Check operational state of the iproxy; Perform `clc stop/start;` Report to VoiceGenie [with logs] if problem persists
01F00404	CRIT	Failed to initialize configuration object	Call manager will not start	CMP agent is not operating normally; configuration object cannot be obtained from database	Check proper installation of the system; Check operational state of the system; Report to VoiceGenie [with logs]
01F00405	CRIT	Cannot start CallManager	Call manager will not start	Session manager component fails to initialize	Check for other alarms; Report to VoiceGenie [with logs]
01F007D2	EROR	App Module Library <app module name> failed to load	This particular application module will not run	Cannot load the library as a DLL	Check `sessmgr.modules` configuration parameter; Check the existence of the file; Report to VoiceGenie

Alarm#	Level	Definition and Possible Message/Info	Impacts	Potential Causes	Detailed Recommended Actions
01F007D3	EROR	App module <app module name> failed to initialize	This particular application module instance will not run	This application module instance cannot initialize	Check other alarms; Check configuration parameters related to this application module; Report to VoiceGenie
01F007D4	EROR	Library <app module name> did not contain any valid app module	This particular application module will not run	No application module instance for this library is running	Check other alarms; Check configuration parameters related to this application module; Report to VoiceGenie
01F007D5	EROR	Library <app module name> does not define MakeAppModule ()	This particular application module will not run	The library file is corrupted or is not a valid VoiceGenie DLL	Check proper installation of the system; Report to VoiceGenie
01F007D6	EROR	VXML App module not loaded	VXML application module is not loaded; No VXML capabilities will be available	The VXML application module cannot initialize, or is not configured	Check configuration parameters related to this application module if VXML application module is not disabled; Report to VoiceGenie
020007D1	EROR	Failed to initialize CMGR CMP agent. <fail reason>.	Call manager cannot start due to call manager.s CMP agent fails to initialize.	Misconfiguration; system error (see detailed fail reason).	Check the correctness of installation; Check disk space and other system failures; Report to VoiceGenie [with logs]

Alarm#	Level	Definition and Possible Message/Info	Impacts	Potential Causes	Detailed Recommended Actions
020007D2	EROR	Provision Handling failure. <detail>	The provision entry modification request will not be accepted.	Misconfiguration; system error (see detailed fail reason).	Check for other alarms; Check correctness of DNIS-URL, dialing rules, hunt group, and partition provisioning; Check operational state of the server; Report to VoiceGenie [with logs]
02000BB9	WARN	May not fully function CMP features. <fail reason>	Some CMP features may not be available.	Misconfiguration; system error (see detailed fail reason).	Check the correctness of installation; Check disk space and other system failures; Report to VoiceGenie [with logs]
02000BBA	WARN	Provision Handler Registeration failure. Type=<provision type>	No real-time web provisioning updates will be available for this provision type	Misconfiguration; system error	Check the correctness of installation; Check disk space and other system failures; Report to VoiceGenie [with logs]
02000FA1	INFO	<service status>	Informational channel status trace	N/A	N/A
02000FA2	INFO	<service status>	Informational board status trace	N/A	N/A
021007D1	EROR	Telephone Number is more than 60 chars	The outbound calling or transfer request is rejected	The telephone number destination is longer than 60 characters	Check application

Alarm#	Level	Definition and Possible Message/Info	Impacts	Potential Causes	Detailed Recommended Actions
021007D2	EROR	Telephone Number contains invalid chars	The outbound calling or transfer request is rejected	The destination number contains invalid character (e.g., fails to pass RFC2806 parsing for telephone number)	Check application; Report to VoiceGenie [with logs]
021007D3	EROR	Post Dial Number is more than 196 chars	The outbound calling or transfer request is rejected	The extension or postdial in telephone destination is longer than 196 characters	Check application
021007D4	EROR	Post Dial Number contains invalid chars	The outbound calling or transfer request is rejected	The destination number (extension or postdial portion) contains invalid character	Check application; Report to VoiceGenie [with logs]
021007D5	EROR	URI has both x and postd= (conflicting extensions)	The outbound calling or transfer request is rejected	The destination number format is invalid	Check application and make sure it uses either `x` or `postd=` to specify extension dialing (not both)
021007D6	EROR	HuntGroup <hunt group number> definition consists of an invalid Trunk <trnk>	The hunt group definition update is rejected	Invalid trunk number is specified in the hunt group	Check and correct hunt group configuration
02100BB9	WARN	HuntGroup <hunt group number> definition consists of a valid but non-existing Trunk <trunk>	Outbound calling for this hunt group may not work properly	The specified trunk is not configured to make outgoing calls.	Check and correct hunt group configuration

Alarm#	Level	Definition and Possible Message/Info	Impacts	Potential Causes	Detailed Recommended Actions
02100BBA	WARN	Transfer/Outbound request using non-existing HuntGroup <hunt group number>	Outbound calling for this call will use the default huntgroup	The hunt group required by the application is not provisioned	Check and correct hunt group configuration
02100BBB	WARN	A w (wait for dialtone) character was present in the URI string and was ignore	The wait for dialtone will be ignored during dialing	Character w (wait for dialtone) is in the telephone destination	Check application
02100BBC	WARN	A parameter attribute has exceeded the 256 character limit and was truncated	The parameter in the destination will be ignored	The telephone number destination contains attribute that is longer than 256 characters	Check application
02100BBD	WARN	A parameter value has exceeded the 256 character limi and was truncated	The parameter in the destination will be ignored	The telephone number destination contains attribute value that is longer than 256 characters	Check application
022003E9	CRIT	Failed to start working thread (<thread name>)	Some callmgr components are not starting properly	Unexpected system error	Check hardware and operational state of the server; Report to VoiceGenie [with logs]
022007D1	EROR	(<result>) Failed to register application	This application module will not start	Call manager is misconfigured	Check and correct configuration; Report to VoiceGenie [with logs and call manager configuration]

Alarm#	Level	Definition and Possible Message/Info	Impacts	Potential Causes	Detailed Recommended Actions
022007D2	EROR	(<result>) Failed to register application module <app module name> of type <app module type name>	This application module will not start	Call manager is misconfigured (e.g. two application modules are configured with the same name)	Check and correct configuration; Report to VoiceGenie [with logs and call manager configuration]
02200BB9	WARN	(<bind result>) Failed to bind application	The call will be rejected.	Cannot locate an application to handle this call.	Check and correct DNIS-URL provisioning and call manager configuration; Report to VoiceGenie [with logs and call manager configuration]
02200BBA	WARN	Event received but application module not yet initialized, event ignored	Events for the application module will not be handled.	Calls are arriving or actions are requested before platform finishes initialization.	Check hardware and operational state of the server if the alarms persist after system startup; Report to VoiceGenie [with logs]
023007D1	EROR	Unexpected vxmli event status [<fail status from vxmli>]	The call will not be accepted.	Mismatched vxml interpreter and call manager software version; Unexpected system error	Check and correct configuration; Report to VoiceGenie [with logs]
023007D2	EROR	Failed to setup inbound call	The call will not be accepted.	Unexpected system error during inbound call establishment	Check hardware, operational state of the server; Report to VoiceGenie [with logs]
023007D3	EROR	(<result>) Failed to create call	The call will not be created.	Unexpected system error during call establishment	Check hardware, operational state of the server; Report to VoiceGenie [with logs]

Alarm#	Level	Definition and Possible Message/Info	Impacts	Potential Causes	Detailed Recommended Actions
023007D4	EROR	vxmli handler not registered – (<event>) ignored	The event will not be handled.	Unexpected system error	Check hardware, operational state of the server (including both vxmli and callmgr); Report to VoiceGenie [with logs]
023007D5	EROR	Cannot parse PWMAPI msg: <message>	The message from vxmli will not be handled.	Mismatched vxml interpreter and call manager software version; Unexpected system error	Check configuration, hardware, operational state of the server (including both vxmli and callmgr); Report to VoiceGenie [with logs]
023007D6	EROR	Cannot parse call id argument(s)	The message from vxmli will not be handled.	Mismatched vxml interpreter and call manager software version; Unexpected system error	Check configuration, hardware, operational state of the server (including both vxmli and callmgr); Report to VoiceGenie [with logs]
023007D7	EROR	Badly formatted script file – Something wrong with the first 5 parameters on line <line number>.	The prompt (or transfer audio) will not be played.	Mismatched vxml interpreter and call manager software version; Unexpected system error	Check configuration, hardware, operational state of the server (including both vxmli and callmgr); Report to VoiceGenie [with logs]
023007D8	EROR	Cannot Open script file <prompt file>	The prompt (or transfer audio) will not be played.	Mismatched vxml interpreter and call manager software version; Unexpected system error	Check configuration, hardware, operational state of the server (including both vxmli and callmgr); Report to VoiceGenie [with logs]
023007DB	EROR	Unexpected route event	Media routing may not be handled correctly.	Error caused by `<join/>`, `<release/>` or `<transfer/>`	Unexpected system error.

