Solution Blueprint GIR and GIA

Reference Architecture

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Revision History

Rev	Date Published	Author	Reason for Revision
0.1	10/15/2014	Rakesh Tailor	Initial skeleton document
0.2	10/29/2014	Rakesh Tailor	Initial Draft for GIA Only Solution
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1.0	01/07/2015	Rakesh Tailor	Reviewed GIR content
1.1	05/15/2016	Tim Brown	Updates following SEO architecture review
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1 Introduction

The purpose of this Blueprint is to document the architecture for Genesys Interaction Recording (GIR) and Genesys Interaction Analytics (GIA) which are major parts of the Workforce Optimization (WFO) Solution. This document provides a prescriptive list of components (both Genesys and 3rd party) that should be included in the solution. It also provides guidance for implementing and deploying the solution including sizing and configuration as well as addressing several system concerns such as security, high availability, disaster recovery and serviceability.

1.1 Document Overview

The document contains the following sections:

- 2: Definitions and Acronyms
- 3: Overall Architecture
- 4: Deployment View
- 5: Interaction View
- 6: Implementation View
- Appendix A: Screen Recording Service
- Appendix B: Recording – SIP in front – Skype for Business
- Appendix C: Migration Strategy

1.2 Intended Audience

Describing system and solution architectures can be difficult as there are multiple audiences each with different expectations. This document is intended for multiple audiences with various chapters being more interesting to some readers than others.

The Overall Architecture and Deployment View are likely meaningful to most audiences. However the Interaction View and the Implementation View may be of more interest to those configuring the network and components.

2 Definitions, Acronyms, and Document Standards

2.1 Definitions

This document uses various abbreviations and acronyms that are commonly used in Genesys product documentation and the telecommunications and contact center industries. The following table defines terms that will be referenced subsequently in this document.

2.2 Glossary

CME	Configuration Management Environment, another name for the Configuration Layer
CS	Config Server
CSP	Config Server Proxy
CTI	Computer-telephony integration, the adding of computer intelligence to monitoring and control of telephone calls
DB	Database
DBMS	Database Management System
DHCP	Dynamic Host Configuration Protocol
DN	Directory number
DNS	Domain Name System
FTP	File Transfer Protocol
GA	Genesys Administrator
GAX	Genesys Administrator Extension
GIM	Genesys InfoMart
GI2	Genesys Interactive Insights
GUI	Graphical User Interface
GVP	Genesys Voice Platform
GWS	Genesys Web Services
HA	High Availability
HTTP	Hypertext Transfer Protocol
ICON	Interaction Concentrator
IP	Internet Protocol
IVR	Interactive Voice Response

IWS	Interaction Workspace
IVR	interactive voice response
LAN	Local Area Network
LCA	Local Control Agent
LM	License Manager
OEM	Original Equipment Manufacturer
ORS	Orchestration Server
OS	Operating System
RDBMS	Relational Database Management System
SCS	Solution Control Server
SCXML	State Chart XML: State Machine Notation for Control Abstraction
SQL	Structured Query Language
TLib	TServer Library
URS	Universal Routing Server
VM	Virtual Machine
WAN	Wide Area Network

2.3 Document Conventions

The following documentation and naming conventions are used throughout the document:

- Code and `configuration property names & values` will appear in `console font`.
- References to other documents are bracketed ([]).

3 Solution Architecture (Overview)

The Genesys Workforce Optimization (WFO) solution is centered around a number of separate but related products that are all focused on optimizing your agent workforce in various ways. This document focuses on two particular solutions that are part of the larger WFO solution: Genesys Interaction Analytics (GIA) and Genesys Interaction Recording (GIR).

Genesys Interaction Analytics (GIA)

Genesys Interaction Analytics (GIA) discerns and collects analytical information from the myriad of interactions that occur between customers and customer contact agents. These interactions include voice interactions captured in audio recordings as well as text interactions resulting from email, chat, and social media interactions.

Analytics are performed on voice interactions in two distinct ways. Firstly, audio recordings are processed through speech recognition technology to find instances of specific utterances or phrases that have been programmed into the system; this approach is called grammar-based speech analytics. Secondly, the audio recording is processed to create a transcript of the conversation using speech to text processing. The resulting data, from both the grammar based approach as well as the speech to text approach, along with other non-linguistic properties of the audio and other attached metadata, can be used to categorize all voice interactions encountered by the system to give insight and analytics on the customer interactions.

Similarly, text interactions can also be categorized in much the same way to give a holistic view into customer interactions through all channels across an enterprise.

The voice and text interactions processed by GIA can be sourced from any location. Voice recordings may be provided by Genesys' native recording platform (Genesys Interaction Recording) or imported from third party recording platforms. Likewise, text interactions can be sourced from products within the Genesys Digital Channels solution or other third party non-voice-centric customer contact platforms.

Genesys Interaction Recording (GIR)

Genesys Interaction Recording (GIR) records all types of interactions within your deployment. This can include voice recordings as well as screen recordings of agents' desktops.

Genesys Interaction Recording is a compliance and control platform based on Genesys SIP, T-Lib protocol, and the Genesys proprietary event model. Fully integrated to the Customer Interaction Management (CIM) platform, Genesys Interaction Recording provides economies and powerful recording control by way of a host of integrations across the suite. Interaction recording uses Media Stream Replication (MSR) with Dual Channel Recording. SIP sessions to the recorder provide basic call information and voice (Real-time Transport Protocol (RTP) data. MSR is where Media Server replicates the RTPs and makes them available to the recording server. Additional events and information are provided by the T-Server part of SIP Server and TServers themselves.

The GIR and GIA solutions share many components; however, they can be deployed in different combinations based on the needs of the customer. There are three major scenarios that may exist:

a. GIA Only—in this scenario interactions are coming from one or more third party recording platforms
b. GIR Only—in this scenario Genesys software is recording customer interactions and analytics are not being performed
c. GIR with Analytics—in this scenario Genesys software is recording the customer interactions and analytics are also performed on these interactions

3.1 Solution Component Summary

3.1.1 Genesys Interaction Analytics (GIA) Solution Components

The following lists the key solution components required to support the Genesys Interaction Analytics (GIA) solution.

- The Genesys SpeechMiner system (Interaction Analytics)
  - UPlatform service - Manages all the processing tasks of SpeechMiner—fetching (in the case of Analytics mode), recognition and exploration (in the case of Analytics and Analytics & Recording UI modes), categorization compression, and indexing (in all modes).
  - UConnector Service - Retrieves interaction data (audio or text) and metadata from the 3rd party recording systems.
  - SpeechMiner Web Service - Runs the SpeechMiner web-based interface that enables users to view and work with the interaction data after it has been processed.
- Recognition Engine (3rd Party Nuance recognizer) and license server
- SpeechMiner administration tool (SMART) - An application that enables users to configure the speech-analytics system to search calls for specific topics and other characteristics.
- SMConfig - An application that is used by system administrators to configure SpeechMiner
- SMUpgrade - An application used to upgrade the SpeechMiner database from the previous version to current version.
- Microsoft SQL Server database (3rd Party) – Either version 2008 SP1 or 2012 with Reporting Services
- Microsoft .NET Framework 4.5 SP1 (4.5.1) or higher and .NET 2.0
- Microsoft Visual C++ 2010 Redistributable and Microsoft Visual C++ 2013 Redistributable
- Internet Information Server (IIS)
- Microsoft's Report Viewer (Optional)

3.1.2 Genesys Interaction Recording (GIR) Solution Components

The following lists the key solution components required to support the Genesys Interaction Recording (GIR) solution.

- Management Framework — The foundation for all Genesys-based interaction management systems. Management Framework provides you with the following administration functions: Configuration, Access Control, Solution Control, Alarm Processing, Troubleshooting, and Fault Management.
- Genesys Administrator and Genesys Administrator Extension — The user-friendly interfaces that perform complex operations such as configuration, maintenance, and administrator of the contact center objects.
- SIP Server — Genesys SIP based ACD which is used for both routing and call control. SIP Server is responsible to initiate call recording by using media control to direct media towards Media Server.
- Resource Manager — A SIP Proxy that manages a pool of Genesys Media Servers and applies runtime policies such as ensuring call legs to the same conference are pinned to the same Media Server. Resource Manager also generates the call detail record (CDR) that allows correlation with individual call recordings.
- Media Server — Performs the actual file-based recording. Media Server is also responsible for negotiating the media between the endpoints, to minimize the need for transcoding, or to preserve security of the audio stream.
- Reporting Server — An optional component in this solution; this provides storage of CDR and call events for Resource Manager and Media Server, and provides a web service to provide the user (through Genesys Administrator) the ability to query CDR and other call event information.
- Workspace Desktop Edition — The smart-client application that provides agents and knowledge workers with non-intrusive access to the information, processes, and applications they need to perform their jobs more efficiently and to ensure increased customer satisfaction.
- SpeechMiner Web Service — This is the same user interface used by Genesys Interaction Analytics (GIA) however with GIR the function of SpeechMiner is limited to perform indexing, searching and playback of recordings.
- Recording Processor — Responsible for sending call recording metadata to the Web Services and SpeechMiner servers.
- Recording Crypto Server — Provides the Key Management System for the Genesys Interaction Recording solution.
- Recording Plug-in for Genesys Administrator Extension—Provides the user the ability to manage and administer recording certificates and policies.
- Screen Recording Client — Records screens at the direction of Genesys Web Services, encrypts them using certificates indicated by Genesys Web Services, and then uploads them to Genesys Web Services.
- Genesys Web Services API — A REST API that provides a web client interface to access Genesys services.
- Interaction Concentrator (ICON) — The server for the Interaction Database (or IDB) to store detailed reporting data from various sources in a contact center empowered with Genesys software.

- Interaction Server — The component that is responsible, in concert with the Routing components, to route interactions (non-voice) according to interaction workflows and routing strategies.
- Apache HTTP Server (WebDAV)
- Cassandra Database
- Microsoft SQL Server database (3rd Party) – Either version 2008 SP1 or 2012 with Reporting Services
- Microsoft .NET Framework 4.5 SP1 (4.5.1) or higher and .NET 2.0
- Microsoft Visual C++ 2010 Redistributable and Microsoft Visual C++ 2013 Redistributable
- Internet Information Server (IIS)

3.1.3 Genesys Interaction Recording and Analytics Solution Components

For completeness the follow lists the key solution components required to support a combined Genesys Interaction Recording (GIR) and Genesys Interaction Analytics solution.

- Management Framework — The foundation for all Genesys-based interaction management systems. Management Framework provides you with the following administration functions: Configuration, Access Control, Solution Control, Alarm Processing, Troubleshooting, and Fault Management.
- Genesys Administrator and Genesys Administrator Extension — The user-friendly interfaces that perform complex operations such as configuration, maintenance, and administrator of the contact center objects.
- SIP Server — Genesys SIP based ACD which is used for both routing and call control. SIP Server is responsible to initiate call recording by using media control to direct media towards Media Server.
- Resource Manager — A SIP Proxy that manages a pool of Genesys Media Servers and applies runtime policies such as ensuring call legs to the same conference are pinned to the same Media Server. Resource Manager also generates the call detail record (CDR) that allows correlation with individual call recordings.
- Media Server — Performs the actual file-based recording. Media Server is also responsible for negotiating the media between the endpoints, to minimize the need for transcoding, or to preserve security of the audio stream.
- Reporting Server — An optional component in this solution; this provides storage of CDR and call events for Resource Manager and Media Server, and provides a web service to provide the user (through Genesys Administrator) the ability to query CDR and other call event information.
- Workspace Desktop Edition — The smart-client application that provides agents and knowledge workers with non-intrusive access to the information, processes, and applications they need to perform their jobs more efficiently and to ensure increased customer satisfaction.
- Recording Processor — Responsible for sending call recording metadata to the Web Services and SpeechMiner servers.

- Recording Crypto Server — Provides the Key Management System for the Genesys Interaction Recording solution.
- Recording Plug-in for Genesys Administrator Extension—Provides the user the ability to manage and administer recording certificates and policies.
- Screen Recording Client — Records screens at the direction of Web Services, encrypts them using certificates indicated by Web Services, and then uploads them to Web Services.
- Web Services API — A REST API that provides a web client interface to access Genesys services.
- Interaction Concentrator (ICON) — The server for the Interaction Database (or IDB) to store detailed reporting data from various sources in a contact center empowered with Genesys software.
- Interaction Server — The component that is responsible, in concert with the Routing components, to route interactions (non-voice) according to interaction workflows and routing strategies.
- The Genesys SpeechMiner system (Interaction Analytics)
    - UPlatform service - Manages all the processing tasks of SpeechMiner—fetching (in the case of Analytics mode), recognition and exploration (in the case of Analytics and Analytics & Recording UI modes), categorization compression, and indexing (in all modes).
    - Interaction Receiver — A web service which receives calls (audio and metadata) from the Genesys Interaction Recording system.
    - SpeechMiner Web Service - Runs the SpeechMiner web-based interface that enables users to view and work with the interaction data after it has been processed.
- Recognition Engine (3rd Party Nuance recognizer) and license server
- SpeechMiner administration tool (SMART) - An application that enables users to configure the speech-analytics system to search calls for specific topics and other characteristics.
- SMConfig - An application that is used by system administrators to configure SpeechMiner
- SMUpgrade - An application used to upgrade the SpeechMiner database from the previous version to current version.
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- Microsoft Visual C++ 2010 Redistributable and Microsoft Visual C++ 2013 Redistributable
- Internet Information Server (IIS)
- Microsoft's Report Viewer (Optional)

3.2 Logical Architecture Model

3.2.1 GIA Logical Architecture Model

The following is a logical model of the Genesys Interaction Analytics (GIA) architecture.

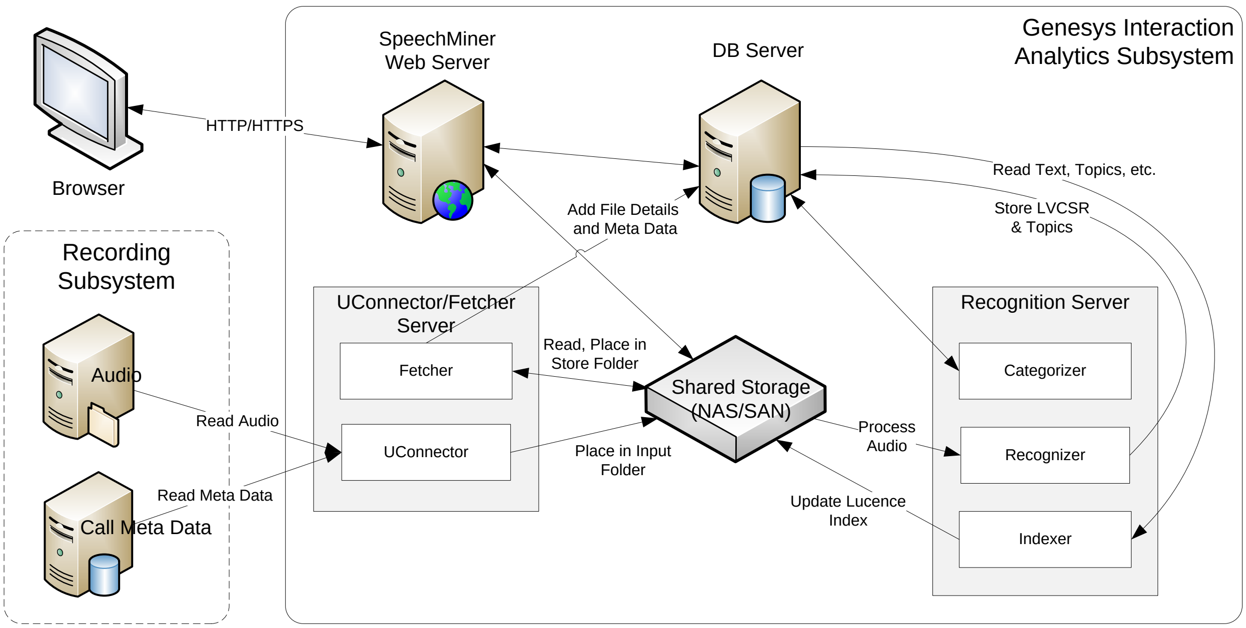


Figure 1 – GIA Solution Logical Model

3.2.2 GIR Logical Architecture Model

The following is a logical model of the Genesys Interaction Recording (GIR) architecture.