Alarm#	Level	Definition and Possible Message/Info	Impacts	Potential Causes	Detailed Recommended Actions
02300BB9	WARN	Ignoring Audio Control with Bargein on line <line number> in <content>.	Audio control will be disabled for this set of prompts	DTMF bargein is enabled together with audio control feature	Change call manager configuration (`vxml.audio_control_bargein_enable`) to enable audio control with dtmf bargein, if required.
02300BBA	WARN	Unexpected play done status on broadcast call leg: Result(<result>), ErrCode(<error code>)	The prompt may not be played properly on the broadcast call leg	error trying to fetch/open the prompt file; Race condition with call being disconnected; Unexpected system error	Check application; Check operational state of the server; Report to VoiceGenie [with logs]
02300BBB	WARN	Parse warning in calllog <log> [<reason>]	Full call recording may not be triggered	error trying to parse the full call recording request in `<log>` tag	Check application; See VG release notes documentation on PR12056 about a known parsing issue
024007D1	EROR	Error unexpected state ... expected <expected state>, actually <actual state>	CCM (CTI Call Manager) module may not function correctly	Unexpected system error	Check operational state of the server and ICM component; Report to VoiceGenie [with logs]
024007D2	EROR	Error we should have been disabled before being deleted	CCM (CTI Call Manager) module may not function correctly	Unexpected system error	Check operational state of the server and ICM component; Report to VoiceGenie [with logs]
024007D3	EROR	Error expected RELEASE_CAUSE in this message	CCM (CTI Call Manager) module may not function correctly	Unexpected system error	Check operational state of the server and ICM component; Report to VoiceGenie [with logs]

Alarm#	Level	Definition and Possible Message/Info	Impacts	Potential Causes	Detailed Recommended Actions
024007D4	EROR	Error unexpected RELEASE_CAUSE value in this message	CCM (CTI Call Manager) module may not function correctly	Unexpected system error	Check operational state of the server and ICM component; Report to VoiceGenie [with logs]
024007D5	EROR	Error unexpected application state expected <expected state> actually <actual state>	CCM (CTI Call Manager) module may not function correctly	Unexpected system error	Check operational state of the server and ICM component; Report to VoiceGenie [with logs]
02400BB9	WARN	Error AppCreate received when we're not up	CCM (CTI Call Manager) module may not function correctly	Misconfigurations; Unexpected system error	Check configurations and operational state of the server and ICM component; Report to VoiceGenie [with logs]
02400BBA	WARN	Error unexpected values	CCM (CTI Call Manager) module may not function correctly	Misconfigurations; Unexpected system error	Check configurations and operational state of the server and ICM component; Report to VoiceGenie [with logs]
02400BBB	WARN	Error failed to register	CCM (CTI Call Manager) module may not function correctly	Misconfigurations; Unexpected system error	Check configurations and operational state of the server and ICM component; Report to VoiceGenie [with logs]
02400BBC	WARN	Error failed to bind	CCM (CTI Call Manager) module may not function correctly	Misconfigurations; Unexpected system error	Check configurations and operational state of the server and ICM component; Report to VoiceGenie [with logs]
025007D1	EROR	Failed to create call	The continuity check application module cannot create a new call	Unexpected system error	Check operational state of the server; Report to VoiceGenie [with logs]

Alarm#	Level	Definition and Possible Message/Info	Impacts	Potential Causes	Detailed Recommended Actions
025007D2	EROR	Failed to setup inbound call	The continuity check application module cannot create a new call	Unexpected system error	Check operational state of the server; Report to VoiceGenie [with logs]
02500BB9	WARN	Received event in unexpected state	The continuity check application module call may have entered a bad state	Unexpected system error	Check operational state of the server; Report to VoiceGenie [with logs]
026007D1	EROR	(<result>) Failed to register remdial module <module name> of type <module type name>	The remote dial application module cannot be started. Remdial will not work.	Call manager is misconfigured (e.g. two application modules are configured with the same name)	Check and correct configuration; Report to VoiceGenie [with logs and call manager configuration]
026007D2	EROR	Could not create the server socket at <port>	The remote dial application module cannot be started. Remdial will not work.	Misconfigured system (e.g., invalid remdial.port); Network connection problem	Check and correct configuration; Check operational state of the server; Report to VoiceGenie [with logs and call manager configuration]
026007D3	EROR	TN_FATAL error on fd <socket id>	The remote dial application module may stop working. Remote dial sessions may be dropped	Network connection problem	Check operational state of the server including network connection; Report to VoiceGenie [with logs]
02600BB9	WARN	Cannot have <current calls> max calls [> MAXCALL (500)]	`Remdial.maxcalls` parameter will not take effect	`Remdial.maxcalls` (which control number of maximum concurrent remdial calls) is greater than 500	Check and correct configuration

Alarm#	Level	Definition and Possible Message/Info	Impacts	Potential Causes	Detailed Recommended Actions
02600BBA	WARN	Maximum remdial clients reached[MAXLIMIT (<max>)]	No new remote dial client can connect to the remote dial server at the moment	There are more than `remdial.maxclientsockets` remote dial clients connecting to the remote dial server	Check and correct configuration (default is 64)
02600BBB	WARN	No active session found for socket <socket id>	Processing of one remdial call command has failed	Network connection problem or other unexpected system error	Check operational state of the server including network connection; Report to VoiceGenie [with logs]
02600BBC	WARN	Max calls (<max>) reached, fd <socket id>	New call(s) cannot be made at the moment	There are more than `remdial.maxcalls` concurrent calls on the system	Check and correct configuration; Provision more VoiceGenie remote dial servers as required; Report to VoiceGenie [with logs]
027007D1	EROR	Failed to setup inbound call	Inbound call fails to be connected.	Protocol error prevents inbound call establishment.	Check operational state of the server
027007D2	EROR	Error in SOAP Envelope creation (<request identifier>)	SOAP message cannot be sent to the Policy Server.	SOAP library encountered internal error.	Check operational state of the server
027007D3	EROR	Error in communication with Policy Server (<request identifier>)	SOAP message cannot be sent to the Policy Server.	Policy Server cannot be reached or is down.	Check operational state of the server;Check and correct configuration
027007D4	EROR	Error in SOAP Envelope retrieval (<request identifier>)	VoiceGenie cannot receive SOAP message.	Policy Server sent erroneous SOAP message to VoiceGenie.	Check operational state of the server;Check and correct configuration

Alarm#	Level	Definition and Possible Message/Info	Impacts	Potential Causes	Detailed Recommended Actions
027007D5	EROR	SOAP field value truncated from <old length> to <new length>	VoiceGenie cannot receive SOAP message in entirety.	Policy Server sent SOAP value greater than 1023.	Check operational state of the server
02700BB9	WARN	[<call-id>] Error Info: <Error info>	Policy Server sent SOAP message with EROR info field filled.	Policy Server encountered error in processing SOAP request.	Notice/observation
02700FA1	INFO	PolicyServerRequest: <ODRG>|<destination>|<request type>	VoiceGenie sent SOAP request to Policy Server.	N/A	A request is sent to the CallTree project policy server
02700FA2	INFO	PolicyServerResponse: <success|fail>|<trunk group>|<release link>|<authorization>	VoiceGenie received SOAP response from Policy Server.	N/A	Notice/observation
02700FA3	INFO	TransferType: <transfer name>	VoiceGenie is attempting transfer.	N/A	Notice/observation
02700FA4	INFO	TrunkGroupID: <trunk group>	VoiceGenie is using TrunkGroupID <trunk group> for outdial.	N/A	Notice/observation
028007D1	EROR	Received unexpected ACK	The ACK message will be ignored and will not be processed	An unexpected ACK message is received during the call flow	Check the client SIP device behavior; Check for firewall and network connections; Report to VoiceGenie [with logs and SIP message traces]

Alarm#	Level	Definition and Possible Message/Info	Impacts	Potential Causes	Detailed Recommended Actions
028007D2	EROR	Media error detected; terminating call	The call will be terminated	RTP connection timeout (RTP/RTCP packets are not received from remote end); media API calls failure for SIP clear channel configurations	Check the client SIP device behavior; Check hardware and operational state of the server and the client device; Report to VoiceGenie [with logs and SIP message traces]
028007D4	EROR	REGISTER request times out. Possibly registrar not active or misconfiguration	Call manager cannot be registered with the registrar	408 response for SIP `REGISTER` request	Check operational state of the registrar; Check configuration; Check for firewall and network connections; Report to VoiceGenie [with logs and SIP message traces]
028007D5	EROR	REGISTER request considered a bad request (invalid), rejected by registrar	Call manager cannot be registered with the registrar	400 response for SIP `REGISTER` request	Check operational state of the registrar; Check configuration; Report to VoiceGenie [with logs and SIP message traces]
028007D6	EROR	VG platform not authorized to change address of record	Call manager cannot be registered with the registrar	403 response for SIP `REGISTER` request	Check operational state of the registrar; Check configuration; Report to VoiceGenie [with logs and SIP message traces]
028007D7	EROR	Address of record not found by registrar	Call manager cannot be registered with the registrar	404 response for SIP `REGISTER` request	Check operational state of the registrar; Check configuration; Report to VoiceGenie [with logs and SIP message traces]

Alarm#	Level	Definition and Possible Message/Info	Impacts	Potential Causes	Detailed Recommended Actions
028007D8	EROR	Address of record not acceptable by registrar	Call manager cannot be registered with the registrar	`488` response for SIP `REGISTER` request; Domain in Request URI not matching the one handled by the Registrar	Check operational state of the registrar; Check configuration; Report to VoiceGenie [with logs and SIP message traces]
028007D9	EROR	Other kinds of response for REGISTER request from registrar	Call manager cannot be registered with the registrar	SIP `REGISTER` rejected by registrar	Check operational state of the registrar; Check configuration; Report to VoiceGenie [with logs and SIP message traces]
028007DB	EROR	Failed to parse SDP content due to <reason>	The SDP in the message will not be further processed and the call may fail	Cannot parse the SDP content of the `INVITE` message	Check the client SIP device behavior; Report to VoiceGenie [with logs]
028007DC	EROR	SIP Authentication Algorithm not supported by Media Platform	Call manager cannot be registered with the registrar	An unsupported SIP authentication algorithm is provisioned	Check configuration; Report to VoiceGenie [with logs and SIP message traces]
028007DD	EROR	Authentication error. Potentially mismatching user name password pairs	Call manager cannot be registered with the registrar	SIP registration authentication has failed	Check operational state of the registrar; Check configuration; Report to VoiceGenie [with logs and SIP message traces]
028007DE	EROR	Received SIP INFO message does not match with any existing calls	The SIP `INFO` message will be ignored	An unexpected SIP `INFO` message is received	Check SIP clients/servers in the environment on SIP `INFO` usages; Report to VoiceGenie [with logs and SIP message traces]