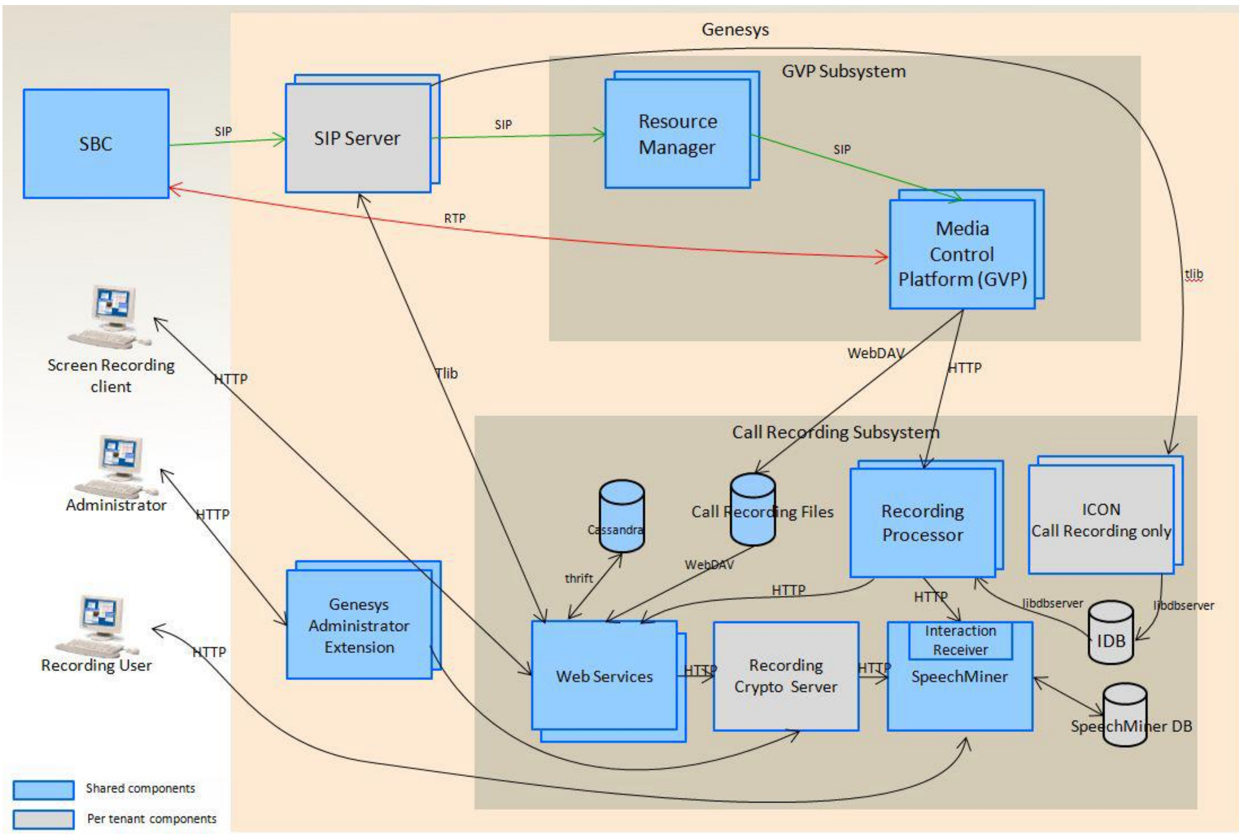


Figure 2 – GIR Solution Logical Model

3.2.3 GIA and GIR Logical Architecture Model

The following is a logical model of the combined Genesys Interaction Analytics (GIA) and Genesys Interaction Recording (GIR) architecture.

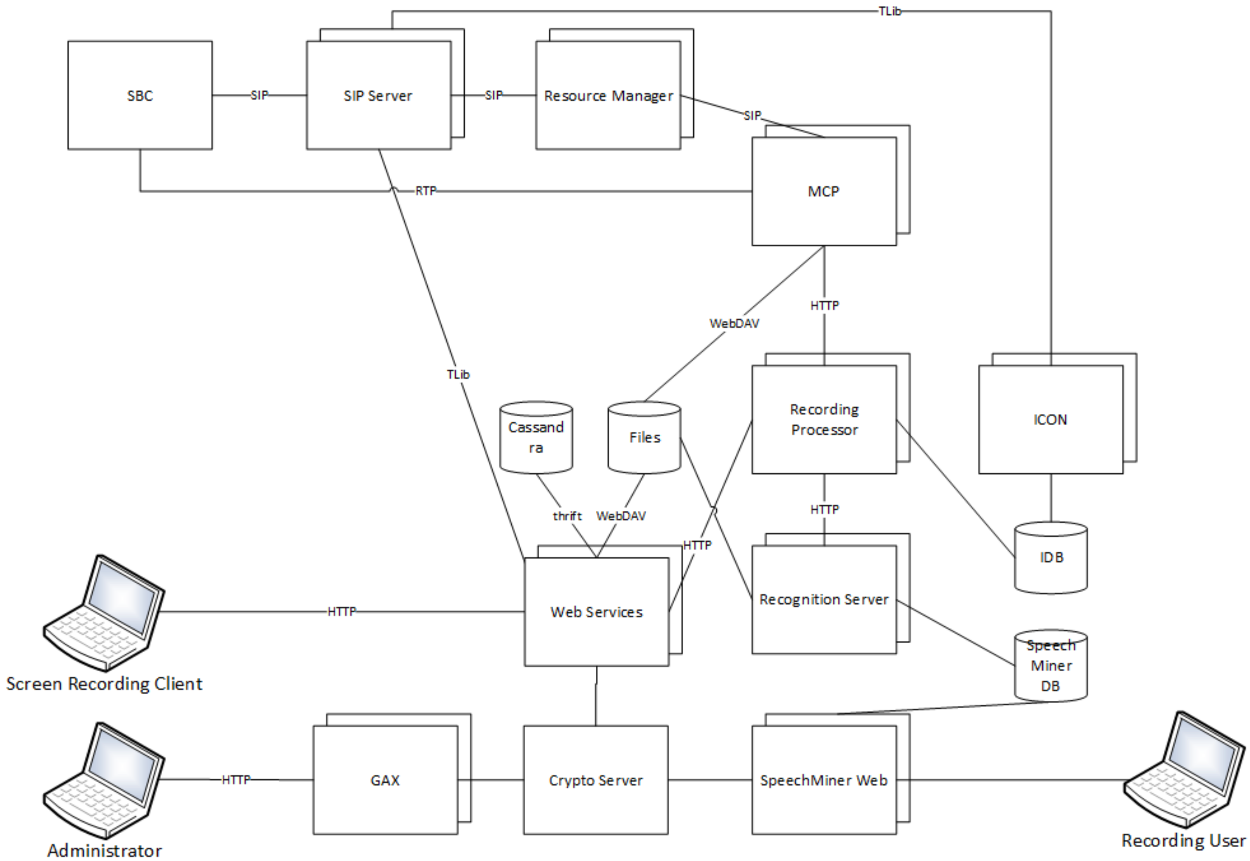


Figure 3 – GIR and GIA Solution Logical Model

3.3 Functional View

The core functionalities of the GIR and GIA solutions can be broken down into the following areas:

- Interaction Acquisition
- Analytics Processing (GIA only)
- Configuration Tools
- User Interface and Reporting

3.3.1 Interaction Acquisition

3.3.1.1 Interaction Acquisition – GIR only and GIR/GIA

Genesys Interaction Recording (GIR) is a compliance and control platform based on Genesys SIP, the T-Lib protocol, and the Genesys proprietary event model. Fully integrated to the Customer Interaction Management platform, GIR provides economies and powerful recording control via a host of integrations across the suite. The voice portion of interaction recording relies on Media Server to perform Dual Channel Recording, where Media Server captures the audio from the RTP streams and makes them available to the recording storage. The screen portion of interaction recording relies on a Screen Recording Service running on the Agent Desktop to perform screen captures and makes them available to the recording storage by way of Genesys Web Services (GWS). Additional events and information are provided by the SIP Server (via ICON) for voice interactions and by the Interaction Server for non-voice interactions (like emails, chats, etc.).

Recording can be implemented with the following controls:

- Full-time Recording: Records every call for a specific DN through configuration.
- Selective Recording: A decision to record a party in the call is made at a Routing Point and the recording starts as soon as the call is established.
- Dynamic Recording: Recording sessions are established on an as-needed basis after the communication session is established. T-Library recording functions are provided to allow third parties, such as Agent Desktop, to record on demand.
- A call recording can be started while supervisor monitoring is enabled.
- Real-time control of the call recording: The recording can be paused and resumed on demand by the agent or by the workflow when the customer provides sensitive data such as a PIN. This is the same functionality as Dynamic Recording.

3.3.1.2 Interaction Acquisition – GIA only

To perform analytics on the voice and text interactions, these interactions must first be acquired from third party platforms and formatted in such a manner so as to allow processing by the analytics engine. This function is carried out by the UConnector and Fetcher components.

The UConnector performs the following functions:

- Queries the third party recording platform for new interactions
- Collects the interaction contents (e.g., audio or text) and constructs a related metadata XML file that contains relevant details on the interaction such as the agent ID, call IDs, case ID, etc.
- Prepares audio (e.g., transcodes) for processing by the analytics engine
- Stores all collected information into configured input folders

The Fetcher performs the following functions:

- Queries the input folders for new interactions to process
- Updates the database with relevant interaction details
- Compresses and stores the audio for long term storage and playback

3.3.2 Analytics Processing – GIA Only

The analytics processing carries out the following functions:

- Uses Nuance Recognizer to find the utterances and phrases as outlined by the system configuration and updates the database
- Performs a speech to text transcription of the audio between found utterances and phrases within the audio interaction and updates the database
- Indexes the text transcripts to allow for ad hoc searches of the text as well as discovery analytics around trending word clusters and word cloud representation of the data
- Categorizes the interactions based on the found utterances, call properties, meta data and non-linguistic characteristics such as silence, music and tones

3.3.3 Configuration Tools

The GIR solution uses Genesys Administrator and Genesys Administrator Extensions (GAX) to provide user friendly interfaces that perform complex operations such as configuration, maintenance and administration of the contact center environment. Specifically, for GIR GAX:

- Requires the GIR plugin
- Provides ability to provision the IVR profile for recording
- Manages encryption keys
- Adds GIR roles, access groups and alarm conditions.

The GIA solution contains two tools for configuration:

1. SMConfig is a Windows thick client application that allows configuration of the sites, machines and processes which are part of the solution as well as the audio and index specifics. This tool also allows starting and stopping of services
2. SMART is a Windows thick client application that provides functionality for configuring the phrases, topics, programs and categories used by the analytics processing components.

3.3.4 User Interface and Reporting

The main business user interface used by customers is the web based SpeechMiner user interface. This user interface provides the following functionality:

- Reporting dashboards that show drill down reports on categories/KPIs
- Search, Trending and Discovery tools
- Reporting configuration
- Quality Management and Coaching functionality
- System Administration tools
- Users, Groups and Roles configuration

3.4 Standard Use Cases

At the time of writing this blueprint architecture for GIR and GIA there are no standard use cases available. When use cases have applicable content for GIR and/or GIA then this document will need to be updated to address these use cases.

3.5 Component View

The Component View describes the higher-level modules that make up the solution.

The following table lists the Genesys components that make up the Genesys Interaction Recording (GIR) and Genesys Interaction Analytics (GIA). Optional components are noted in the table.

Category	Component	Version	Notes
Genesys Interaction Recording (GIR)			
Interaction Acquisition	Recording Processor	8.5.2+	
	Recording Crypto Server	8.5.2+	
	Screen Recording Client	8.5.2+	Optional for CDR reports
	ICON	8.1+	
Configuration Tools	GIR Plug-in for GAX	8.5.2+	
Genesys Interaction Analytics (GIA)			

Category	Component	Version	Notes
Interaction Acquisition	UConnector	8.5.2+	
	Fetcher	8.5.2+	
Recognition Server	Recognizer	8.5.2+	
	Indexer	8.5.2+	
	Categorizer	8.5.2+	
Configuration Tools	SMConfig	8.5.2+	Windows thick client
	SMART	8.5.2+	Windows thick client
Common across Genesys Interaction Analytics (GIA) and Genesys Interaction Recording (GIR)			
Web User Interface	SpeechMiner Web	8.5.2+	IIS

Table 1 - Genesys Component List

3.5.1 Required Genesys Core Services

The following is a list of the core Genesys components required with the Genesys Interaction Recording (GIR) solution. Genesys Interaction Analytics when deployed in isolation does not require any of these components unless paired with a GIR deployment.

Category	Component	Version	Notes
SIP Platform	SIP Server	8.1.1+	
	SIP Feature Server	8.1.2+	Optional
	SIP Proxy	8.1.1+	Optional HA component
eServices	Interaction Server	8.5+	Optional for non-voice interactions
Media Server	Genesys Voice Platform	8.5+	Optional in place of Media Server
	Genesys Media Server	8.5+	Optional in place of GVP
	GVP Reporting Server	8.5+	Optional
Routing	Orchestration Server	8.1.3+	Required for selective recording
	Universal Routing Server	8.1.2+	Required for selective recording
	StatServer	8.1.2+	
Administration	Genesys Administrator	8.1.3+	
	GAX	8.1.4+	
Configuration	Configuration Server	8.1.3+	
	License Manager (FlexLM)	11.9+	
	Local Control Agent (LCA)	8.1.3+	
	Message Server	8.1.3+	
	SNMP Master Agent	8.1.3+	Optional
	Solution Control Server	8.1.3+	
Database	DBServer	8.1.3+	
Common	Genesys Web Services	8.5+	

Table 2 - Genesys Component List

3.5.2 3rd Party Components

The following table lists the recommended 3rd party components for this solution. Alternatives are also noted though the recommended components are encouraged.