Alarm#	Level	Definition and Possible Message/Info	Impacts	Potential Causes	Detailed Recommended Actions
028007DF	EROR	Unable to set custom parameter value for header	The customized SIP header value(s) and/or parameter(s) may not be set.	Invalid values are assigned from the application	Check VXML application
02800BB9	WARN	Attempt to accept call not in INVITE state	The call will be terminated/rejecte d	Unexpected system error	Report to VoiceGenie [with logs and SIP message traces]
02800BBA	WARN	Error sending INVITE for <destination>	The outbound calling request to the destination will fail	Bad destination; network connection problem	Check hardware, configuration, operational state of the server including network connection and I/O system; Report to VoiceGenie [with logs and SIP message traces]
02800BBB	WARN	Rejecting INVITE; Error creating local RTP session	The call will be rejected	Misconfigured number of media session objects; network connection problem; unexpected system error	Check hardware, configuration [`rtp.maxsessions` in call manager], operational state of the server including network connection and I/O system; Report to VoiceGenie [with logs and SIP message traces]
02800BBC	WARN	Rejecting INVITE; Error creating local PSTN session	The call will be rejected	Misconfigured number of media session objects; media hardware problem; unexpected system error	Check hardware, configuration, operational state of the server including network connection and I/O system; Report to VoiceGenie [with logs and SIP message traces]

Alarm#	Level	Definition and Possible Message/Info	Impacts	Potential Causes	Detailed Recommended Actions
02800BBD	WARN	Received SDP that contains dynamic payload(<payload>) that is not defined in rtpmap	The payload will be ignored, but the SDP negotiation and the call will continue.	The incoming SDP may be malformed or may not have contained the appropriate rtpmap information.	Check SIP clients/servers in the environment; Report to VoiceGenie [with logs and SIP message traces]
02800BBE	WARN	Received a DTMF digit that is not supported	SIP `INFO` DTMF may not be detected properly.	An unsupported telephony event is sent via SIP `INFO`.	Check SIP clients/servers in the environment; Report to VoiceGenie [with logs and SIP message traces]
02800BBF	WARN	Received a clock rate of 0 for payload	Default clock rate will be assigned for the payload	The incomfing SDP may be malformed or may not have appropriate information.	Check SIP clients/servers in the environment; Report to VoiceGenie [with logs and SIP message traces]
02800FA1	INFO	Request/response received/sent	Informational logging when a SIP message is sent/received	N/A	N/A
02800FA2	INFO	SIP2 processing delay is <delay>	Information logging when LMSIP2 processing delay is high	The system may be overloaded	Check operational state of the server; Report to VoiceGenie [with logs] if the server is not operating correctly
029007D1	EROR	Conference failed	The conference request has failed	Race condition between caller hanging up and the application triggering the conference; unexpected system error	If this is not due to a user hangup, check operational state of the server and report to VoiceGenie [with logs]

Alarm#	Level	Definition and Possible Message/Info	Impacts	Potential Causes	Detailed Recommended Actions
029007D2	EROR	Received unexpected conference change reason	The conference request has failed	Unexpected system error	Check operational state of the server; Report to VoiceGenie [with logs]
02900FA1	INFO	Conference established	A non-VXML-initiated conference session is created	N/A	N/A
02900FA2	INFO	Conference terminated	A non-VXML-initiated conference session is terminated	N/A	N/A
02A00BB9	WARN	Soft limit exceeded	This is a warning message regarding the port count soft limit exceeding. If more calls are still routed to the paritition, hard limit can be reached and calls can be dropped/re-routed to other applications.	Too many calls are routed for this partition compared to the expected provisioning	Provision better resource allocation among the partition
02A00BBA	WARN	Hard limit exceeded	Hard port count limit is reached and calls will be dropped/re-routed to other applications based on partition configuration.	Too many calls are routed for this partition compared to the expected provisioning	Provision better resource allocation among the partition

Alarm#	Level	Definition and Possible Message/Info	Impacts	Potential Causes	Detailed Recommended Actions
02A00BBB	WARN	Minimum required limit exceeded	Calls will be dropped/re-routed to other applications based on partition configuration.	Minimum required port count limit for a provision is reached because too many calls are routed for other partitions sharing the platform cluster	Provision better resource allocation among the partition
02A00FA1	INFO	Port count info expired	Port count statistics from one machine within the cluster will be removed	That machine is being shutdown; network connection issue	Check hardware and operational state of the servers within the cluster and network connection
02F003E9	CRIT	Failed to initialize MTMPC	Call Manager may not start; VoIP configuration will not function properly.	Misconfiguration; unexpected system error	Check operational state of the server; Check and correct mpc related call manager configuration
0B0007D1	EROR	Invalid media	Call Manager unable to play media content.	Incorrectly encoded media content.	Check file/directory existence and permission and free disk space.
0B0007D2	EROR	RTSP unexpected disconnect	Call Manager unable to complete playing the RTSP prompt.	Error on the RTSP server.	Check RTSP server log and configuration. Report to VoiceGenie [with logs].
0B0007D3	EROR	RTSP request error	Fail to send RTSP request to RTSP server. Call Manager unable to complete playing the RTSP prompt.	Error on the RTSP server.	Check RTSP server log and configuration. Report to VoiceGenie [with logs].

Alarm#	Level	Definition and Possible Message/Info	Impacts	Potential Causes	Detailed Recommended Actions
0B0007D4	EROR	RTSP reply error	Fail to send RTSP reply to RTSP server. Call Manager unable to complete playing the RTSP prompt.	Error on the RTSP server.	Check RTSP server log and configuration. Report to VoiceGenie [with logs].
0B0007D5	EROR	RTSP rtp interface error	Fail to establish connection with RTSP server. Call Manager unable to complete playing the RTSP prompt.	Error on the RTSP server. Out of free TCP port on the Media Platform server.	Check RTSP server log and configuration. Check Media Platform server TCP port status Report to VoiceGenie [with logs].
0B0007D6	EROR	Video format unsupported	Call Manager is unable to play the video content	Playing an `.avi` file that contains unsupported format	Check the application and video content
0B0007D8	EROR	Number of audio channels unsupported	Call Manager is unable to play the media content	Playing a media file that contains more than one audio channels	Check the application and media content(s); Note that VoiceGenie only supports single audio channel.
0B0007D9	EROR	Bad AVI chunk size found	Call Manager is unable to play the media content	Playing an `.avi` that is corrupted	Check the application and media content(s)
0B0007DA	EROR	Malformed AVI header found	Call Manager is unable to play the media content	Playing an `.avi` that is corrupted	Check the application and media content(s)
0B0007DB	EROR	Recording buffer for iso too small	Call Manager unable to play the file stored in ISO file format.	Call Manager configuration parameter mpc.mediamgr.isofilerecordheadersize too small.	Increase the size of mpc.mediamgr.isofile recordheadersize
0B0007DC	EROR	Unable to allocate new memory	Call Manager unable to play the file stored in ISO file format	System out of memory.	Report to VoiceGenie [with logs].

Alarm#	Level	Definition and Possible Message/Info	Impacts	Potential Causes	Detailed Recommended Actions
0B0007DD	EROR	No iso media track recognized	Call Manager unable to play the file stored in ISO file format.	Malformed ISO file.	Check the application. Make sure ISO file is encoded correctly.
0B0007DE	EROR	Bad size in iso box found	Call Manager unable to play the file stored in ISO container format.	Malformed ISO file.	Check the application. Make sure ISO file is encoded correctly.
0B0007E0	EROR	Brand incompatible with 3GPP	Call Manager unable to play the 3GP file.	Incorrectly encoded 3GP file. The brand of the file is incompatible with the 3GPP version of the file.	Check the application. Re-encode the 3GP file.
0B0007E1	EROR	Bad 3GPP major brand found	Call Manager unable to play the 3GP file.	Incorrectly encoded 3GP file. The major brand of the file is incorrect.	Check the application. Re-encode the 3GP file.
0B0007E2	EROR	Error iso box value found	Call Manager unable to play the file stored in ISO container format.	Malformed ISO file.	Check the application. Make sure ISO file is encoded correctly.
0B0007E3	EROR	Unable to start recording	Call Manager unable to start recording.	Unsupported URI specified in the <record> tag.	Check the application.
0B0007E4	EROR	Bad preloading index table found	Call Manager may not be able to play the media file.	Unsuppored media file.	Check the application Report to VoiceGenie with the media file.
0B0007E5	EROR	No Media Info Object	Call Manager may not be able to play the media file.	Unsuppored media file.	Check the application

Alarm#	Level	Definition and Possible Message/Info	Impacts	Potential Causes	Detailed Recommended Actions
0B000BB9	WARN	Video recording frame discarded	Dropped video frame in recording.	Received invalid video frame.	Check video device. Report to VoiceGenie [with logs].
0B000BBA	WARN	Unexpected rtsp reply received	Potental error with the RTSP prompt.	Error on the RTSP server.	Check RTSP server configuration. Report to VoiceGenie [with logs].
0B000BBB	WARN	Bad value in iso box found	Call Manager may not be able to properly play the file stored in ISO container format.	Malformed ISO file.	Check the application. Make sure ISO file is encoded correctly.
0B000BBC	WARN	Bad type in iso box found	Call Manager may not be able to properly play the file stored in ISO container format.	Malformed ISO file.	Check the application. Make sure ISO file is encoded correctly.
0B000BBD	WARN	Mandatory iso box missing	Call Manager may not be able to properly play the file stored in ISO container format.	Malformed ISO file.	Check the application. Make sure ISO file is encoded correctly.
0B000BBE	WARN	Buffer size too small to parse iso header	Call Manager may not be able to properly play the file stored in ISO container format.	Call Manager configuration parameters mpc.mediamgr.audiobuffersize and mpc.mediamgr.videobuffersize not big enough.	Increase values of Call Manager parameters mpc.mediamgr.audiobuffersize and mpc.mediamgr.videobuffersize.
0B000BBF	WARN	Audio sampling rate unsupported	Call Manager is unable to play the media content	Playing a media file that contains unsupported audio sampling rate	Check the application and media content(s); Note that VoiceGenie only supports 8000Hz audio sampling rate

Alarm#	Level	Definition and Possible Message/Info	Impacts	Potential Causes	Detailed Recommended Actions
0B000BC0	WARN	Video sampling rate unsupported	Call Manager unable to play the video file.	Video content is not encoded in 90000Hz sampling rate.	Check the application. Encode the video in 90000Hz sampling rate.
0B1003E9	CRIT	Initializing VGMediaInfo failed	Call Manager can not start.	Misconfiguration	Check configurations; Report to VoiceGenie [with logs]
0B1007D1	EROR	Failed to parse SDP	SDP negotiation would fail.	Unsupported SDP message sent by Remote SIP device.	Check SDP related configurations; Report to VoiceGenie [with logs]
0B100BB9	WARN	Invalid MPC configuration parameter value	Call Manager may not start or may not function properly	One or more `mpc-` prefix call manager parameters are not set properly.	Check and correct mpc related call manager configuration
0B100BBA	WARN	MPCConnection::Initialize <ConnID> initialization failed	A call may be rejected/dropped.	Misconfiguration; incompatible phone/gateway; unexpected system error	Check VoIP clients/servers in the environment; Check the configuration; Report to VoiceGenie [with logs]
0B100BBB	WARN	Cannot modify MPC connection	A call may be dropped.	Misconfiguration; incompatible phone/gateway; unexpected system error	Check VoIP clients/servers in the environment; Check the configuration; Report to VoiceGenie [with logs]
0B200BB9	WARN	Unable to access the media content	Call Manager is unable to play the media content	Inaccessible or corrupted media content; unexpected system error	Check the application, media content and its availability

Alarm#	Level	Definition and Possible Message/Info	Impacts	Potential Causes	Detailed Recommended Actions
0B200BBA	WARN	Media sink buffer is full	Media data is dropped during media processing	System is overloaded; unexpected system error; a caller is streaming data at faster than real-time	Report to VoiceGenie [with logs]; make sure all clients/gateways are streaming data at real-time speed
0B200BBB	WARN	Media source buffer is full	Media data is dropped during media processing	System is overloaded; unexpected system error; a caller is streaming data at faster than real-time	Report to VoiceGenie [with logs]; make sure all clients/gateways are streaming data at real-time speed
0B200BBC	WARN	Unable to allocate packet buffer	Output media data is dropped before transmission	System is overloaded; unexpected system error; trying to play media content that contains large frames	Check the configuration (`mpc.rtp.audiobuffersize`, `mpc.rtp.videobuffersize`); Report to VoiceGenie [with logs]; downgrade the media content resolution/quality; re-encode the video content with smaller GOBs (Group-Of-Blocks)
0B200BBD	WARN	RTP packet size greater than maximum	Output media data is dropped before transmission	Trying to play media content that contains large frames	Check the configuration (`mpc.rtp.maxrtppacketsize`); Report to VoiceGenie [with logs]; downgrade the media content resolution/quality; re-encode the video content with smaller GOBs (Group-Of-Blocks)

Alarm#	Level	Definition and Possible Message/Info	Impacts	Potential Causes	Detailed Recommended Actions
0B200BBE	WARN	Media frame size larger than buffer size	Output media data is dropped before transmission	System is overloaded; unexpected system error; trying to play media content that contains large frames	Check the configuration (`mpc.rtp.audiobuffersize`, `mpc.rtp.videobuffersize`); Report to VoiceGenie [with logs]; downgrade the media content resolution/quality; re-encode the video content with smaller GOBs (Group-Of-Blocks)
0B200BBF	WARN	Bridge object not found when Network object is destroyed	System error.	Unexpected system error.	Report to VoiceGenie [with logs].
0B200BC0	WARN	H263 sorter does not have enough packets to break down the H.263 frame properly.	Cannot play H.263 video properly.	Video prompt contains extremely large H.263 frame.	Report to VoiceGenie [with logs].
0B200BC1	WARN	Record can not be opened	Cannot record media.	Invaid reord URL or file system error.	Check application and platform configuration. Report to VoiceGenie [with logs].
0B200BC2	WARN	Invaid RTP packets are dropped	Part of media stream are not recorded or forwarded	Invalid RTP packets are sent by remote devices.	Check application.
0B400BB9	WARN	Do not have license for DSP	Cannot play the prompt that requires transcoding correctly. Cannot continue the call.	Out of DSP transcoder license.	Check application and platform configuration. Report to VoiceGenie [with logs].

Alarm#	Level	Definition and Possible Message/Info	Impacts	Potential Causes	Detailed Recommended Actions
0B400BBA	WARN	Can not find the needed transcoder	Cannot transcode the media	The needed transcoder is not available.	Check platform configuration. Report to VoiceGenie [with logs].

Chapter

10 Legacy Interpreter Alarms

Alarm	Severity	Description	Impacts	Causes	Recommended Action
04500001	CRIT	Memory allocation failed	`<script>` element execution will fail with an error.	Insufficient free memory.	Restart the services or reboot the server.
03000065	CRIT	Id generator directory not accessible	Interpreter will exit.	Insufficient disk space.	Check file system integrity and free disk space.
03000066	CRIT	`system.id` file not accessible and couldn't be created	Interpreter will exit.	Invalid folder/file permissions or insufficient disk space.	Check file system integrity and free disk space.
03000067	CRIT	Couldn't retrieve system IP address	Interpreter will exit.	System issue.	Check configuration.
03000068	CRIT	Id file cannot be opened for updating	Interpreter will exit.	Invalid folder/file permissions or insufficient disk space.	Check file system integrity and free disk space.
03000069	CRIT	Could not update sequence number to id file	Interpreter will exit.	Insufficient disk space.	Check file system integrity and free disk space.

Alarm	Severity	Description	Impacts	Causes	Recommended Action
081003E9	CRIT	Assertion failed	Interpreter will exit with a core dump.	Should never happen.	Report to Genesys with logs and traces.
033007D1	EROR	Unable to reply to client	Unexpected session behaviour.	System resource issues.	Check operational state of platform.
033007D3	EROR	Bad message format	Unexpected session behaviour.	Should never happen.	Report to Genesys with logs and traces.
03300BB9	WARN	Failed to send message because message is too big	Unexpected session behaviour.	Should never happen.	Check operational state of platform. Report to Genesys with logs and traces.
0300697B, 0310697B, 0320697B, 0330697B, 0360697B, 0370697B, 0380697B, 0390697B, 03B0697B, 03C0697B, 03D0697B, 03E0697B, 03F0697B, 0400697B, 0420697B, 0440697B, 0460697B, 0470697B, 0480697B, 0490697B, 04A0697B, 04B0697B, 04C0697B, 04D0697B	CRIT	Memory allocation failed for array	Interpreter will exit.	Insufficient memory.	Restart the services or reboot the server. Reboot server and check hardware.

Alarm	Severity	Description	Impacts	Causes	Recommended Action
0300697C, 0310697C, 0320697C, 0330697C, 0360697C, 0370697C, 0380697C, 0390697C, 03B0697C, 03C0697C, 03D0697C, 03E0697C, 03F0697C, 0400697C, 0420697C, 0440697C, 0450697C, 0460697C, 0470697C, 0480697C, 0490697C, 04A0697C, 04B0697C, 04C0697C, 04D0697C	CRIT	Memory allocation failed	Interpreter will exit.	Insufficient memory.	Restart the services or reboot the server. Reboot server and check hardware.
036069A4	CRIT	Failed to allocate string buffer	Interpreter will exit.	Insufficient memory.	Restart the services or reboot the server. Reboot server and check hardware.
03D069B6	CRIT	Cannot allocate memory for lexical analysis	Interpreter will exit.	Insufficient memory.	Restart the services or reboot the server. Reboot server and check hardware.
030069BB	CRIT	Out of memory creating states	Interpreter will exit.	Insufficient memory.	Restart the services or reboot the server. Reboot server and check hardware.
03F07F45	CRIT	Failed to get built-in DTMF grammar	Interpreter will exit.	Unexpected internal software problem.	Check operational state of platform. Report to Genesys with logs and traces.

Alarm	Severity	Description	Impacts	Causes	Recommended Action
03F07F46	CRIT	Failed to get external DTMF grammar	Interpreter will exit.	Unexpected internal software problem.	Check operational state of platform. Report to Genesys with logs and traces.
03F082DE	CRIT	Failed to get built-in grammar	Interpreter will exit.	Unexpected internal software problem.	Check operational state of platform. Report to Genesys with logs and traces.
03F082DF	CRIT	Failed to get external grammar	Interpreter will exit.	Unexpected internal software problem.	Check operational state of platform. Report to Genesys with logs and traces.
030093BD	CRIT	Current page is lost after fetching root document	VoiceXML application compilation failure.	Unexpected internal software problem.	Check operational state of platform. Report to Genesys with logs and traces.
030093BE	CRIT	Root page is lost after fetching next page	Root document will not be compiled.	Unexpected internal software problem.	Check operational state of platform. Report to Genesys with logs and traces.
036186A1, 03F186A1, 040186A1	EROR	Failed to open file	Loss of some functionality, e.g. built-in grammars, utterance recording.	Invalid folder/file permissions or insufficient disk space.	Check configuration. Report to Genesys with logs and traces.
036186A2	EROR	Cannot write to file	Utterance record file is not written properly.	Invalid folder/file permissions or insufficient disk space.	Check file system integrity and free disk space.
03018705	EROR	Id file reset due to invalid content	None.	Bad installation.	Check file system integrity and free disk space. Report to Genesys with logs and traces.
0301876A	EROR	system.id file reset because content was invalid	None.	Bad installation.	Check file system integrity and free disk space. Report to Genesys with logs and traces.