Component	Recommended	Version	Note
Genesys Interaction Analytics (GIA)			
Recording Platform	Genesys Interaction Recording (GIR) Alternatives: Verint/Witness, Nice, CallCopy, Cisco MediaSense, Autonomy/Qfinity, Audiolog, VPI, I3, Calabrio, Envision		Note that GIA has a tighter integration with GIR and that more advanced functionality is available in such a configuration. If the customer has no incumbent Recording Platform then we should recommend GIR.
Genesys Interaction Recording (GIR)			
Session Border Controller	eSBC (OEM from AudioCodes)	6.8	Other alternatives include SBC from AudioCodes, ACME Packet
Media Gateway	AudioCodes	6.6+	Sonus, Cisco
Operational Management	HP OpenView		Alternative includes OpenNMS
Phone Sets	Polycom VVX 3x0, 4x0, 500, 600 and VVX 1500 Polycom SoundPoint IP 335		Alternatives include Yealink and AudioCodes
SIP Soft Clients	Genesys SIP Endpoint, Counterpath Bria		Note that the recommendation is to use the Genesys SIP Client instead of Bria. Consider Bria and alternative.
Common across Genesys Interaction Analytics (GIA) and Genesys Interaction Recording (GIR)			
Web Application Server	Microsoft IIS		
	Apache Tomcat 6.x	6.x	
File Server	NAS/SAN		

Virtualization	VMWare	5.1+	
Database	MS-SQL	2012	MS-SQL 2008 SP1 is also supported
	Cassandra	1.1.12+	

Table 3 - 3rd Party Components

Note that other databases such as those required by Management Framework are not listed. RDBMS is often a customer preference. Additional details of database considerations is covered in the Common Component blueprint architecture.

3.6 Limits and Constraints

3.6.1 Genesys Interaction Recording limits and constraints

- SIP in Front Recording for 3rd party Voice is an unsupported architecture and should not be considered a sellable solution
- Support for Skype for Business remains a development/roadmap item
- GIR SIPREC support is still a roadmap item, and is awaiting technical feasibility

3.6.2 Genesys Interaction Analytics limits and constraints

<insert GIA limitations>

4 Deployment View

4.1 Solution Deployments – GIA Only

4.1.1 GIA Centralized Deployment

The location of third party recording platforms and regulations around the transmission of recordings are key elements for consideration when deciding on a network topology for the solution. Ideally, all analytics processing occurs at a central location and is collocated with the database server. The following diagram is a sample network architecture for a centralized deployment.

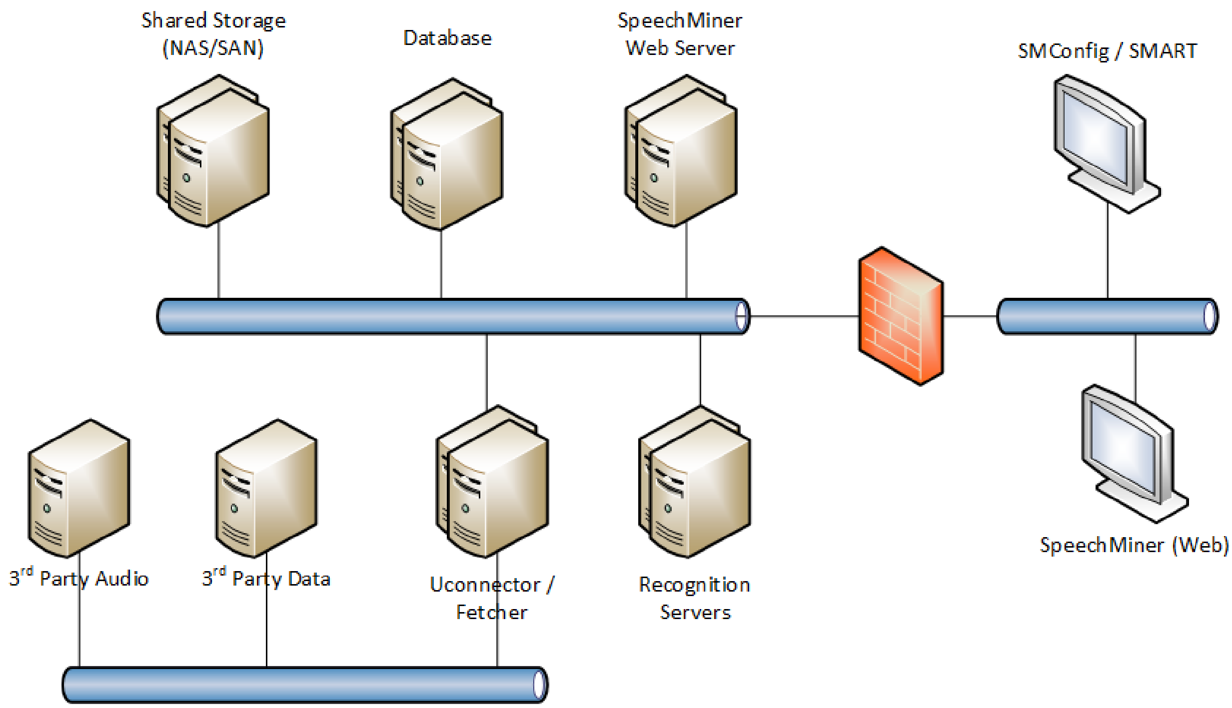


Figure 4 – GIA Central Deployment Model

The UConnector performs the following functions:

- Queries the third party recording platform for new interactions
- Collects the interaction contents (e.g., audio or text) and constructs a related metadata XML file that contains relevant details on the interaction such as the agent ID, call IDs, case ID, etc.
- Prepares audio (e.g., transcodes) for processing by the analytics engine
- Stores all collected information into configured input folders

The Fetcher performs the following functions:

- Queries the input folders for new interactions to process

- Updates the database with relevant interaction details
- Compresses and stores the audio for long term storage and playback

The SpeechMiner Web and Recognition Servers carry out the following functions:

- SpeechMiner uses Nuance Recognizer to find the utterances and phrases as outlined by the system configuration and updates the database
- Performs a speech to text transcription of the audio between found utterances and phrases within the audio interaction and updates the database
- Indexes the text transcripts to allow for ad hoc searches of the text as well as discovery analytics around trending word clusters and word cloud representation of the data
- Categorizes the interactions based on the found utterances, call properties, meta data and non-linguistic characteristics such as silence, music and tones

SpeechMiner Web interface provides the following functionality:

- Reporting dashboards that show drill down reports on categories/KPIs
- Search, Trending and Discovery tools
- Reporting configuration
- Quality Management and Coaching functionality
- System Administration tools
- Users, Groups and Roles configuration

SMConfig is a Windows thick client application that allows configuration of the sites, machines and processes which are part of the solution as well as the audio and index specifics. This tool also allows starting and stopping of services.

SMART is also a Windows thick client application that provides functionality for configuring the phrases, topics, programs and categories used to by the analytics processing components.

The following table lists the components that make up each of the nodes in the deployment model.

Node	Component	Comments
SpeechMiner Web	UPlatform	Configured for Web UI
	ULogger	
	Nuance License Server	Required on only one server
Recognition	Nuance Recognition Engine	
	UPlatform	Configured for recognition, categorization and indexing
	ULogger	
Fetcher / UConnector	UPlatform	Configured for fetching and compression
	UConnector	

	ULogger	
Database	MS SQL Server	With Reporting Services

Table 4 - Data Center Node Components

Please see section 6.1 GIR Solution Sizing Guidelines for details on the sizing of each VM node.

4.1.2 GIA High Availability Deployment

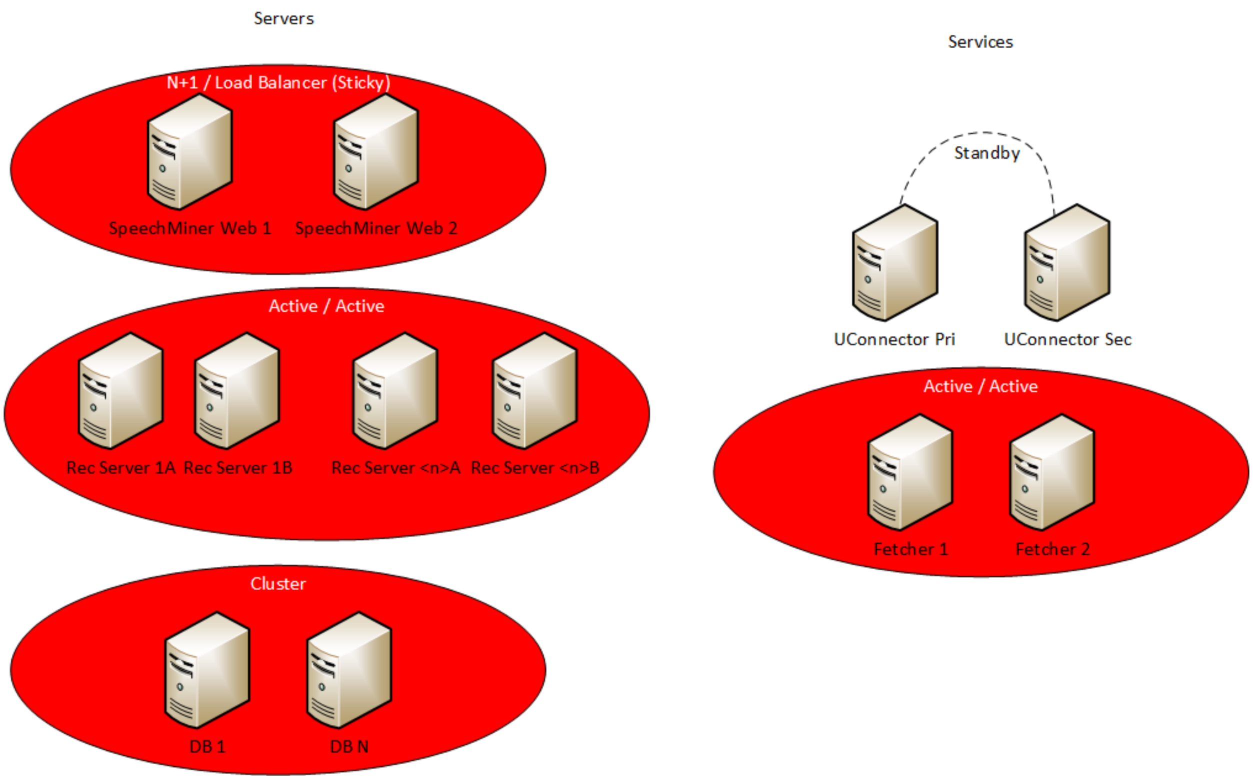


Figure 5 – GIA High Availability (HA)

To provide resiliency in both deployment modes, high availability (HA) options are recommended for all components. Similar approaches should be used to setup HA in both deployment modes.

The standard approach to HA in the GIA Solution is to provide an active-active HA pair of servers. This section outlines details on how HA is accomplished for each component within the solution.

4.1.2.1 UConnector HA

Each UConnector is slightly different based on the integration with the third party recording platform as well as the specifics of the customer environment. Typically, the UConnector is a .NET application that runs periodically to check for new recordings to import into the solution. It is possible to deploy multiple UConnectors within a deployment; however, the UConnector must be built to accommodate HA. Currently, no implementations with full HA support exist. Instead multiple UConnectors are deployed for scaling purposes.

4.1.2.2 Fetcher HA

Fetcher high availability can be achieved by running two Fetcher tasks on separate servers. Both can be active simultaneously with no negative impact on the system as synchronization of the tasks is managed through the centralized database.

4.1.2.3 Recognizer HA

Recognizer high availability can be achieved by running two Recognizer processes on separate servers. Both can be active simultaneously with no negative impact on the system as synchronization of the tasks is managed through the centralized database. Subsequently, the Recognizer can be scaled up from there based on the volume of recordings that need to be processed for analytics.

4.1.2.4 Indexer HA

Indexer high availability can be achieved by running two Indexer tasks on separate servers. Both can be active simultaneously with no negative impact on the system as synchronization of the tasks is managed through the centralized database. In general, no more than 2 Indexer instances are needed in a deployment.

4.1.2.5 Index HA

The Lucene Index, which is created and managed by the Indexer, can be configured to create a periodic backup. In the event that there is a failure in the primary index, then the Indexer can access the backup Index to ensure high availability.

4.1.2.6 Categorizer HA

Categorizer high availability can be achieved by running two Categorizer tasks on separate servers. Both can be active simultaneously with no negative impact on the system as synchronization of the tasks is managed through the centralized database.

4.1.2.7 Database HA

Each database vendor has various strategies for providing high availability for their database system. Customers may have their own setup that needs to be adhered to as well. Microsoft SQL Server has a number of high availability options; these should be explored with the customer to see which is the best fit.

4.1.2.8 Shared Storage HA

The shared storage should be RAID 10 or better to ensure high availability and redundancy in case of failure.

4.1.2.9 SpeechMiner Web Server HA

The SpeechMiner web user interface is deployed on Microsoft IIS. Multiple instances (i.e., two or more) of the web server should be deployed to provide high availability and scale. These web server instances should be behind a HTTP load balancer that supports sticky sessions. The load balancer would detect failures and ensure that requests are only routed to web servers that are functional.

4.1.3 GIA Solution Component Survivability Matrix

Component	HA	Notes
UConnector	Warm Standby	Multiple independent instances to spread load
Fetcher	Active-Active pair	Multiple pairs to spread load
Recognizer	Active-Active pair	Multiple pairs to spread load
Indexer	Active-Active pair	
Categorizer	Active-Active pair	
Database	Vendor specific	
Shared Storage	RAID 10 or better	
SpeechMiner UI	Multiple instances HTTP Load Balanced with Sticky Sessions	

Table 5 - GIA Solution Component Survivability Matrix

4.1.4 Dual Data Center/Geo Distribution

For GIA a Dual Data Centre deployment suits a Disaster Recovery scenario and requires that the customer have two major data center sites so that in the event that a catastrophic failure occurs, the other site has the resources to continue all functionality. To achieve disaster recovery, all persistent data must be replicated over to the disaster recovery site on an ongoing basis. This includes replicating database content as well as persistent audio data. Database replication is achieved through Microsoft replication capabilities. Replication of audio data is deployment specific.

In addition, the disaster recover site must have redundant hardware and software that remains dormant and is only activated in the event of a catastrophic failure. This needs to be accounted for during sizing as additional Nuance licenses may need to be purchased for the disaster recovery site.

Finally, all network addresses used should be fully qualified domain names (FQDNs) such that DNS resolvers can direct traffic to the disaster recovery site servers after service recovery.