Alarm	Severity	Description	Impacts	Causes	Recommended Action
030187CF	EROR	Environment variable VG_IDGEN_DIR or VG_SYSTEMID_PATH undefined	Interpreter will exit.	Configuration problem.	Check configuration.
0301B19A	EROR	Invalid command line option	If it’s an invalid instance id, the Interpreter will exit. Other invalid options are just ignored.	Configuration problem. Invalid command line option.	Check configuration.
0301B19B	WARN	Invalid format in configuration parameter	Default value will be used for the configuration parameter.	Invalid value set for the configuration parameter in voicexml.cfg	Check configuration.
0301F01A	EROR	Unable to open initial file	Interpreter will exit.	Invalid folder/file permissions or insufficient disk space.	Check configuration.

Alarm	Severity	Description	Impacts	Causes	Recommended Action
0301F01D, 0311F01D, 0321F01D, 0331F01D, 0361F01D, 0371F01D, 0381F01D, 0391F01D, 03B1F01D, 03C1F01D, 03D1F01D, 03E1F01D, 03F1F01D, 0401F01D, 0421F01D, 0441F01D, 0451F01D, 0461F01D, 0471F01D, 0481F01D, 0491F01D, 04A1F01D, 04B1F01D, 04C1F01D, 04D1F01D	EROR	Non-zero reference count when destroying object	Interpreter will exit.	Unexpected internal software problem.	Check operational state of platform. Report to Genesys with logs and traces.

Alarm	Severity	Description	Impacts	Causes	Recommended Action
0301F01E, 0311F01E, 0321F01E, 0331F01E, 0361F01E, 0371F01E, 0381F01E, 0391F01E, 03B1F01E, 03C1F01E, 03D1F01E, 03E1F01E, 03F1F01E, 0401F01E, 0421F01E, 0441F01E, 0451F01E, 0461F01E, 0471F01E, 0481F01E, 0491F01E, 04A1F01E, 04B1F01E, 04C1F01E, 04D1F01E	EROR	Index out of bounds	Interpreter will exit.	Unexpected internal software problem.	Check operational state of platform. Report to Genesys with logs and traces.
0331F025, 0371F025	EROR	Cannot find session	Unexpected session behaviour.	Unexpected internal software problem.	Check operational state of platform. Report to Genesys with logs and traces.
0331F026	EROR	Cannot switch to session	Unexpected session behaviour.	Unexpected internal software problem.	Check operational state of platform. Report to Genesys with logs and traces.
0331F02A	EROR	Cannot create call data structure	Call will be rejected.	Insufficient memory.	Restart the services or reboot the server. Reboot server and check hardware.
0331F02B	EROR	Interpreter busy	Call will be rejected.	Unexpected internal software problem.	Check operational state of platform. Report to Genesys with logs and traces.

Alarm	Severity	Description	Impacts	Causes	Recommended Action
0331F02C, 0371F02C, 0461F02C	EROR	Cannot insert session into session list	Call will be rejected or forking of a new session will fail.	Insufficient memory.	Check operational state of platform. Report to Genesys with logs and traces.
0331F02D	EROR	Invalid state	Unexpected session behaviour.	Unexpected internal software problem.	Check operational state of platform. Report to Genesys with logs and traces.
0371F042, 04E1F042	EROR	Cannot delete session instance	Session object will linger. May or may not have external impact.	Unexpected internal software problem.	Check operational state of platform. Report to Genesys with logs and traces.
0361F046	EROR	Unable to open log file	Interpreter will exit.	`fopen()` failure.	Check file system integrity and free disk space.
0401F04F	EROR	Cannot open file for writing	Interpreter will exit.	Disk/system resource issue	Check file system integrity and free disk space.
03D1F055	EROR	Action failed on file for lexical analysis	VXML application compilation failure.	`fopen()` or `fseek()` failure.	Check operational state of platform. Report to Genesys with logs and traces.
03D1F057	EROR	Couldn't read expected number of bytes while reading page	VXML application compilation failure.	`fread()` failure.	Check operational state of platform. Report to Genesys with logs and traces.

Alarm	Severity	Description	Impacts	Causes	Recommended Action
0301F059, 0311F059, 0321F059, 0331F059, 0361F059, 0371F059, 0381F059, 0391F059, 03B1F059, 03C1F059, 03D1F059, 03E1F059, 03F1F059, 0401F059, 0421F059, 0441F059, 0451F059, 0461F059, 0471F059, 0481F059, 0491F059, 04A1F059, 04B1F059, 04C1F059, 04D1F059	EROR	Failed to create entity	Interpreter will exit.	Unexpected.	Check operational state of platform. Report to Genesys with logs and traces.
0301F05D, 03E1F05D	EROR	Failed to destroy dictionary	Unexpected.	`dtclose()` failed.	Check operational state of platform. Report to Genesys with logs and traces.
04D1F0E4	EROR	`STATE_AFTER_INPUT` wasn't resolved at compile time	The vxmli session will be terminated.	Report to VG with logs.	Report to Genesys with logs and traces.
03E1F0ED, 0401F0ED, 04D1F0ED	EROR	Audio file fetch request failed	The voice will not be fetched or played.	Multiple reasons.	Check operational state of platform. Report to Genesys with logs and traces.
0401F0F1	EROR	Failed to open TTS file for writing	Event	`Fopen()` failure.	Check file system integrity and free disk space.

Alarm	Severity	Description	Impacts	Causes	Recommended Action
03F1F0FC, 04D1F0FC	EROR	Grammar file fetch request failed	Event	Multiple reasons.	Check operational state of platform. Report to Genesys with logs and traces.
03F1F0FD	EROR	Failed to open grammar file for writing	VXMLI will exit or an error event will be thrown	Disk/system resource issue	Check file system integrity and free disk space.
0301F113	EROR	Failed to initialize CMP	`voicexml.cfg` won't be effective.	Unexpected.	Check configuration.
0301F3FF, 0321F3FF, 0371F3FF, 0381F3FF, 03B1F3FF, 03C1F3FF, 03E1F3FF, 03F1F3FF, 0401F3FF, 0421F3FF, 0451F3FF, 0471F3FF, 0481F3FF, 0491F3FF, 04A1F3FF, 04B1F3FF, 04D1F3FF	EROR	Assertion failed	The application will be terminated, or the vxmli process exits.	Software internal problem.	Check operational state of platform. Report to Genesys with logs and traces.
0301FBD1, 0361FBD1, 0371FBD1, 03B1FBD1, 03C1FBD1, 03E1FBD1, 03F1FBD1, 0401FBD1, 0451FBD1, 0481FBD1, 0491FBD1, 04A1FBD1, 04B1FBD1, 04D1FBD1	EROR	Unexpected null pointer	Unexpected.	Software internal problem.	Check operational state of platform. Report to Genesys with logs and traces.

Alarm	Severity	Description	Impacts	Causes	Recommended Action
0301FBD3, 0371FBD3, 03B1FBD3, 0471FBD3, 04B1FBD3	EROR	Failed to define object	Unexpected.	Software internal problem.	Check operational state of platform. Restart the services or reboot the server.
0301FBD5, 0401FBD5, 0441FBD5, 0451FBD5	EROR	Failed to evaluate script	Event `EROR.semantic`, or `EROR.badfetch` or `EROR` will be thrown, depends on the cause of the failure.	Script fetching failure, invalid script in application or vxmli internally generated, or other unexpected causes.	Check operational state of platform. Restart the services or reboot the server.
0301FBDC, 0451FBDC	EROR	Failed to define session variable	The applicable session variable will be undefined.	Software internal problem.	Check operational state of platform. Restart the services or reboot the server.
0381FBDD, 03C1FBDD, 03F1FBDD, 0451FBDD, 0471FBDD, 0481FBDD, 0491FBDD, 04A1FBDD, 04B1FBDD, 04C1FBDD, 04D1FBDD	EROR	Unexpected element found	Unexpected.	Software internal problem.	Check operational state of platform. Report to Genesys with logs and traces.
04D1FBE0	EROR	Cannot get the element which created this state	Unexpected.	Software internal problem.	Check operational state of platform. Report to Genesys with logs and traces.
03F1FBE1	EROR	File seek failed	Grammar processing failure. Only apply to Nuance asr.	`fseek()` failure.	Check operational state of platform. Report to Genesys with logs and traces.

Alarm	Severity	Description	Impacts	Causes	Recommended Action
0301FBE2, 0331FBE2	EROR	Failed to set session variable	An error event will be thrown or the session variable won't be set	Session variable configuration problem	Check operational state of platform. Restart the services or reboot the server.
0401FBE3, 0481FBE3, 04D1FBE3	EROR	Unexpected tag found	Unexpected	Software internal problem.	Check operational state of platform. Report to Genesys with logs and traces.
0481FBE5, 04D1FBE5	EROR	Cannot get current dialog	Event `EROR.internal` might be thrown, or vxmli session might be terminated.	Software internal problem.	Check operational state of platform. Report to Genesys with logs and traces.
04D1FBE6	EROR	Could not remove channel from secondary channel list	Unexpected.	Software internal problem.	Check operational state of platform. Report to Genesys with logs and traces.
0401FBE7	EROR	Couldn't read expected number of bytes while reading script file	If the file is build-in grammar file, event nomatch is thrown.	`Fread()` failure.	Check operational state of platform. Report to Genesys with logs and traces.
0441FBE8	EROR	Failed to clear property	Event `EROR.internal` will be thrown.	Software internal problem.	Check operational state of platform. Restart the services or reboot the server.
0391FBE9, 0441FBE9	EROR	Cannot get object	Event `EROR.internal` will be thrown.	Software internal problem.	Check operational state of platform. Restart the services or reboot the server.
04D1FBEA	EROR	Failed to create new application for root	Event `EROR.internal` will be thrown.	Software internal problem.	Check operational state of platform. Restart the services or reboot the server.