The following diagram is an example architecture for GIA disaster recovery:

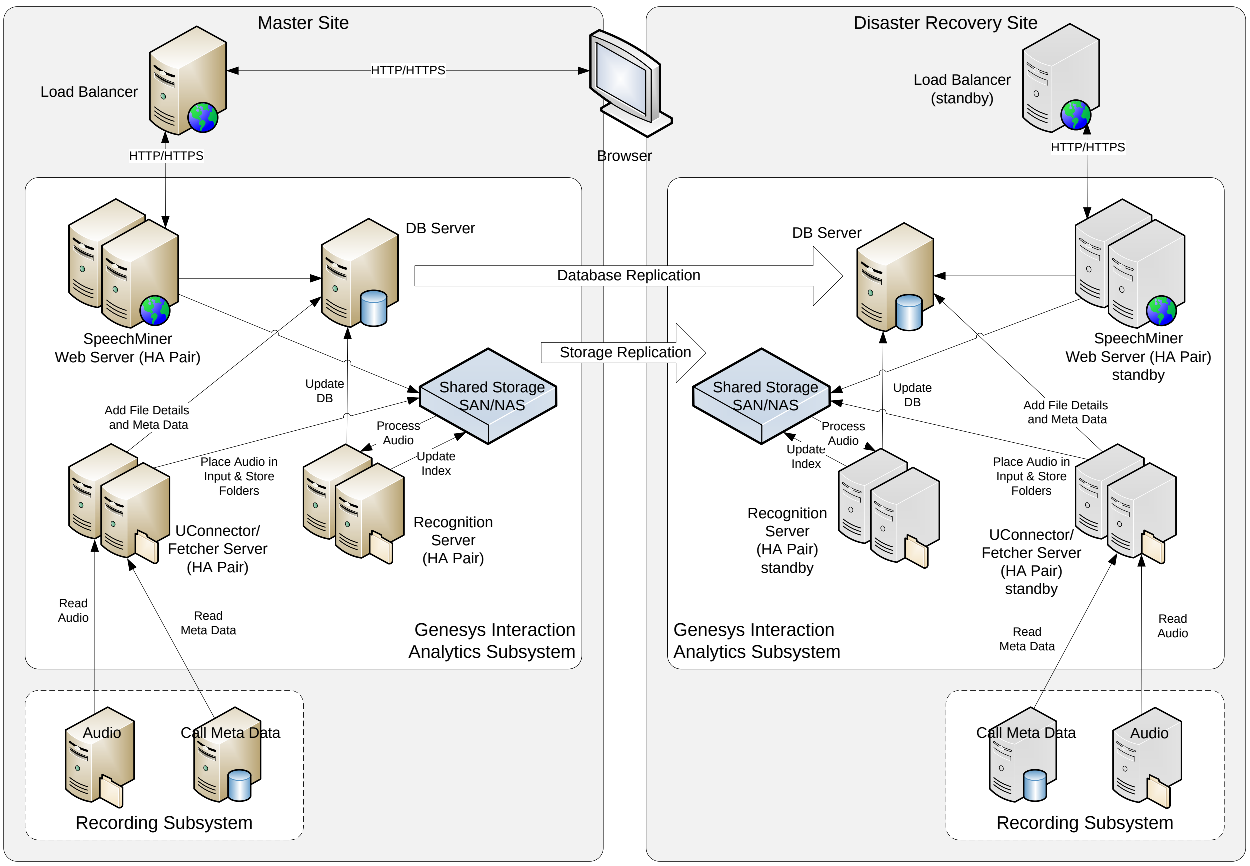


Figure 6 - GIA disaster recovery

4.1.4.1 Geographic Distribution of Solution

The multi-site distributed deployment processing deployment assumes that all recordings and metadata are analyzed by the Recognizer Servers at the remote site and that the results are simply updated to the centralized database server. This is not the preferred deployment mode for a large multi-site deployment; however, in cases where there are restrictions around the transmission of recordings, this is the only option. The following diagram depicts a multi-site deployment with high availability for all components and all processing carried out in a distributed fashion:

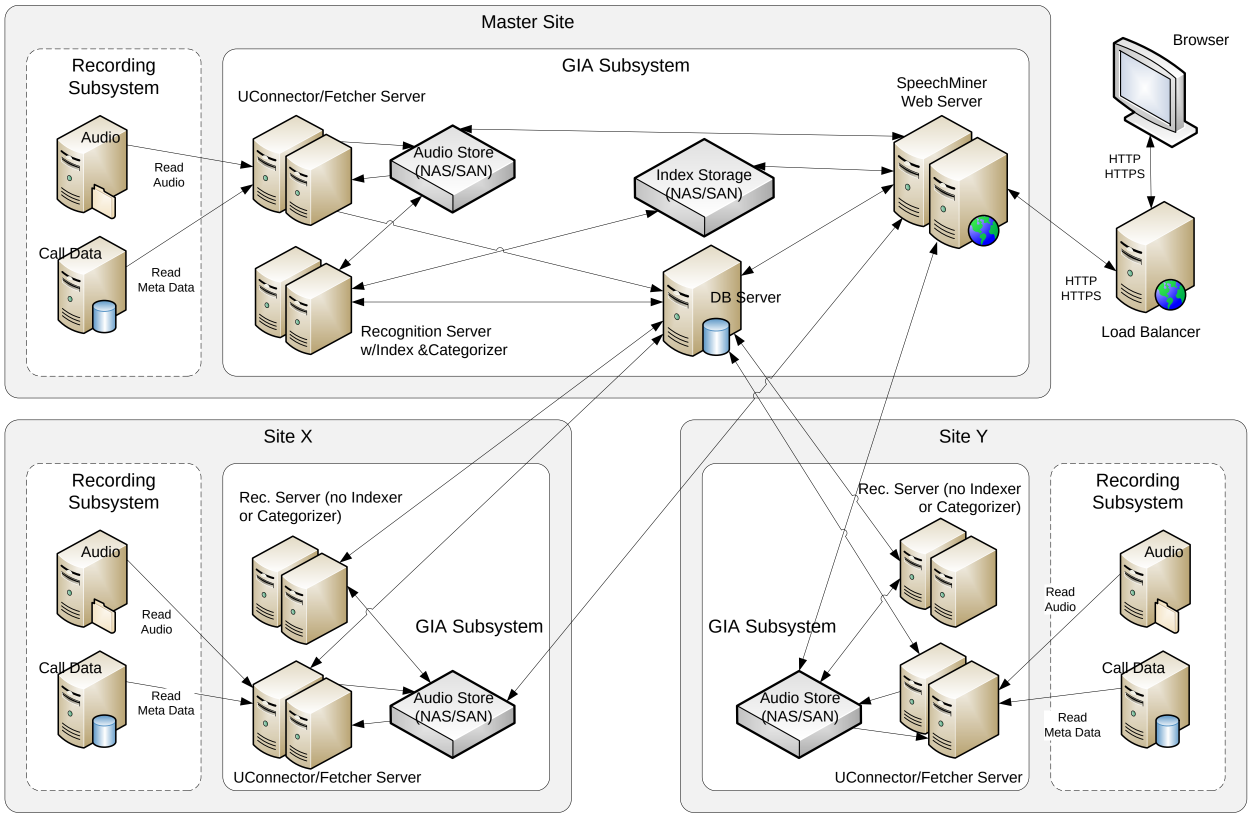


Figure 7 - Configuration Distribution

4.2 Solution Deployments – GIR Only

4.2.1 GIR High Availability Deployment

4.2.1.1 Scalability

SIP Server uses the same unified media server instances to provide any new media services, and in this case, the recording service.

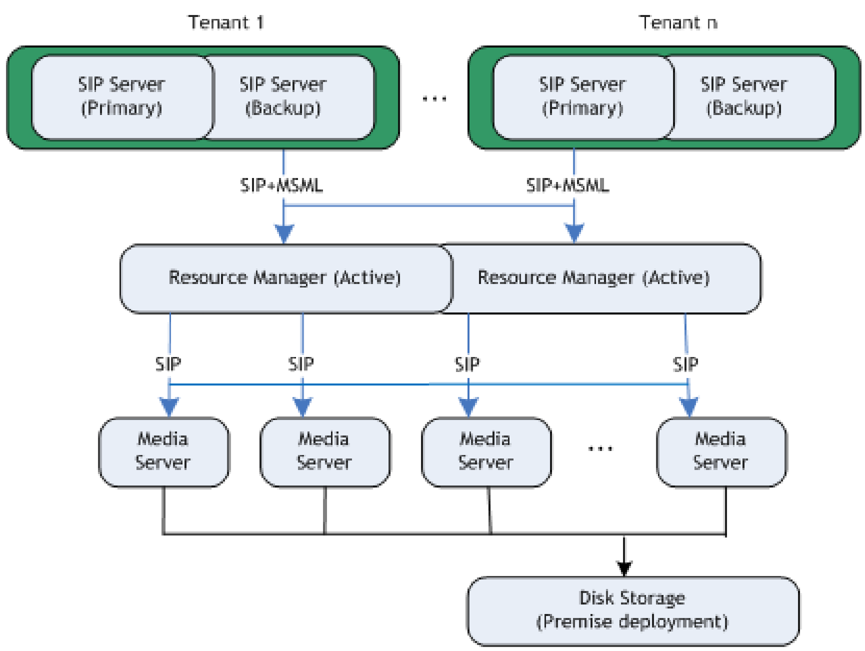


Figure 1: Scaling the Solution

A media server (MCP) cluster is managed by a pair of Resource Managers running in Active-Active mode. Resource Manager identifies the tenant coming from SIP Server and selects the appropriate IVR Profile for the call. Each MCP instance in the cluster is able to handle call recording for any tenant based on the IVR Profile information. MCP writes call recording to Amazon S3 or disk storage depending on the deployment type. Amazon S3 being a cloud service is already designed to scale. For disk storage, the premise deployment shall use a disk storage mechanism such as a disk array that can be scaled for the cluster of media servers.

4.2.1.2 High Availability

- **SIP Server -** SIP Server can be deployed in an active/hot-standby pair; whenever the primary instance fails, the hot-standby instance will take over and will have knowledge of all established sessions. If a recording session has been established with a recording server, a failure of SIP Server will not affect the operation of the communication and recording sessions.
- **Resource Manager -** Resource Manager (RM) is a SIP Proxy that can operate in either active/hot-standby or active/ active pair. When there is a failure at the Resource Manager, the remaining instance of Resource Manager will become active and continue to accept incoming requests. The remaining instance will remember the affinity of the recording sessions with Media Server as well as the recording server.
- **Media Control Platform -** During call recording the media is bridged through the Media Control Platform (MCP) and MCP becomes a single point of failure for the duration of the communication session. If MCP fails, Resource Manager notifies SIP Server about the failure so that SIP Server can take alternative action on the call. SIP Server first joins the endpoints together to ensure the communication session is not interrupted. SIP Server then attempts to record the call again by initiating a new recording session with the same parameters. RM most likely will select another MCP instance while the failed MCP is unavailable.
- **Web Services and Cassandra -** Web Services and Cassandra runs in an N+1 cluster.
- **Recording Processor -** The Recording Processor runs in both active-active, and active-backup modes.
- **Recording Crypto Server -** The Recording Crypto Server runs in active-active pair mode.
- **SpeechMiner** - SpeechMiner requires reliable shared storage that can be accessed (read/write) by all SpeechMiner components. The shared folders are mapped as an UNC path on the SpeechMiner machines and all SpeechMiner machines must refer to the same UNC path.
    - **SpeechMiner UI –** Deploy N+1 active instances
    - **SpeechMiner Interaction Receiver –** Deploy N+1 active instances.
    - **SpeechMiner UPlatform –** Deploy active/standby instances of the Indexer and categorizer task on UPlatform. When the active instance fails the standby instance automatically takes over to become the active instance.
- **HTTP Load Balancer:**
    - GIR components use HTTP for server to server communication
    - To provide high availability for the communication paths, HTTP load balancing is recommended to provide automatic failover for the components:
        - WebServices
        - Recording Processor
        - Recording Crypto Server
        - WebDAV
        - Interaction Receiver
    - To ensure high availability of the HTTP load balancer, GIR uses a virtual IP mapped to the load balancer host name.

4.2.1.2.1 High Availability for recorded interactions only

Under some scenarios, customers will want High Availability for components critical to the recording, storage and processing of calls but not for components used to access recorded interactions. This is usually done to reduce the server footprint and therefore TCO. To ensure no loss of interaction recording the following is the minimum list of components requiring high availability:

- Recording Processor
- Interaction Receiver
- WebDAV
- Database
- Storage
- Indexer
- Categorizer

4.2.2 GIR Solution Component Survivability Matrix

Component	HA	Notes
SIP Server	Hot-Standby	
Resource Manager	Active-Active	
Media Control Platform	N+1	
Genesys Web Services	N+1 Cluster	HTTP Load Balanced
Cassandra	N+1 Cluster	
Recording Processor	Active-Active or Active-Standby	HTTP Load Balanced
Crypto Server	Active-Active pair	HTTP Load Balanced
Interaction Receiver	N+1	HTTP Load Balanced
Indexer	Active-Active or Active-Standby	
Categorizer	Active-Active or Active-Standby	
Database	Vendor specific	
Shared Storage	RAID 10 or better	
SpeechMiner UI	Multiple instances HTTP Load Balanced with Sticky Sessions	
WebDAV	N+1	HTTP Load Balanced

Table 6 - GIR Solution Component Survivability Matrix

4.2.3 Dual Data Center/Geo Distribution

4.2.3.1 Multiple SIP Servers (MultiSite)

SIP Server to SIP Server Transfer/Conference

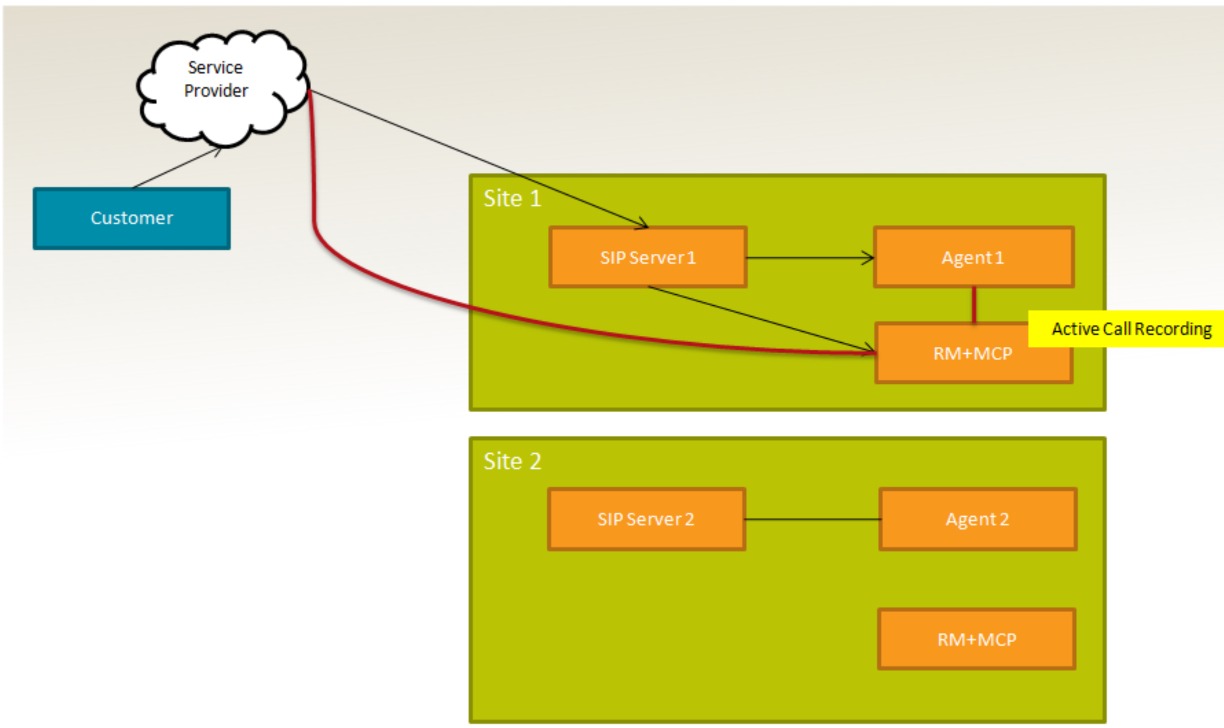


Figure 2: SIP Server to SIP Server Transfer and Conference

For premise deployments, multiple SIP Servers can be deployed in different locations. Each SIP Server queues the calls and distributes them to the agent. Recording is performed on the local SIP Server where the agent resides. If the agent transfer the call to another location, such as an agent group or routing point the call may then be distributed to an agent on another SIP Server. It is important to understand how call recording is performed under such as scenario.