Alarm	Severity	Description	Impacts	Causes	Recommended Action
04D1FBEB	EROR	Failed to reset application scope	Event `EROR.internal` will be thrown.	Software internal problem.	Check operational state of platform. Restart the services or reboot the server.
04D1FBEC	EROR	Failed to set document to application	Event `EROR.internal` will be thrown.	Software internal problem.	Check operational state of platform. Restart the services or reboot the server.
04D1FBED	EROR	Failed to reset document scope	Event `EROR.internal` will be thrown.	Software internal problem.	Check operational state of platform. Restart the services or reboot the server.
03E1FBEE	EROR	Failed to get scope	The `field` JS Object won't be set properly after the user's input.	Software internal problem.	Check operational state of platform. Restart the services or reboot the server.
0451FBF0	EROR	Failed to return subdialog parameter	Event `EROR.semantic` will be thrown.	Application issue or Software internal problem.	Check operational state of platform. Report to Genesys with logs and traces.
03B1FBF1	EROR	Failed to pass subdialog parameter value	Event `EROR.semantic` will be thrown.	Application issue or Software internal problem.	Check operational state of platform. Report to Genesys with logs and traces.
03B1FBF2	EROR	Failed to get subdialog parameter status string	Event `EROR.semantic` will be thrown	Software internal problem.	Check operational state of platform. Report to Genesys with logs and traces.
03B1FBF3	EROR	Failed to get subdialog parameters	Event `EROR.semantic` will be thrown	Software internal problem.	Check operational state of platform. Report to Genesys with logs and traces.
0301FBF4	EROR	Failed to set attribute value	Interpreter will exit.	Software internal problem.	Check operational state of platform. Report to Genesys with logs and traces.

Alarm	Severity	Description	Impacts	Causes	Recommended Action
04A20788	EROR	Bad `<form>` element	Unexpected.	Software internal problem.	Check operational state of platform. Report to Genesys with logs and traces.
03F2097C	EROR	Bad `<field>` element	The applicable grammar will be discarded	Software internal problem.	Check operational state of platform. Report to Genesys with logs and traces.
04520C39	EROR	Script file not ready	Event `EROR` will be thrown.	Multiple reasons.	Check operational state of platform. Report to Genesys with logs and traces.
04D21070	EROR	Cannot evaluate `expr` attribute for `<grammar>`	Event `EROR` will be thrown.	Application issue or Software internal problem.	Check VoiceXML application or application server.
04D21071	EROR	Cannot evaluate `expr` attribute for `<prompt>`	Event `EROR` will be thrown.	Application issue or Software internal problem.	Check VoiceXML application or application server.
03C2128D	EROR	Rejecting call due to failure in fetching the initial page	The applicable call will be rejected.	Multiple reasons:	Check VoiceXML application or application server.
03C2128F	EROR	Child context discontinues due to failure in fetching the initial page	The applicable vxmli session	Multiple reasons:	Check VoiceXML application or application server.
040212DD	EROR	Invalid literal format		Application issue.	Check VoiceXML application or application server.
040212DE	EROR	Invalid number format		Application issue.	Check VoiceXML application or application server.

Alarm	Severity	Description	Impacts	Causes	Recommended Action
040212DF	EROR	Invalid currency format		Application issue.	Check VoiceXML application or application server.
040212E0	EROR	Invalid date format		Application issue.	Check VoiceXML application or application server.
040212E1	EROR	Invalid time format		Application issue.	Check VoiceXML application or application server.
03F213D6	EROR	Couldn’t read expected number of bytes while reading grammar file	Event `error.application` will be thrown.	`Fread()` failure.	Check operational state of platform. Report to Genesys with logs and traces.
04D217A0	EROR	Infinite event loop	The applicable vxmli session discontinues.	Application issue or other unexpected causes.	Check VoiceXML application or application server.
04D217A1	EROR	No event list	The applicable vxmli session discontinues.	Application issue or other unexpected problem.	Check VoiceXML application or application server.
04D217A2	EROR	No event handler for event	The applicable vxmli session discontinues.	Application issue or other unexpected problem.	Check VoiceXML application or application server.
033218ED	EROR	Couldn’t update real line number	Unexpected.	`dtinsert()` failure.	Check operational state of platform. Report to Genesys with logs and traces.
033218F2	EROR	Call connected without being accepted	The call will be disconnected.	VG software problem.	Check operational state of platform. Report to Genesys with logs and traces.
03021A5C	EROR	Failed to get the host IP	Vxmli will exit.	`getHostName()` or `gethostbyname()` failure.	Check configuration.

Alarm	Severity	Description	Impacts	Causes	Recommended Action
03021A5F	EROR	Failed to create temporary directory %s	The temporary files for this particular call will be written under the `tmp` directory.	Multiple reasons.	Check file system integrity and free disk space.
03021A61	EROR	Failed to create default VoiceXML page	The new call will be rejected.	Unexpected.	Check operational state of platform. Report to Genesys with logs and traces.
03021A66	EROR	Failed to initialize standard classes for script engine	Unexpected.	Software internal issue.	Check operational state of platform. Report to Genesys with logs and traces.
08421A7A	EROR	Failed while logging user data	User data will be lost	Disk/system resource issue	Check file system integrity and free disk space.
08421A7B	EROR	Failed to open file	User data will be lost	Disk/system resource issue	Check file system integrity and free disk space.
08421A7C	EROR	File name too long	User data will be lost	The filename specified in the VoiceXML application is too long	Check VoiceXML application or application server.
08421A7D	EROR	Error while stopping user data logger thread	None	Resource issue	Report to Genesys with logs and traces.
08421A7E	EROR	Failed to create `LogMsgBase`	Vxmli will exit	Memory allocation failed by calling `new`	Restart the services or reboot the server. Reboot server and check hardware.
08421A7F	EROR	Failed to change to directory	The application specified log file (using `<log>` tag) will not be created.	`chdir` failed	Check file system integrity and free disk space.

Alarm	Severity	Description	Impacts	Causes	Recommended Action
08421A80	EROR	Failed to create directory	The application specified log file (using `<log>` tag) will not be created.	`mkdir` failed	Check file system integrity and free disk space.
04021A98	EROR	Cannot fetch audio	The alternate prompt, if any, will be played.	Multiple reasons.	Check operational state of platform. Report to Genesys with logs and traces.
04021A99	EROR	Timeout while fetching audio	The alternate prompt, if any, will be played.	Multiple possibilities.	Check operational state of platform. Report to Genesys with logs and traces.
039355C5	WARN	Transcode string is longer than expected	The NLSML ASR result will not be parsed properly.	Software internal issue.	
033376C4	WARN	Lost service	All the active call session will be terminated.	Multiple reasons, e.g.	
033376C8	EROR	Message token too long	The parameter will be truncated.	It can be the configuration problem if the applicable parameter is configurable.	Check VoiceXML application or application server.
033376CF	WARN	Hangup failed	Unexpected.	Callmgr unable to perform hang-up request for some reason.	Report to Genesys with logs and traces.
033376D0	WARN	Session not found in session list	Unexpected	Unexpected	Report to Genesys with logs and traces.
033376D1	WARN	Hanging up call but call instance not freed	Vxmli will terminate the corresponding session.	Unexpected.	Report to Genesys with logs and traces.

Alarm	Severity	Description	Impacts	Causes	Recommended Action
033376D8	WARN	Action failed – hanging up	The call will be disconnected.	Callmgr returns error on certain request made by vxmli.	Report to Genesys with logs and traces.
033376D9, 040376D9	WARN	Bogus status for action	None	Should not happen	Report to Genesys with logs and traces.
033376DA	WARN	Got `CMGR_NCONNECT` when stopping recording	None impact is expected.	Inbound line dropped for some reaon (e.g. caller hung up) before `TRANSFER` reply.	
033376DC	WARN	Cannot find target session for message	None	Can happen during normal operation	Check VoiceXML application or application server.
033376DD	WARN	Inbound call dropped before transfer reply	None		
038376DE	WARN	Cannot send log before establishing a primary channel	Logged data will be discarded.		Check VoiceXML application or application server.
030376E1, 036376E1	WARN	Unable to `unlink()`	A temporary file won't be deleted.		Check file system integrity and free disk space.
036376EB	WARN	Relative URL used as initializer	The relative url used in the application won't be resolved properly.	Initial url configuration issue.	Check VoiceXML application or application server.
036376EC	WARN	Cannot open file for concatenation	Data will be omitted	Disk/system resource issue	Check file system integrity and free disk space.

Alarm	Severity	Description	Impacts	Causes	Recommended Action
032376ED, 036376ED	WARN	Failed to create file link	The fetched VXML page can not be saved under platform `tmp` directory.	Installation/configuration issue.	Check file system integrity and free disk space.
036376F0	WARN	Cannot open encoded audio file	Post operation will fail	Disk/system resource issue	Check file system integrity and free disk space.
036376F1	WARN	Failed reading from file	Post operation will fail	Disk/system resource issue	Check file system integrity and free disk space.
03D376F8	WARN	Unable to get token – failed lexical analysis	VXML page parsing failure.	Software internal problem.	Check VoiceXML application or application server.
030376FC	WARN	Cannot add name/value pair	Call will be rejected.	Memory problem, or	Report to Genesys with logs and traces.
04537783, 04D37783	WARN	Invalid command invoked on state	None	Should not happen	Report to Genesys with logs and traces.
04D37785	WARN	`TARGET_NOT_RESOLVED` still exists at runtime	Vxmli state machine will move to the next state in sequence.	Software internal problem.	Report to Genesys with logs and traces.
04D37786	WARN	Requested invalid state	Vxmli session discontinues.	Software internal problem.	Check operational state of platform. Report to Genesys with logs and traces.
04D3778E	WARN	Cannot insert duple	Vxmli session discontinues.	Memory problem	Report to Genesys with logs and traces.
03F37792, 04037792	WARN	Prepare failed	If asr context file, event	Application issue.	Report to Genesys with logs and traces.

Alarm	Severity	Description	Impacts	Causes	Recommended Action
03C37799	WARN	Missing default followLink state	Vxmli will not be able to move to the right state. The behavior is unexpected.	Software internal issue.	Report to Genesys with logs and traces.
0373779A	WARN	Invalid outbound line	Incorrect `<transfer>` behavior.	Software internal issue.	Report to Genesys with logs and traces.
0373779B	WARN	Wrong conference line for analysis	Incorrect `<call>` behavior	Software internal issue.	Check VoiceXML application or application server.
03038271	WARN	Cannot add file to delete list	Temporary file might not be cleaned up from the `tmp` directory after the call ends.	`dtinsert()` failure.	Check operational state of platform. Report to Genesys with logs and traces.
03738272, 04038272, 04238272, 04538272, 04638272, 04738272	EROR	Failed to define variable	The applicable variable will not be defined properly.	It can be a VXML application problem or vxmli software internal issue.	Check operational state of platform. Restart the services or reboot the server.
03C38274, 04538274	WARN	Failed to get variable name/value	Event `EROR` may be thrown, otherwise, the particular variable is ignored.	It can be an application problem, or VG software internal issue.	Check operational state of platform. Restart the services or reboot the server.

Alarm	Severity	Description	Impacts	Causes	Recommended Action
03038276, 03738276, 03838276, 03B38276, 03C38276, 03E38276, 04038276, 04238276, 04438276, 04538276, 04638276, 04738276, 04C38276, 04D38276	EROR	Failed to evaluate expression	An error event will be thrown or the current operation will be interrupted	There's an invalid ECMAScript expression in the VoiceXML application	Check VoiceXML application or application server.
03038277, 03738277, 04038277, 04738277, 04838277, 04B38277	WARN	Failed to define variable	The applicable shadow variable will not be defined properly.	Software internal issue.	Check operational state of platform. Restart the services or reboot the server.
03F38278	WARN	Cannot get file status	Nuance grammar processing failure, event `EROR` is thrown.	`Stat ()` failed.	Check file system integrity and free disk space.
03738279, 04538279, 04638279, 04738279, 04D38279	WARN	Failed to set variable value	The corresponding variable is not set with correct value.	Software internal issue.	Check VoiceXML application or application server.
0453827A, 04C3827A, 04D3827A	WARN	Failed to set primitive value	The corresponding variable is not set with correct value.	Software internal issue.	Check VoiceXML application or application server.
0423827E, 0493827E, 04A3827E, 04C3827E	WARN	Failed to clear scope	Event	Software internal issue.	Report to Genesys with logs and traces.

Alarm	Severity	Description	Impacts	Causes	Recommended Action
04038284	WARN	Failed to evaluate cond expression	The corresponding grammar might not be effective.	Software internal issue.	Check VoiceXML application or application server.
03F3901D	WARN	`Mode=dtmf` should have been handled by `<dtmf>`	The `<grammar>` element will be ignored	Should not happen	Report to Genesys with logs and traces.
03C3992C	WARN	Cannot insert into `formResults`	The particular variable won't be submitted when executing `<submit>`	Software internal issue.	Check operational state of platform. Report to Genesys with logs and traces.
03C3992E	WARN	Failure in fetching the root document	Event	Multiple possible reasons:	Check VoiceXML application or application server.
0403997C	WARN	Cannot prepare grammar	Event `EROR` is thrown.	Multiple possible reasons.	Report to Genesys with logs and traces.
04039982	WARN	Invalid date format	Incorrect TTS playing behavior.	Application issue.	
04039983	WARN	Invalid time format	Incorrect TTS playing behavior.	Application issue.	
04B39984	WARN	Invalid `inputmode` property on page	The default input mode `BOTH` is taken.	Application issue.	Check VoiceXML application or application server.
04739DAA	WARN	Sender or receiver address not found in message	Event `EROR` is thrown.	Software internal issue.	Check VoiceXML application or application server.
04539E2C, 04C39E2C, 04D39E2C	WARN	Unexpected result from `scriptElement::load()`	Event `EROR` will be thrown, and vxmli session will discontinue.	Software internal issue.	Check VoiceXML application or application server.

Alarm	Severity	Description	Impacts	Causes	Recommended Action
03339F8A	WARN	Null session object pointer found in session list	Unexpected.	Software internal issue.	Report to Genesys with logs and traces.
03339F8C, 04239F8C, 04D39F8C	WARN	Abort action failed	If in event handler, FIA will revisit the current dialog, otherwise, the impact is unexpected.	Unexpected.	Report to Genesys with logs and traces.
0303A102	WARN	Temporary directory is not empty	The temporary directory remains under the platform `tmp` directory after the session ends.	If `sametmpfiles` is enabled, this behavior is expected.	
0303A103	WARN	Failed to remove directory	The temporary directory remains under the platform `tmp` directory after the session ends.	`rmdir()` failure.	Check file system integrity and free disk space.
0303A104	WARN	Failed to open directory	The temporary directory remains under the platform `tmp` directory after the session ends.	`opendir()` failure.	Check file system integrity and free disk space.
0303A105	WARN	Cannot move file from the `nextappfiles` to `appfiles`	Temporary file might not be cleaned up from the `tmp` directory after the call ends.	`dtinsert()` failure.	Check operational state of platform. Report to Genesys with logs and traces.

Alarm	Severity	Description	Impacts	Causes	Recommended Action
0303A107	WARN	No `<vxml>` element in page	The compilation of the page will discontinue and the call will be rejected or disconnected.	VXML application issue.	Check VoiceXML application or application server.
0453A14C	CRIT	UTF-16 conversion for `<script>` failed	Loading scrip failure. Event `EROR` will be thrown.	Software internal issue.	Check VoiceXML application or application server.
0303A17E	WARN	`VxmlElement` already deleted	None	Software internal issue.	Report to Genesys with logs and traces.
0363A1B0	WARN	Unsupported audio format	Utterance recoding won't work properly. Event `EROR.internal` is throw.	The audio format of the utterance recorded by the asr engine is unsupported format (e.g. g726 ADPCM 2-bit).	Check VoiceXML application or application server. Check file system integrity and free disk space.
0453A214	WARN	Malformed access-control PI data for `<data>`	The Malformed access-control data is ignored.	Application problem.	Check VoiceXML application or application server.

Chapter

11 Fetching Module Alarms

Alarm	Severity	Description	Impacts	Causes	Recommended Action
05102710, 05202710, 05402710, 05502710, 05602710, 05802710	CRIT	Memory allocation failed	VXMLi cannot fetch from internet	Share Memory growing too large – Too many concurrent channels or Software problem.	Restart the services or reboot the server.
05502711, 05602711	CRIT	Fetching Module initialization failed	VG software does not start	Most times this is a configuration issue; otherwise it's likely a problem with the state of the share memory in the platform.	Check operational state of platform. Report to Genesys with logs and traces.
05504E20	EROR	Open Session to Fetching Server failed	Either the callmgr or the vxmli cannot open a new session to fetch something via the fetching module	Most likely, the shared memory is corrupted, probably due to a programming error.	Check operational state of platform. Report to Genesys with logs and traces.
05504E21	EROR	Connect to Fetching Server failed.	Either the callmgr or the vxmli cannot connect to the fetching module	Either pwproxy hasn't started, or the shared memory is corrupted.	Check operational state of platform. Report to Genesys with logs and traces.

Alarm	Severity	Description	Impacts	Causes	Recommended Action
05504E22	EROR	Send to Fetching Server failed	Either the callmgr or the vxmli cannot send a message to the fetching module	Most likely, the shared memory is corrupted, probably due to a programming error.	Check operational state of platform. Report to Genesys with logs and traces.
05504E23	EROR	Invalid session ID	When using the fetching module, the callmgr or the vxmli uses an invalid Session ID to use the fetching module	The pwproxy may have restarted due to a programming error; another possibility is the shared memory is corrupted.	Check operational state of platform. Report to Genesys with logs and traces.
05204E24	EROR	CMP configuration setup failed	VG platform would fail to start	Most times this is a configuration issue; otherwise it's likely a problem with the state of the share memory in the platform.	Check configuration.
05204E25	EROR	CMP Agent initialization failed	VG platform would fail to start	Most times this is a configuration issue	Check configuration.
05204E26	EROR	CMP logging service initialization failed	VG platform would fail to start	Most times this is a configuration issue	Check operational state of platform. Report to Genesys with logs and traces.
05404E27	EROR	Invalid shared memory parameter	The particular fetch associated with the shared memory would fail	VoiceGenie software programming problem.	Check operational state of platform. Report to Genesys with logs and traces.
05404E28	EROR	Empty shared memory name	The particular fetch associated with the shared memory would fail	VoiceGenie software programming problem.	Check operational state of platform. Report to Genesys with logs and traces.

Alarm	Severity	Description	Impacts	Causes	Recommended Action
05404E29	EROR	Shared semaphore name generation failed	The particular fetch associated with the shared memory would fail	VoiceGenie software programming problem.	Check operational state of platform. Report to Genesys with logs and traces.
05404E2A	EROR	Shared semaphore creation failed	The particular fetch associated with the shared memory would fail; it's also likely that any further web fetches will fail.	VoiceGenie software programming problem.	Check operational state of platform. Report to Genesys with logs and traces.
05404E2B	EROR	Shared semaphore lock failed	The particular fetch associated with the shared memory would fail	VoiceGenie software programming problem.	Check operational state of platform. Report to Genesys with logs and traces.
05404E2C	EROR	Shared memory map failed for specified file	The particular fetch associated with the shared memory would fail	VoiceGenie software programming problem; may also be due to problem in file system	Check operational state of platform. Report to Genesys with logs and traces.
05404E2D	EROR	Shared memory attach failed for specified ID	The particular fetch associated with the shared memory would fail	VoiceGenie software programming problem.	Check operational state of platform. Report to Genesys with logs and traces.
05404E2E	EROR	Shared memory name generation failed	The particular fetch associated with the shared memory would fail	VoiceGenie software programming problem.	Check operational state of platform. Report to Genesys with logs and traces.