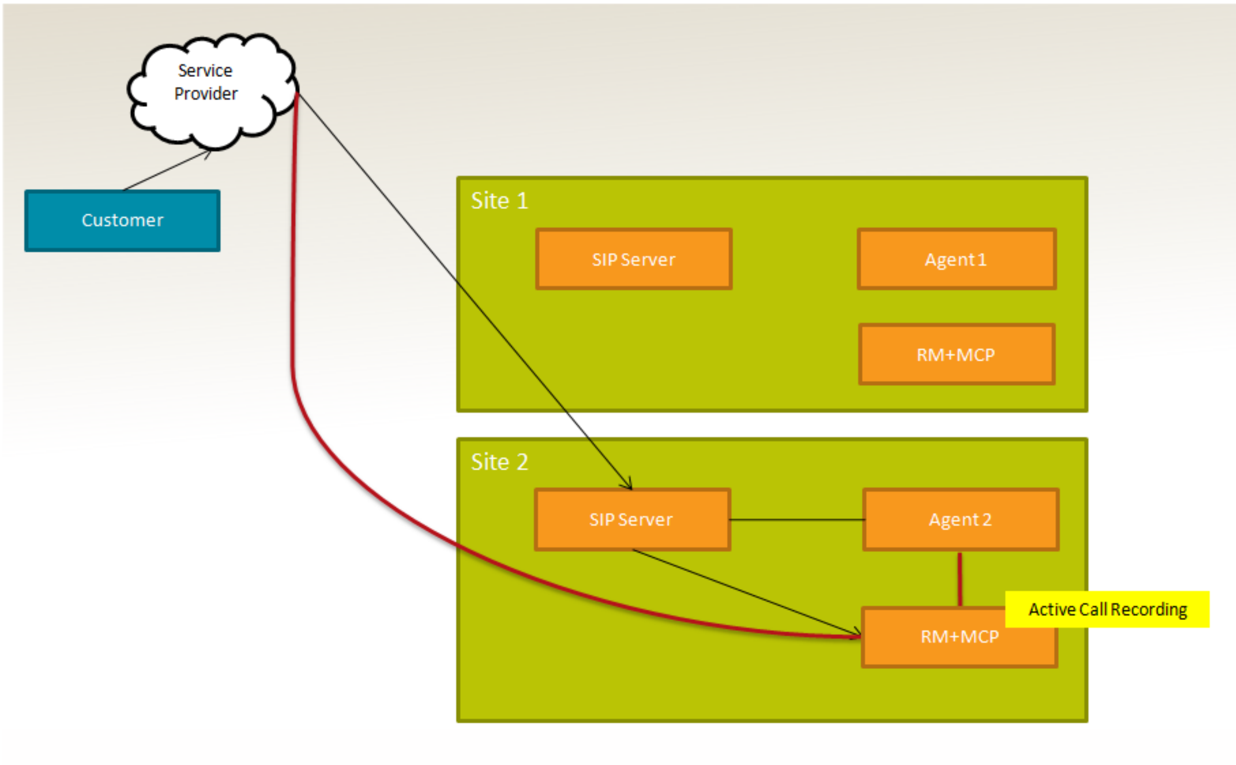


Figure 3: Single and Two-Step Transfer

Assume that Agent1 is being recorded for all transfer cases.

- **Single-Step Transfer** - When Agent1 transfers the call, the call is transferred to a Routing Point on local SIP Server1. The recording on Agent1 stops after the call is transferred. If Agent2 is configured to be recorded, the recording starts on Site 2. At the end of the call, there will be two recording files grouped together as a single recording.
- **Consult (Two-Step) Transfer** - When Agent1 transfers the call, the call is transferred to a Routing Point on local SIP Server1. The call is distributed to Agent2 on SIP Server2. The recording starts on SIP Server2 for Agent2 (Agent1 and Agent2). Agent1 completes the transfer and SIP Server1 is removed from the signaling loop. The recording continues on Agent2 (Customer and Agent 2) provided that Agent 2 is configured to be recorded.

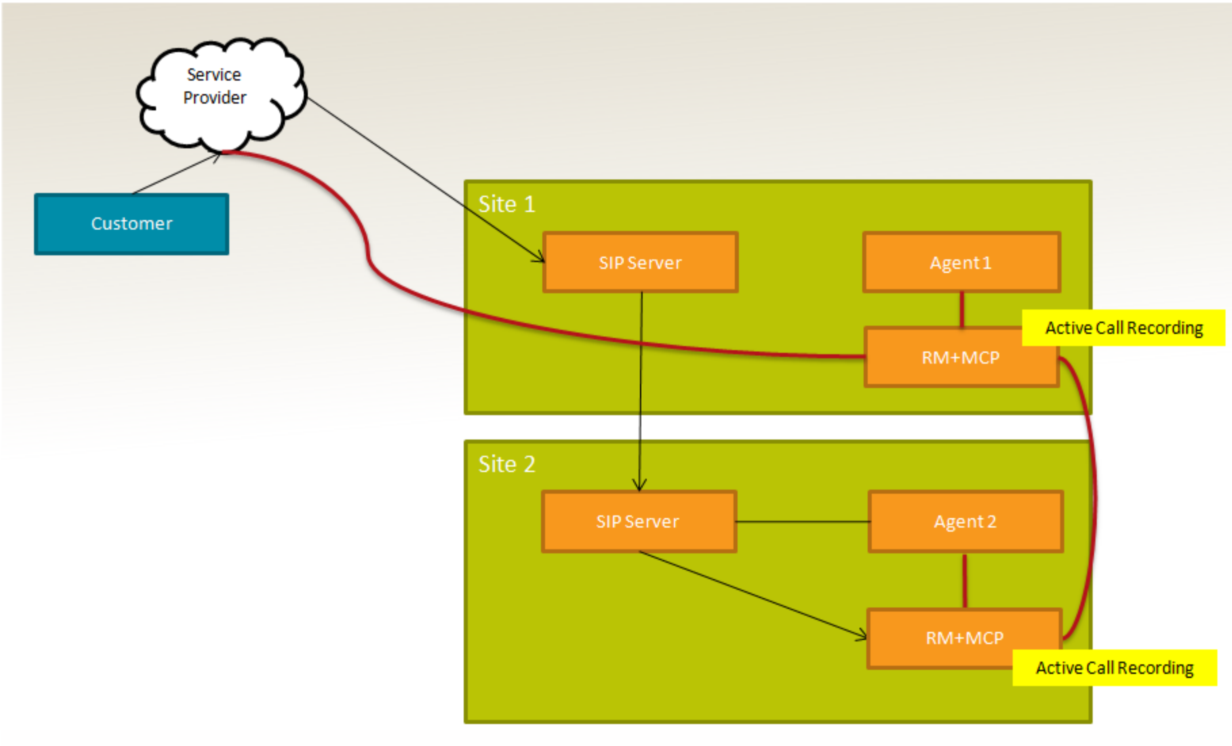
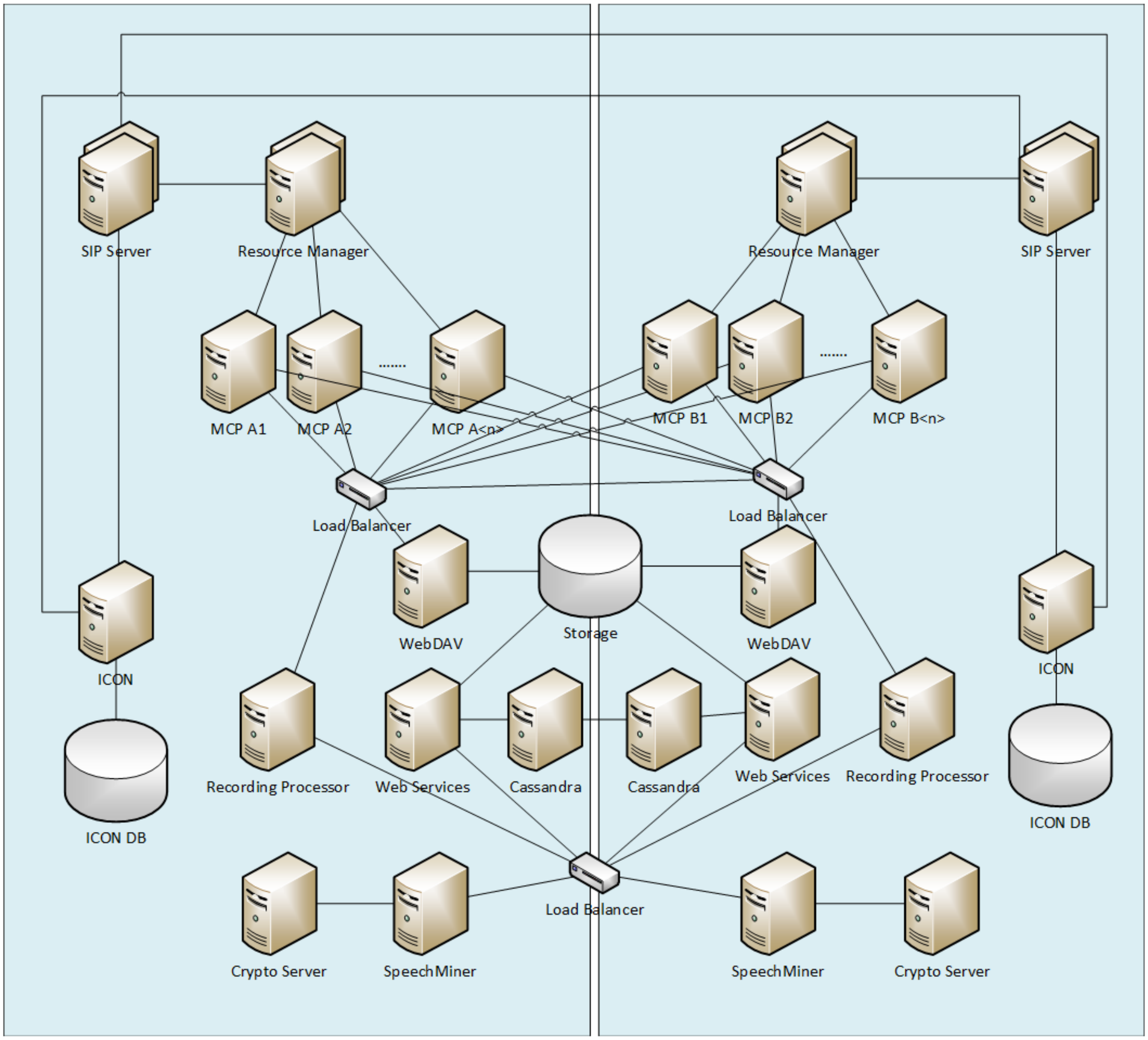


Figure 4: Single and Two-Step Conferencing

- **Single-Step Conference -** When Agent1 conferences a call, the call is transferred to a Routing Point on local SIP Server1. The call is distributed to Agent2 on SIP Server2. The call is conferenced through Media Control Platform (MCP) on SIP Server1. Recording starts on SIP Server2 for Agent2 (Customer, Agent1, and Agent2). Recording on Agent1 remains on SIP Server1 (Customer, Agent1, and Agent2). The recording remains on Agent2 on SIP Server2 (Customer and Agent2).
- **Two-Step Conference -** The end result is similar to the two-step transfer. The recording for Agent2 remains on SIP Server2 after Agent 1 drops out of the call (Customer and Agent2).

4.2.3.2 GIR Dual Data Centre Deployment



When implementing within a dual data center environment consideration needs to be made around the HA configuration of the GIR solution and location of the components across multiple data centers. The end architecture will need to take into account the quality of the link between the two data centers as this may drive the architecture to have HA within each data center as opposed to the HA across the data centers as shown in the diagram above.

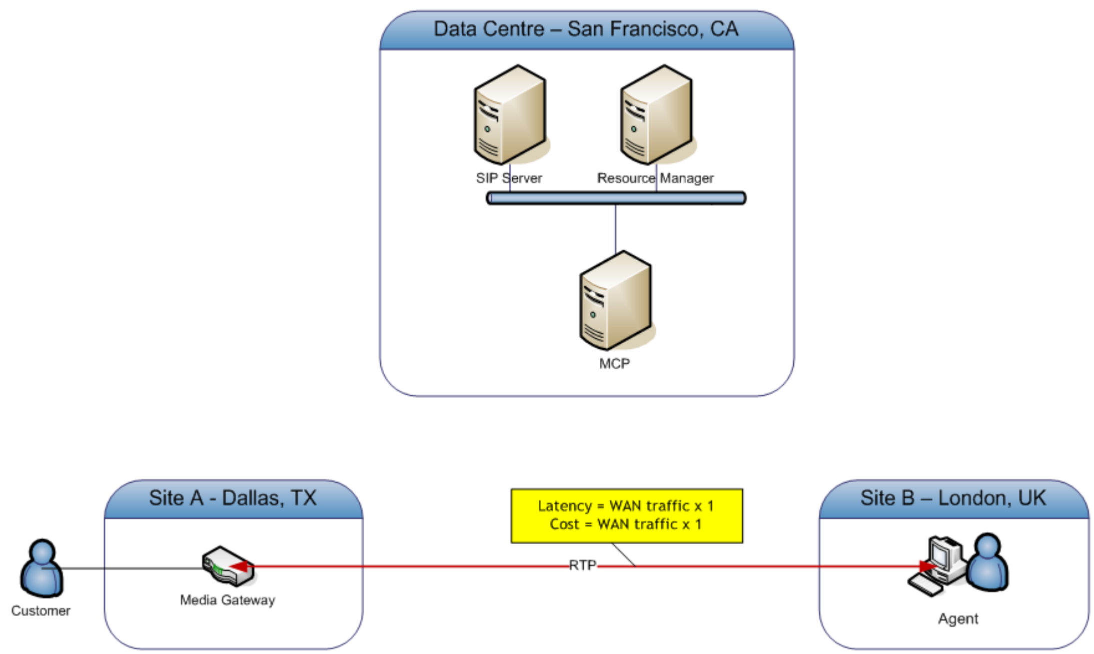
The diagram above has made assumption around the SIP architecture. Further details on the SIP architecture are provided in the SIP Voice blueprint.

4.2.3.3 GIR Geographic Distribution

The Genesys Interaction Recording solution uses geo-location to provide a multi-site deployment with the capability to select specific pools of Media Servers and recording servers that are located at specific sites. The main motivation for selecting specific Media Servers is either to minimize WAN traffic or to minimize the latency introduced to a conversation when call recording is enabled.

- No **Recording**

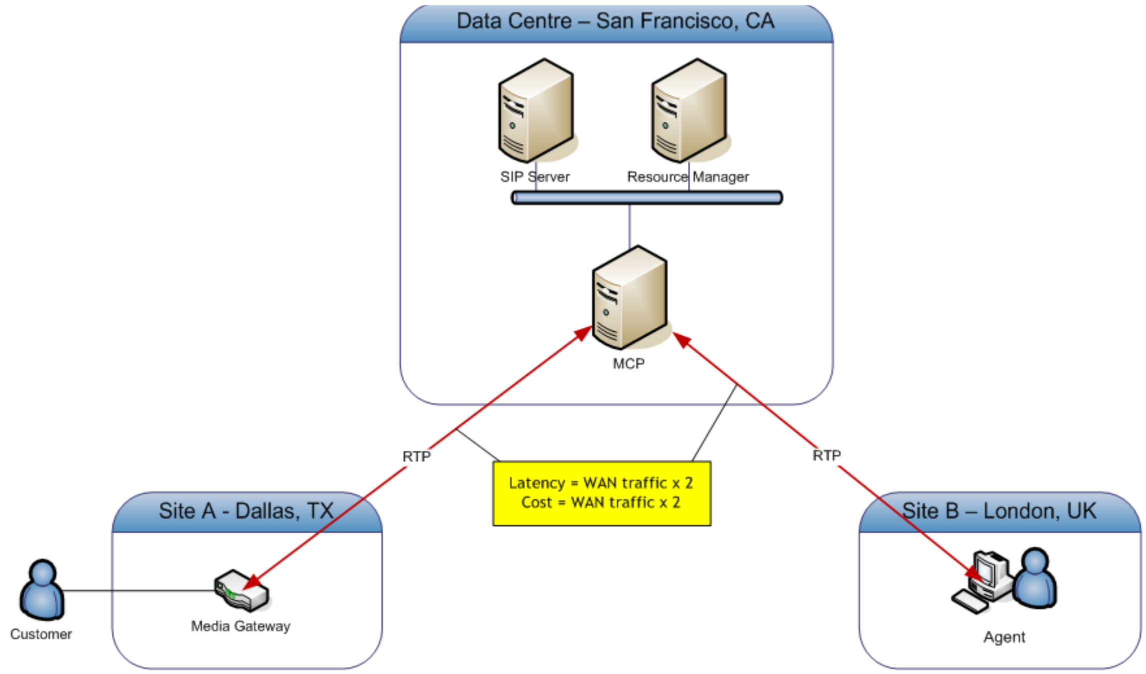
Figure 5: Customer and Agent Call Across the WAN with No Recording



In a typical scenario, the customer might be calling into a contact center site with a media gateway, and the agent is located in a different site.