Alarm	Severity	Description	Impacts	Causes	Recommended Action
05404E2F	EROR	Shared memory creation failed for specified size	The particular fetch associated with the shared memory would fail; it's also likely that any further web fetches will fail.	Share Memory growing too large – Too many concurrent channels or it could be VoiceGenie Programming problem.	Check operational state of platform. Report to Genesys with logs and traces.
05404E30	EROR	Unable to read shared-memory	The particular fetch associated with the shared memory would fail	VoiceGenie software programming problem.	Check operational state of platform. Report to Genesys with logs and traces.
05404E31	EROR	Unable to write shared-memory	The particular fetch associated with the shared memory would fail	VoiceGenie software programming problem.	Check operational state of platform. Report to Genesys with logs and traces.
05404E32	EROR	Failed to get pipe name	The particular fetch associated with the shared memory would fail	VoiceGenie software programming problem, when the callmgr and the vxmli communicates with the pwproxy.	Check operational state of platform. Report to Genesys with logs and traces.
05404E33	EROR	Failed to open pipe	The particular fetch associated with the shared memory would fail	VoiceGenie software programming problem, when the callmgr and the vxmli communicates with the pwproxy.	Check operational state of platform. Report to Genesys with logs and traces.

Alarm	Severity	Description	Impacts	Causes	Recommended Action
05507530, 05807530	WARN	Close Session to Fetching Server failed	Either the callmgr or the vxmli cannot close a new session to fetch something via the fetching module	Most likely, the shared memory is corrupted, probably due to a programming error. If too many of these errors occur, it may lead to memory leak	Check operational state of platform.
05407531	WARN	Shared memory unmap failed for specified file	When closing a fetch session, the cleanup cannot be cleanly done.	Most likely, the shared memory is corrupted, probably due to a programming error. If too many of these errors occur, it may lead to memory leak	Check operational state of platform. Report to Genesys with logs and traces.
05407532	WARN	Shared memory detach failed	When closing a fetch session, the cleanup cannot be cleanly done.	Most likely, the shared memory is corrupted, probably due to a programming error. If too many of these errors occur, it may lead to memory leak	Check operational state of platform. Report to Genesys with logs and traces.
05407533	WARN	Failed to close pipe	When closing a fetch session, the cleanup cannot be cleanly done.	Most likely, the shared memory is corrupted, probably due to a programming error. If too many of these errors occur, it may lead to memory leak	Check operational state of platform. Report to Genesys with logs and traces.

Chapter

12 VGcomm Alarms

VGComm is the message transport library used between the Legacy interpreter and the call manager. Hence, the alarms from the VGComm module can be generated from both the Legacy Interpreter and the call manager process.

Alarm#	Level	Definition and Possible Message/Info	Impacts	Causes	Detailed Recommended Actions
083007D1	EROR	Unable to reply to client	Cannot reply the message	Connection is broken	Check operational state of the server
083007D2	EROR	Unable to send message	Cannot send the message	Connection is broken	Check operational state of the server
083007D3	EROR	Bad message format	Cannot parse the message	Wrong message format	Report to VoiceGenie [with logs]
083007D4	EROR	names/values list too long!	Cannot send the message	The names/values list is too long	Check configuration; check application; check operational state of the server; Report to VoiceGenie [with logs]
083007D5	EROR	Calling `NVPairNext()` with bad `prev_result`	Cannot parse the message	Wrong message format	Report to VoiceGenie [with logs]
083007D6	EROR	Name-Value pair string has odd number of tokens	Cannot parse the message	Wrong message format	Report to VoiceGenie [with logs]
083007D7	EROR	Invalid message header	Cannot parse the message	Wrong message header	Report to VoiceGenie [with logs]

Alarm#	Level	Definition and Possible Message/Info	Impacts	Causes	Detailed Recommended Actions
083007D8	EROR	Failed to start thread	Cannot start the connection manager thread	Unexpected system error	Report to VoiceGenie [with logs]
083007D9	EROR	Error when stopping thread	Cannot stop the connection manager thread	Unexpected system error	Report to VoiceGenie [with logs]
083007DA	EROR	Did not found the ConnID	Cannot reconnect or verify the current connection	Internal Error	Report to VoiceGenie [with logs]
083007DB	ERPR	Failed to create Client Socket	Cannot connect to the remote party	Unexpected system error	Report to VoiceGenie [with logs]
083007DC	EROR	`ConnectToRemote()` failed	Cannot connect to the remote party	Remote died; unexpected system error	Check operational state of the server; Report to VoiceGenie [with logs]
083007DD	EROR	Failed to send data	Cannot send the verification message	Connection is broken	Check operational state of the server; Report to VoiceGenie [with logs]
083007DE	WARN	Failed to send message: msg too big	Cannot send the message	The message is too big	Check operational state of the server; Report to VoiceGenie [with logs]
083007DF	WARN	Failed to send message to service	Cannot send the message	Connection is broken	Check operational state of the server; Report to VoiceGenie [with logs]

Chapter

13 SIP Response Code Handling

13.1 SIP Reponse Codes For Inbound Call Setup Errors

The VoiceGenie Media Platform will signal the following responses if an error occurs during an incoming call setup.

Response Code	Response Phrase	Situations
400	Bad Request	Repeated or malformed Burke Draft init-parameters. gvp.appmodule parameter is found in the SIP Request URI but voicexml parameter is not specified.
487	Request Terminated	Receive `CANCEL` while in `INVITED` or `PROVSENT` state Receive `BYE` while in `INVITED` or `PROVSENT` state
488	Not Acceptable Here	Error in SDP negotiation.
500	Internal Server Error	Unable to create `RTP` session or unable to create `PSTN` session Unable to fetch or parse the VoiceXML document specified in the SIP INVITE.

Response Code	Response Phrase	Situations
503	Service Unavailable	VG media platform is not accepting new call Disconnect while initiated or provisional sent state Note that when using the SIP protocol, a `503` response would be generated for `<meta name="callrequest" content="decline"/>` And when used in combination with the SS7 Connector a call will be shown being rejected with the default cause code `41,` instead of the cause code defined by the Call Manager configuration parameter `sessmgr.disconnect_cause.decline`

13.2 Handling Of Received SIP Error Responses When Making Outbound Calls

The VoiceGenie Media Platform will interpret the following responses from outbound call setup.

Response Code	Metrics Reason	Call End Reason (for call_end, bridge_end, outcall_end)	Transfer Result	VXML Action C–<call> T–<transfer> R–<remdial>
301 404 410 484 502	Moved Permanently Not Found Gone Address Incomplete Bad Gateway	Baddest	baddest	C: `<call>` variable set to `invalid_phone_no` T: `error.connection.baddestination` event thrown R: `UNKNOWN_REASON` logged

Response Code	Metrics Reason	Call End Reason (for call_end, bridge_end, outcall_end)	Transfer Result	VXML Action C–<call> T–<transfer> R–<remdial>
401 402 403 407	Unauthorized Payment Required Forbidden Proxy Authentication Required	Noautho	noautho	C: When `returnwhen=answered`, `<call>` variable set to `failed`. When `returnwhen=immediate`, `com.voicegenie.call.failed` event thrown. T: `error.connection.noroute event thrown (VXMLi)` `error.connection.noauthorization event thrown (NGI)` R: `UNKNOWN_REASON` logged
405 488 501	Method Not Allowed Not Acceptable Here Not Implemented	unsupported	unsupported	C: N/A T: `error.unsupported.transfer.unknown`, `error.unsupported.transfer.blind` or `error.unsupported.transfer.consultation` event thrown or `error.unsupported.transfer.bridge` event thrown R: `UNKNOWN_REASON` logged
408	Request Timeout	noanswer	noanswer	C: When `returnwhen=answered`, `<call>` variable set to `noanswer`. When `returnwhen=immediate`, `com.voicegenie.call.noanswer` event thrown. T: `<transfer>` variable set to `noanswer` R: `NO_ANSWER` logged

Response Code	Metrics Reason	Call End Reason (for call_end, bridge_end, outcall_end)	Transfer Result	VXML Action C–<call> T–<transfer> R–<remdial>
480 486	Temporary Unavailable Busy Here	busy	unknown	C: When `returnwhen=answered`, `<call>` variable set to `busy`. When `returnwhen=immediate`, `com.voicegenie.call.busy` event thrown. T: `<transfer>` variable set to `busy` R: `BUSY` logged
503	Service Unavailable	noresource	noresource	C: When `returnwhen=answered`, `<call>` variable set to `noresource`. When `returnwhen=immediate`, `com.voicegenie.call.failed` event thrown. T: `error.connection.noresource` event thrown R: `NO_RESOURCES` logged
504	Gateway Timeout	networkbusy	busy (VXMLi) network_busy (NGI)	C: When `returnwhen=answered`, `<call>` variable set to `busy`. When `returnwhen=immediate`, `com.voicegenie.call.busy` event thrown. T: `<transfer>` variable set to `busy (VXMLi)` `<transfer>` variable set to network_`busy (NGI)` R: `BUSY` logged
Others	Error	error	error	C: When `returnwhen=answered`, `<call>` variable set to `failed`. When `returnwhen=immediate`, `com.voicegenie.call.failed` event thrown. T: `error.connection.noroute` event thrown R: `CALL_FAILED` logged

Revision History

Version	Date	Change Summary
1.0	March 11th, 2005	Initial release
1.1	April 13th, 2005	Revised Version for VoiceGenie 7 Release
1.2	December 12nd, 2005	Updated Fetching Module traps
1.3	August 30th, 2006	Updated for Release 7.1
1.4	November 7th, 2006	Updated for Release 7.1.1
1.5	November 23rd, 2006	Updated for 7.1.1
1.6	January 30th, 2007	Updated for 7.1.2
1.7	October 16rd, 2007	Updated for 7.2