- Recording in Data Center

Figure 6: Customer and Agent Call Across the WAN with Recording in Data Center

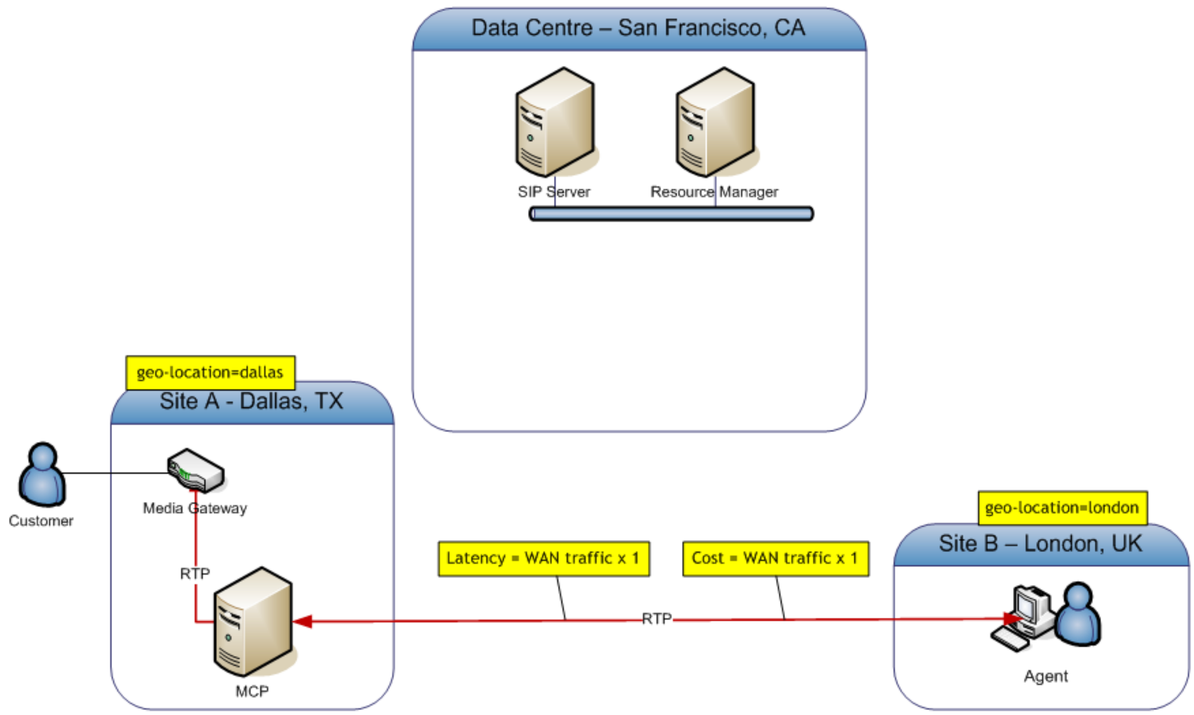


When the Media Control Platform (MCP) is located at the data center site, the deployment needs to double the WAN traffic because the media path needs to be bridged through the data center. This increased WAN traffic leads to increased latency of the media path by doubling the WAN path.

- **Recording with Geo-location**

To minimize latency, the geo-location feature has been introduced in SIP Server and Resource Manager. This feature allows MCPs to be deployed in a remote site that is close to one of the parties in the call. This diagram is a deployment that places MCP in Dallas as set in the *geo-location=dallas* parameter.

Figure 7: Customer and Agent Call Across the WAN with Recording with Geo-location at the Customer Site



4.2.3.4 Multiple Storage Locations

Recording files may have separate storage locations based on the location of the agent handling the recording to meet compliance or bandwidth requirements between data centers.

For example:

- You have data centers within the country where the agent resides and all the servers are located in those data centers. To save network bandwidth across the data centers all recording files must be stored in the same location where the recordings occur. A recording file is only transported across the data center when a client requests it for a media file playback.
- You have multiple data centers in different countries where the agents handle calls. There are compliance requirements to ensure that calls arriving in the country are recorded and stored in that country's data center only, and the calls arriving in another country are recorded and stored in that country's data center only. Your deployment model has Web Services nodes installed in both country's data centers.

These examples require that Web Services nodes be located in each specific data center with its own local storage location. For those locations that do not have local storage configured, a centralized storage location can be used. Mapping storage location based on the Web Services nodes requires no provisioning of the location of the agent since the connection of the screen recording client to the Web Services nodes represent the actual location where the agent resides.

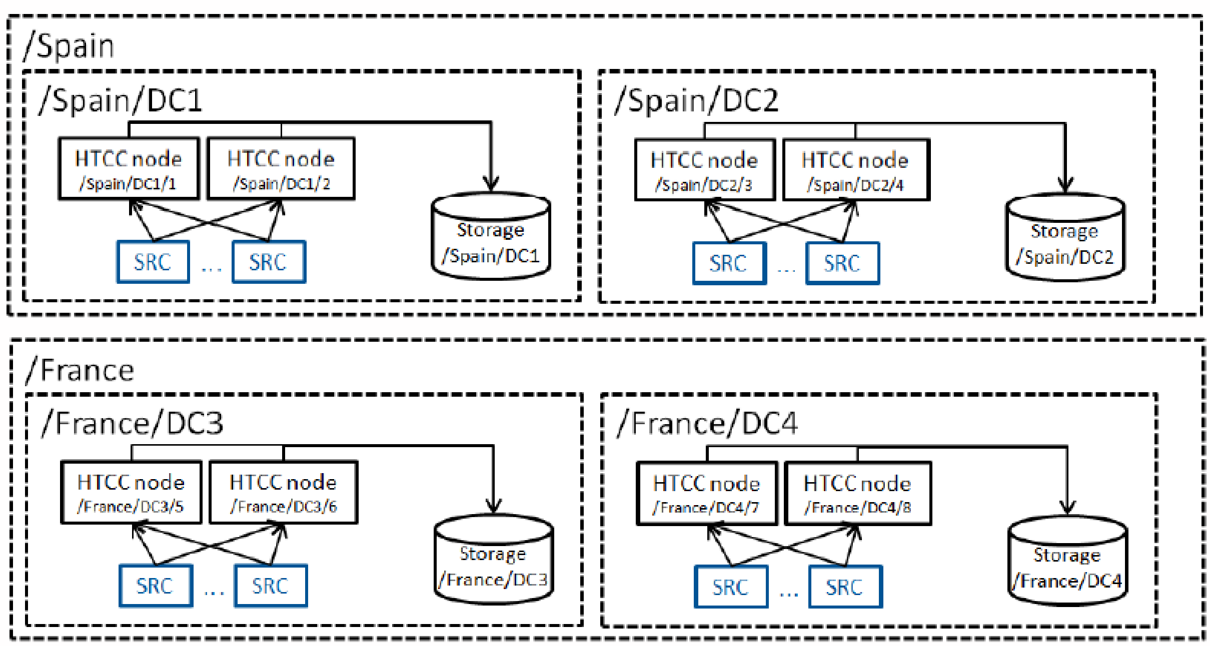


Figure 8: Storage at Multiple Locations

If the data center fails, the Screen Recording Client may be re-configured to connect to the Web Services nodes on a remote data center by changing the DNS mapping of Web Genesys Interaction Recording Services hosts to a remote data center. Recording files will be transported to a different data center and the recording will be stored locally by that data center.

4.3 Database Configuration

The RDBMS is a customer provided component of the solution and must be provided as part of the solution. Genesys-specific databases need to be setup within the database system and made accessible by the Genesys components. Follow the installation guides specific for each product and database vendor. Note that appropriate language/character sets need to be configured for some product databases.

Each database vendor has various strategies for providing high availability for their database system and customers may have their own setup which needs to be adhered to. Genesys should always be communicating with a single logical database.

To ensure site survivability the databases must be replicated to an alternative location such as the secondary data center. Business continuity is typically accomplished via some form of replication or clustering of databases.

The specific configuration such as transaction replication, batch replication, etc. will also define what is possible for the Recovery Point Objective (RPO) and Recovery Time Objective (RTO).

5 Interaction View

5.1 Interactions

5.1.1 GIA Only Interaction view

The diagram below describes internal message and data flow within the Genesys Interaction Analytics solution.

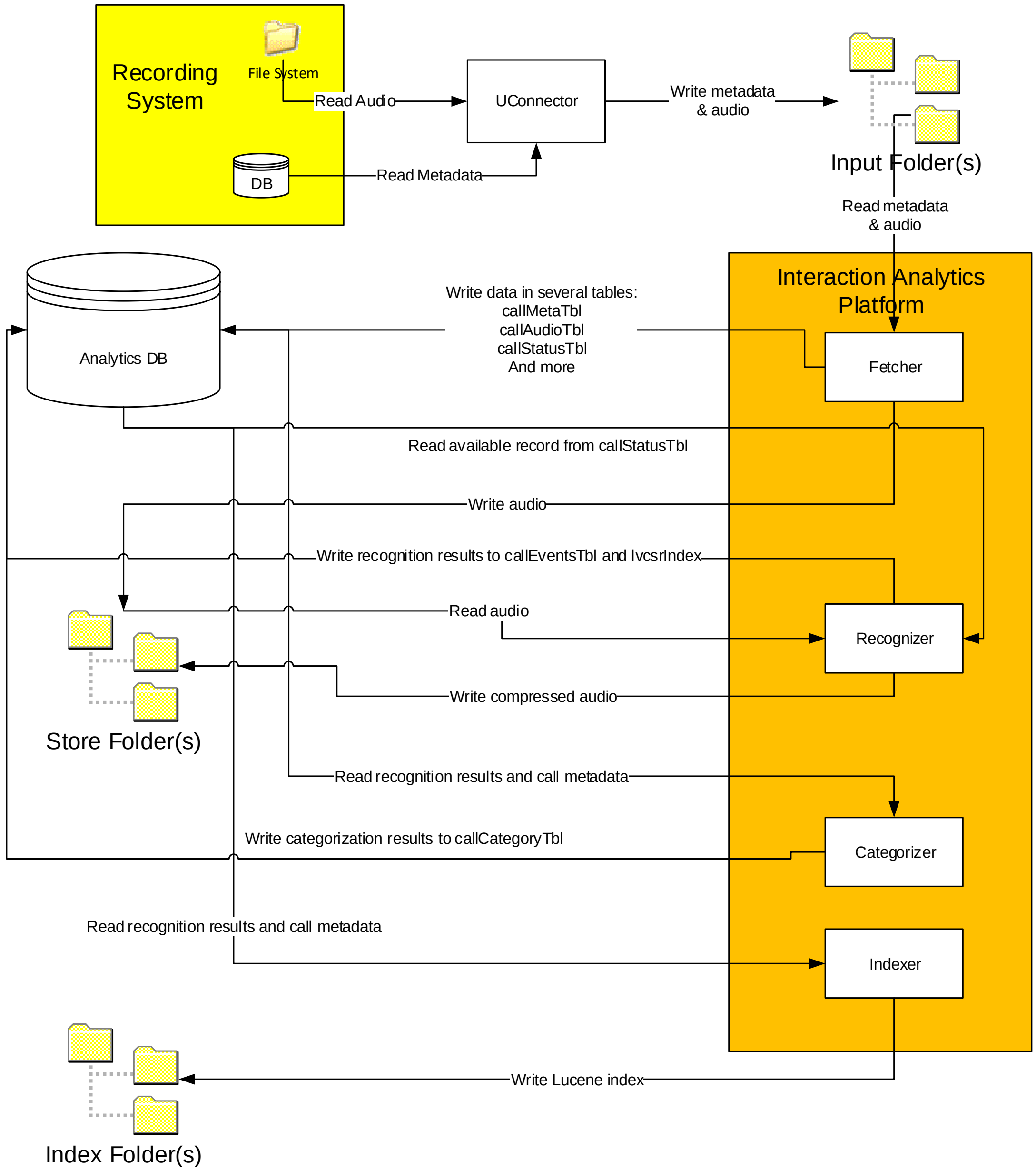


Figure 8 – GIA Interactions

5.1.2 GIR Only Interaction view

The following diagram illustrates the logical components of the GIR solution for a premise deployment:

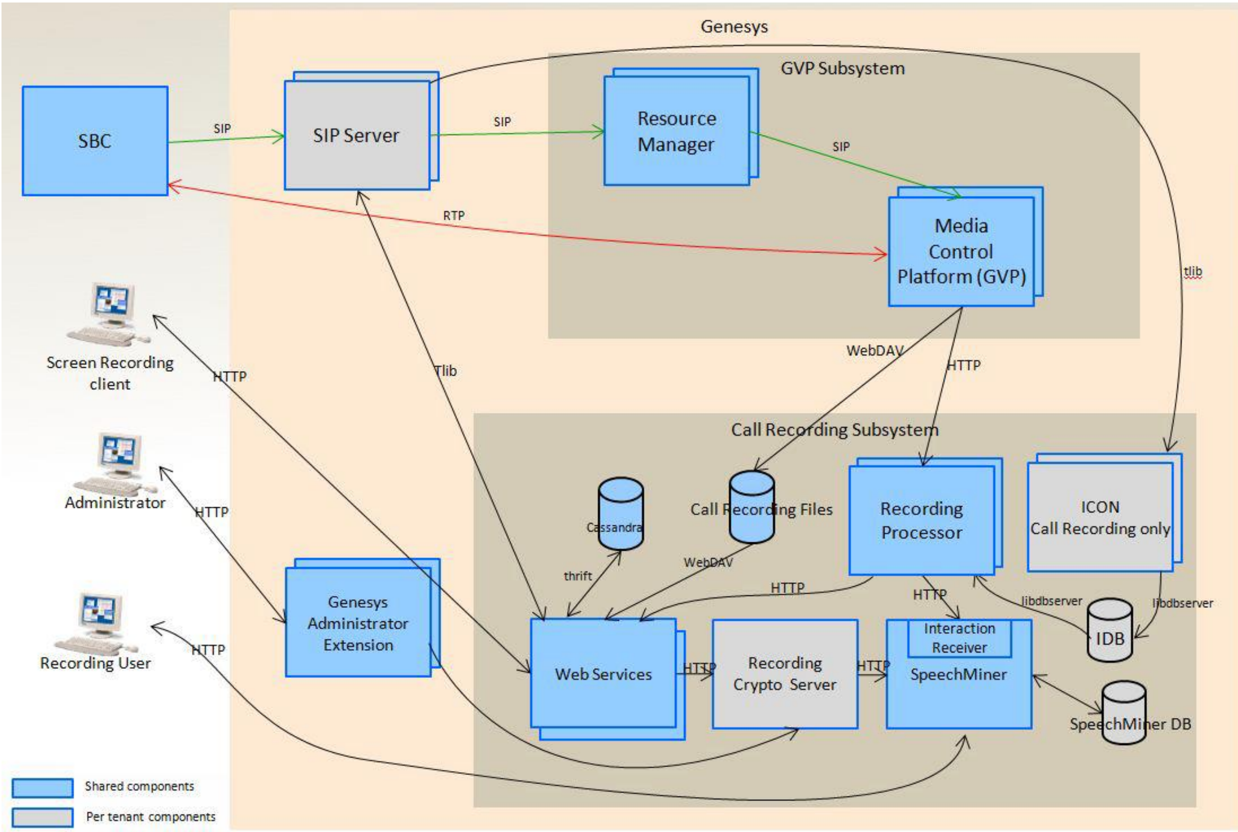


Figure 9: GIR Interactions

5.1.3 GIR Call Flows

- **Simplified Call and Data Flow**

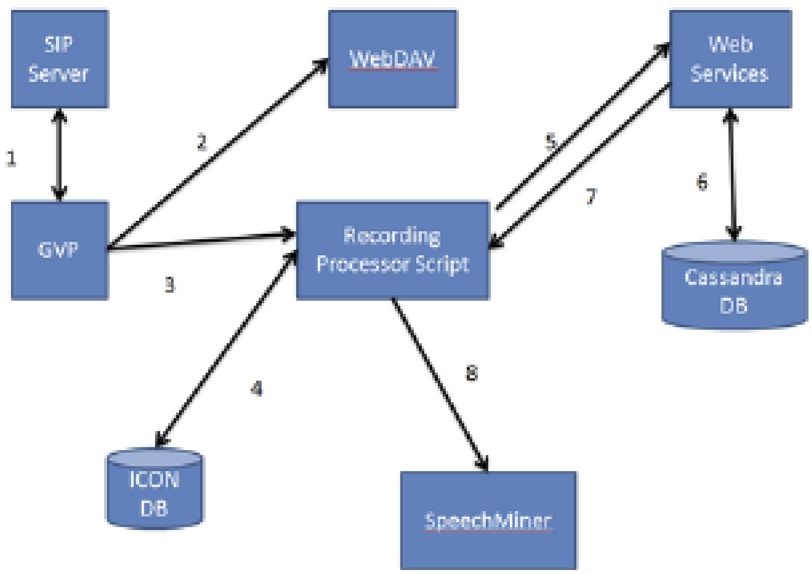


Figure 10: GIR Simplified Call and Data Flow

1. When a call enters the contact center, SIP Server sends the call to GVP (MCP&RM) to process the recording.
2. MCP, using an HTTP PUT, sends the mp3 recording to the WebDAV for storage.
3. When MCP receives a 200 OK from WebDAV, MCP, using an HTTP POST, send the recording's metadata information to the RPS.
4. RPS parses metadata received from the MCP and retrieves corresponding metadata from the ICON database that would have been provided by other possible SIP servers.
5. Once the metadata is retrieved from the ICON database successfully, RPS, using an HTTP POST, sends the information about the recording to the Web Services.
6. Web Services stores the recording information to the Cassandra database
7. Web Services tells RPS, with a 200 OK message, that the recording information is stored.
8. RPS send the recording to the SpeechMiner UI

- **Simplified End User Playback**

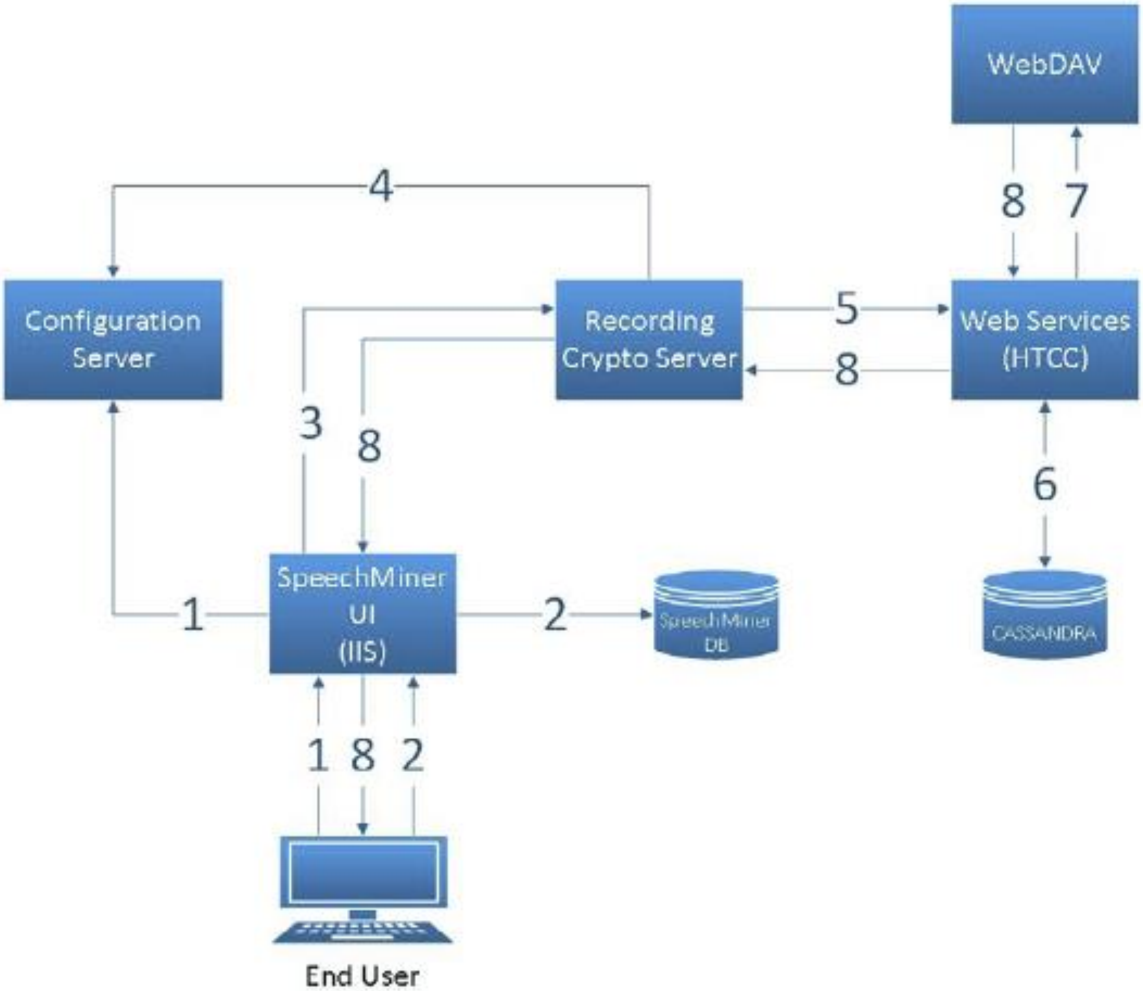
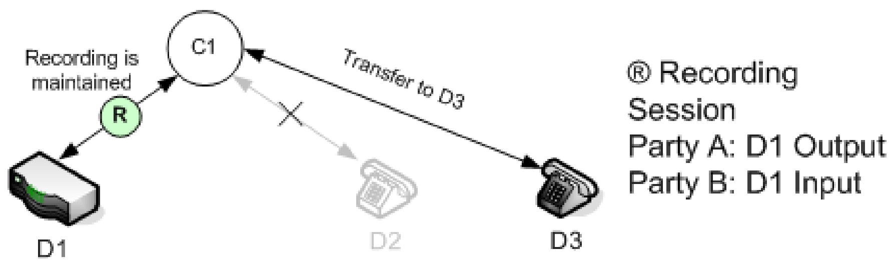


Figure 11: GIR Simplified Playback

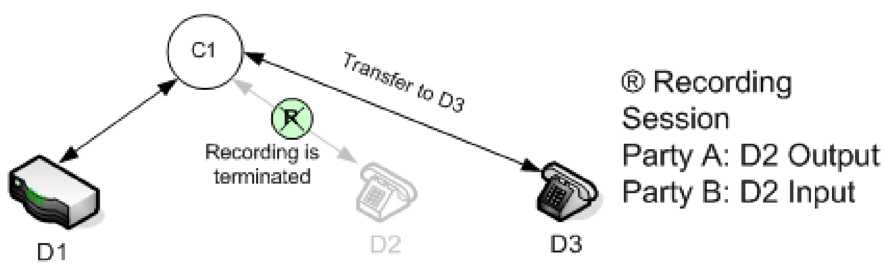
1. User logs into SpeechMiner UI, gets authenticated on Genesys Configuration Server with Roles/Permissions and Access Groups.
2. User searches for recordings based on attached data, UI searches data using the index and gets data from the SpeechMiner DB.
3. User selects a record for playback. Request is sent through RCS.
4. RCS verifies the user's login credentials with Genesys Configuration Server.
5. RCS fetches recording metadata from Web Services (HTCC).
6. Web Services check with Cassandra.
7. Web Services requests the recording file from WebDAV.
8. Recording is streamed from WebDAV back to the End User via HTCC, Recording Crypto Server and SpeechMiner UI.

Transfers

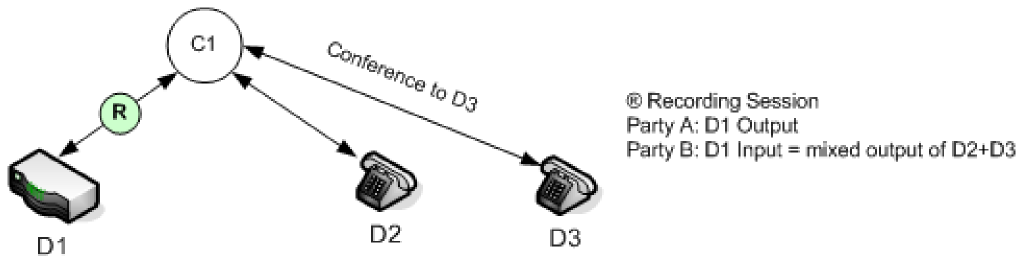
- **Origination: Recording from the Origination Device** - When a call that is being recorded is transferred to another party, the recording can be affected differently depending on where the recording is initiated. The reason for differentiating the originating side of the recording is that call recording is "sticky" on the side of the connection that is chosen for recording. When the connection needs to transfer the call to another device, the call recording stays with the origination device. For example, if the connection is transferred from D2 to D3, the call recording is maintained if recording is initiated from the origination device, while the recording is terminated if the recording is initiated from the terminating device.



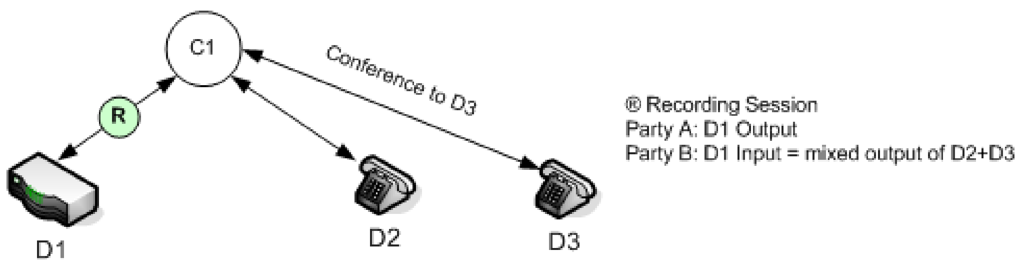
- **Termination: Recording from the Termination Device** - When the call is transferred to D3, the Recording Session is maintained and should expect a reINVITE to re-negotiate the media between D1 and D3. The media control dialog between SIP Server and Media Server is also maintained by only sending reINVITEs to the media control dialog.

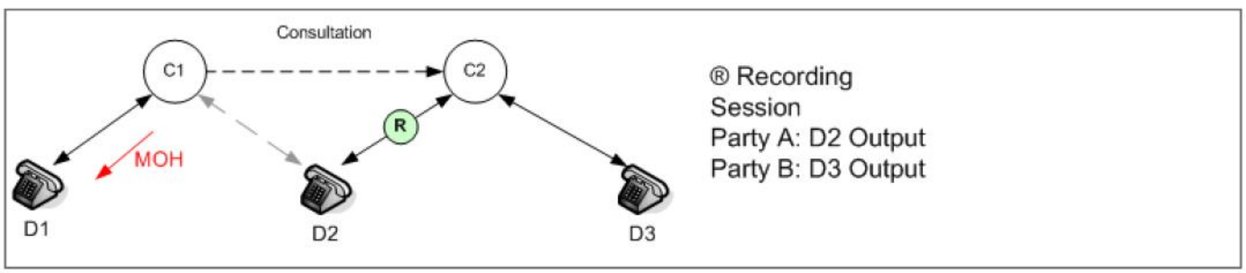


- **Conferences**
  - **Recording: Recording Conference** - The recording is a mixed output of the customer and agent, plus a separate stream of the supervisor.



- **SIP: SIP Dialog** - The structure of the SIP dialog when recording a conference.



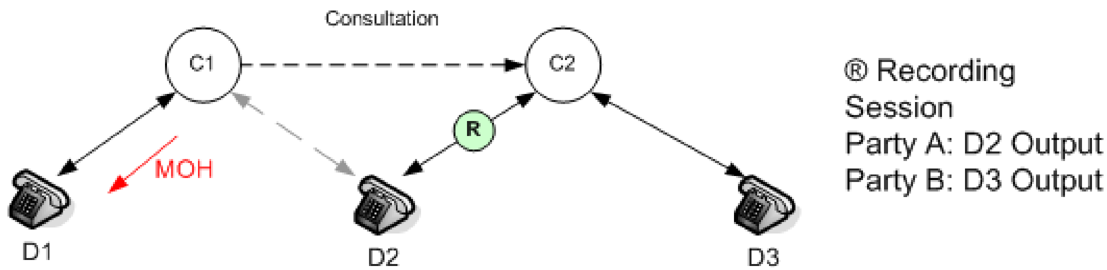


- **Consultations: Consulting Calls** - When the agent initiates a consultation call and call recording is enabled on the agent DN, the call recording to record the consultation session as well is allowed. This is recognized as a single-dialog consultation mode where there is only a single active SIP dialog on the device. The following diagrams illustrate this scenario.

    - **Before Consultation**

        The initial call when the customer (D1) is talking to the agent (D2). The call is being recorded during this conversation.

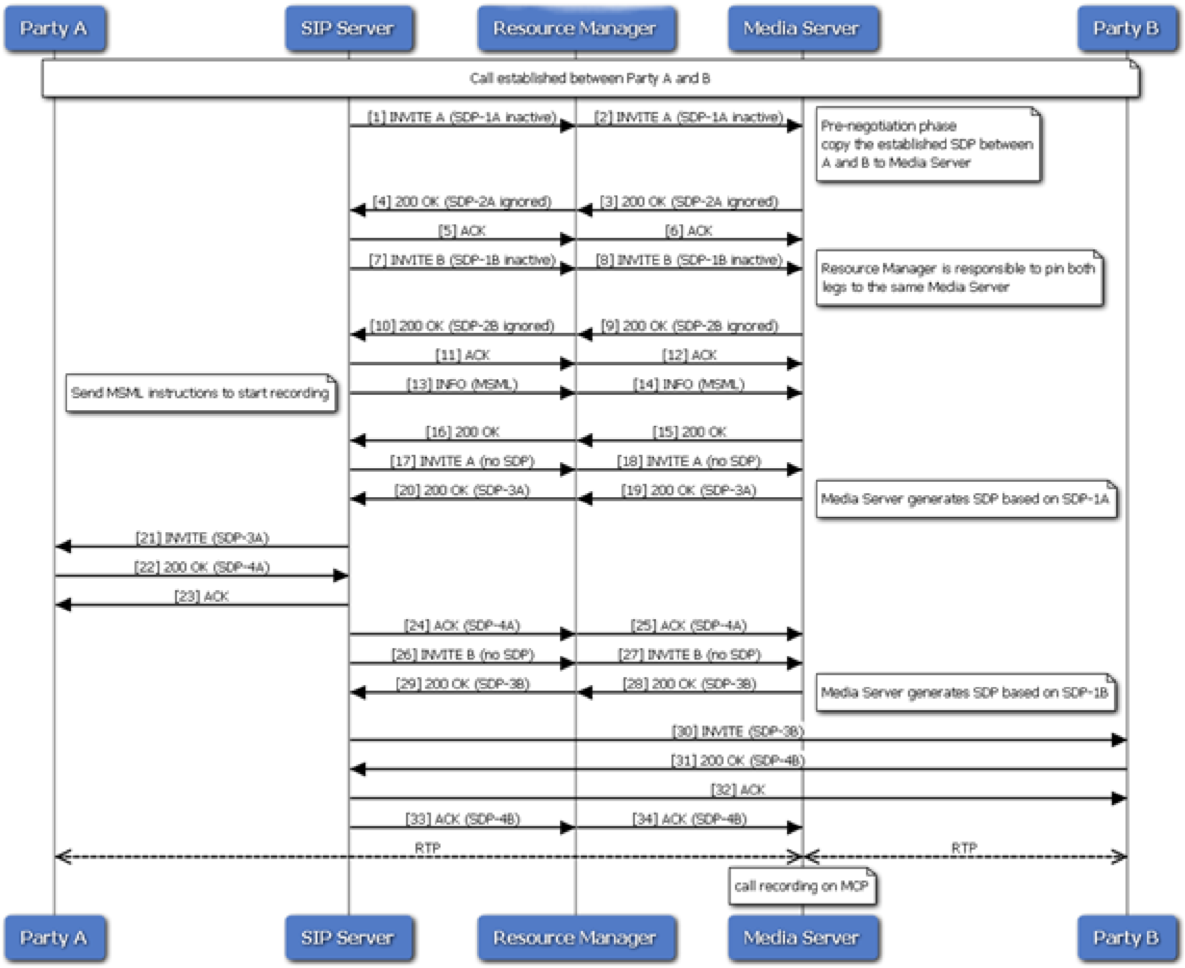
- **During Consultation**



When the agent initiates a consultation to the supervisor (D3), the existing SIP dialog is retained as is the Recording Session.

Important: As a current limitation for consultation calls, recording is not available on the consulted party, so a Recording Session cannot be started on D3.

- **Basic Call Flow**



After Party A and Party B are connected and a recording request is made to SIP Server, SIP Server initiates two sessions, one session for each party, to Media Server. SIP Server first INVITEs with the Session Description Protocol (SDP) offer from the connected parties to Media Server, and a second reINVITE to Media Server to get an SDP offer from Media Server. The offer from SIP Server is sent to the connected parties to proxy the media through Media Server. Once the media is established, Media Server bridges the media between the parties and writes the recording to a file on the disk. The Recording Server fetches and indexes the recording after the call completes.

5.2 Network Considerations

The customer's network infrastructure is a key element for consideration when integrating GIR and/or GIA into their environment. Networks at each customer's sites will be different and unique to their own requirements. In most cases the network will already be in place and the solution will need to contend with its idiosyncrasies. When implementing the solution take note of:

- The customers proposed vLans and the usage of these (e.g. Voice, Data, Management)
- The customers' security zones and any separation of services (e.g. databases existing within separate vLans)
- Bandwidth considerations (refer to the implementation section of this document)
- QoS marking and the honoring of these through-out the end to end solution
- Latency between components to ensure adequate service delivery

5.3 External Interfaces

This section describes the external interface for the solution. These become the integration points between solution components and the elements in the customers' premise.

5.3.1 GIA Only External Interfaces

This section describes the external interface for the solution. These become the integration points between solution components and the elements in the customers' premise.

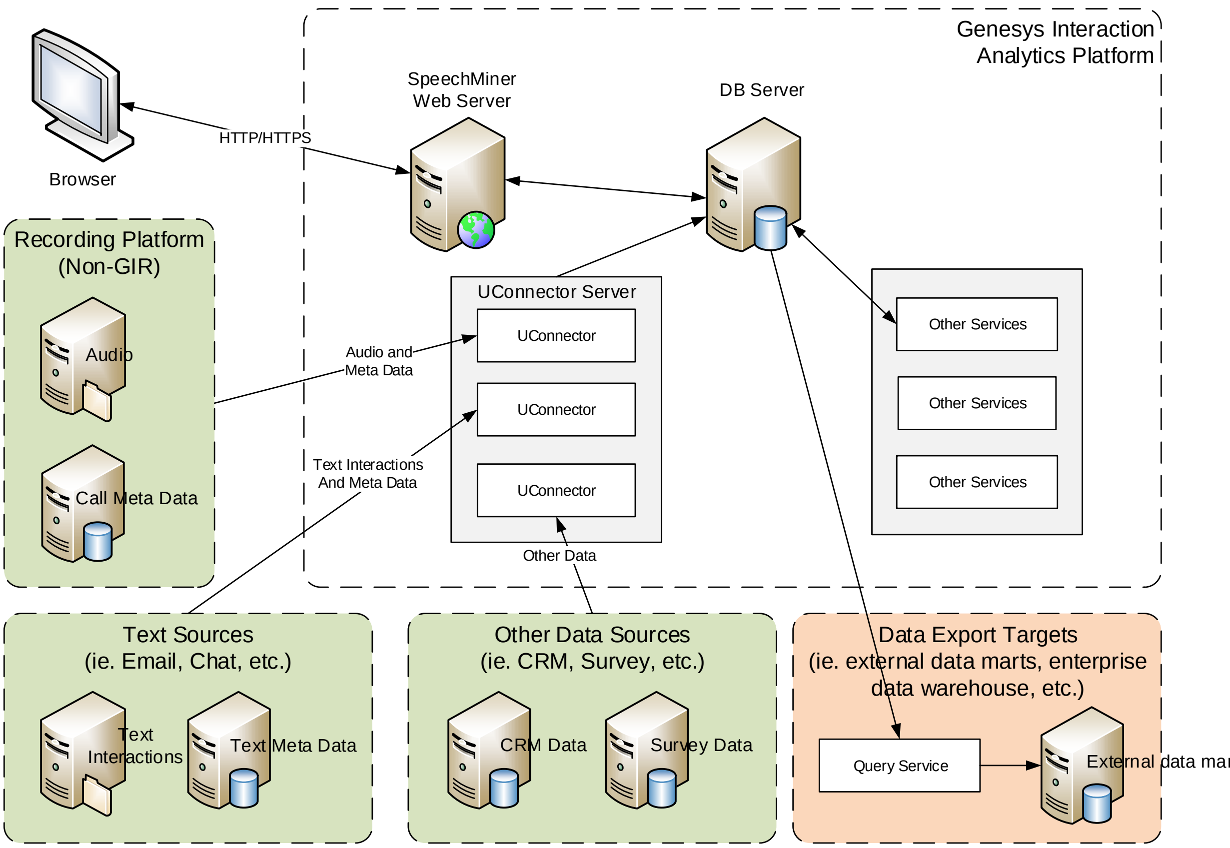


Figure 12: GIA External Interfaces Diagram

The following table details each of the external interfaces, its protocols, the components within the solution that are impacted or connected to these external interfaces and lists the integration tasks required to setup the external interfaces.

Interface	Protocol	Solution Components	Integration Tasks	Description
Browser	HTTP and HTTPS	GIA Web Server, Client Web Browser (IE 8+ or Chrome)	Main user portal for business user access to analytics dashboards, reports, and other tools provided by the web UI.	This interface is used to support normal day-to-day customer use of analytics solution to gain insights on their operation and to conduct various analytics reports and root-cause analysis.
SMART Tool	ODBC	SMART – Thick client windows application and DB Server	Tool for creating/managing topics, categories, and programs for analytics solution.	Used by business users of analytics solution to create and maintain their configuration of report data elements and business rules for the application. Tool is a windows thick client application.
Recording Platform (Non-GIR)	ODBC or via API supplied by recording vendor	Recording platform DB, filesystem, and (optional) export API/service. GIA UConnector service.	Import of call recordings and associated meta data from one or more (non-GIR) call recording platforms	Integration to call recording platforms occurs within a UConnector service. There are 2 primary ways of integration: 1) direct DB and filesystem query/extraction from recording platform, and 2) via recording vendor-supplied API/service.
Text Sources	Various (depending on availability of text source data)	Text source, UConnector service.	Import of text source interaction data and associated meta data from one or more text sources.	Integration of text sources occurs within a UConnector service. The integration mechanism will vary depending on what is available/provided by the text source. This can range from simple copy of text material/meta data from a file system to vendor-supplied integration API or direct DB query.

Interface	Protocol	Solution Components	Integration Tasks	Description
Other Data Sources	Various (depending on availability of other source data)	Other data source, UConnector service	Import of other source data from one or more sources.	Integration of other sources occurs within a UConnector service. The integration mechanism will vary depending on what is available/provided by the data source. This can range from simple copy of text material/meta data from a file system to vendor-supplied integration API or direct DB query.
Data Export Targets	Query of SQL views	GIA DB Server, SQL views, external query service	Export of data from GIA DB to other 3rd party applications and systems.	Analytics results – including results of categorization, topic detection, call transcriptions, etc. - can be exported for purposes of integration to 3rd party applications or systems. GIA provides SQL views for commonly requested data elements. External query services will query these SQL views to obtain data.

GIA External Interfaces List

As stated above, the primary interface for data input into the GIA solution is via the UConnector service. The UConnector is responsible for all communication, import, and any pre-processing or transcoding required – namely:

- Get call audio and metadata and put them into the input folder
- Update the agent/workgroup hierarchy
- Create user logins for agents

In some cases, the UConnector will also need to 'stitch' call segments together if call recordings exist in the source platform as multiple segments.

The UConnector is configured through XML files. Each XML file represents one recording system. That XML file holds all the information needed to extract the calls, e.g.:

- The recording system's database information
- File system where the actual recordings reside
- Start and end time of day when calls can be extracted
- SQL statement that specifies what calls will be extracted

- The program name
- Specifications for audio conversion and the path of the conversion tools

It is also possible to use a shared XML for parameters that are common to more than one system (use the GeneralXmlName element). Connection strings should be defined in the UConnector executable's .config file, and referenced by name. Connection strings can be encrypted but not all UConnectors currently use this scheme.

The UConnector is typically developed / customized by Genesys PS or engineering rather than customers as doing so requires access to GIA source code. As with the rest of the GIA platform, the UConnector is a .NET service. A separate UConnector is required for each data source.

5.3.2 GIR Only External Interfaces

The following table details each of the external interfaces, its protocols, the components within the solution that are impacted or connected to these external interfaces and lists the integration tasks required to setup the external interfaces.

Interface	Protocol	Solution Components	Integration Tasks	Description
Browser	HTTP and HTTPS	GIA Web Server, Client Web Browser (IE 8+ or Chrome)	Main user portal for business user access to analytics dashboards, reports, and other tools provided by the web UI.	This interface is used to support normal day-to-day customer use of analytics solution to gain insights on their operation and to conduct various analytics reports and root-cause analysis.
Web Services	HTTP and HTTPS	Genesys Web Services	A range of APIs are exposed and can be consumed externally by GWS that relates to the GIR platform.	GIR Specific APIs are: • Recording Processor Scripts • Recording Crypto Server • Recording Muxer Scripts • Screen Recording Services • SpeechMiner • Recording Cloud Backup Service • GAX & Recording Plugin for GAX • External API

GIR External Interfaces List

5.4 Operational Management

Once a Genesys solution is in place, managing the solution becomes a primary concern of the customer. There are two approaches to operational management that need to be considered for the solution.

1. If Genesys components are the main focus of the operation, then using Genesys Administrator and GAX becomes the primary mechanism for administering the solution.
2. If Genesys is part of a larger operation, then integration into the customer's operational management tool becomes advisable.

In both cases, Genesys Administration and GAX software need to be installed and configured to manage the solution.

<Call out any further specific items on Operational Management>

5.4.1 Network Management Systems

If the customer does have a Network Management System (NMS), then Genesys components need to be integrated into their NMS. This is typically done by setting up the SNMP Master Agent to send SNMP events and info to their NMS.

Examples of supportable NMS includes HP OpenView and OpenNMS (an open source NMS - http://www.opennms.org/).

5.4.2 Serviceability

Serviceability relates to the ability of technical support to identify issues and defects within the system. Most of this relates to the ability to retrieve logs and configuration information and pass them back to technical support.

Setting up logical logging locations is a best practice that can reduce the time to send logs to support. Configuring 3rd party components to log into the same location is ideal as well. Establishing a "log" directory in the root of the disk structure and logging there is recommended:

```
D:\GCTI\log
/log
```

At the time of this writing, technical support is building a tool that will aggregate and retrieve logs for analysis from a customer's environment. Once in place, this should be a standard part of the solution.

Proactive monitoring provides the most complete servicing of a customer's environment. This is provided through Premium Care and is beyond the scope of this document.

6 Implementation View

The Implementation View describes details such as sizing, security and configuration of the solution based on the previous deployment and interaction views.

6.1 GIR Solution Sizing Guidelines

This section provides guidelines on sizing the GIR solution components to determine the server requirements. Providing a simple and accurate sizing guideline is difficult as there are many variables, between the number of agents, the type of call flows, peak call volumes, network bandwidth, etc. that can occur within the customer's operations.

Please treat this as a rule of thumb. Changes to any variable can impact the overall sizing.

6.1.1 Solution Sizing

GIR sizing can be calculated via the GIR sizing tool
https://docs.genesys.com/Special:Repository/GIR_Sizing_Tool.xlsx?id=f6af92b7-9e60-4c47-841e-be58e5f44de0

13		CPU	Memory (MB)	IOPS	Disk space (GB)	Network bandwidth (Mbps)	# of instances required	Servers needed	Cores per VM
14	RP (per instance)	24%	50	60			3		
15	Muxer								
16	RCS (per instance)	3%	600						
17	HTCC (per instance)	40%	2627	Refer to cassandra	16	13	3		
18	Cassandra (per instance)	3%	2076	12	6		3		
19	MCP (per server)			84		113		2	8
20	Apache httpd	18%		87	refer to recording storage	3			

- Shows CPU, RAM and IOPS. Not number of servers
- Totaling 'CPU' and 'RAM' from individual components gives an indication of total number of CPU Cores and RAM memory required for servers
- '# of instances' is the number of processes that is required, not number of servers (non HA)
- When determining the server layout do not overload CPUs. Total up to a maximum of 60%-70% per CPU Core
- 'Servers needed' has dedicated column for components that cannot be stacked on a single server

6.1.1.1 MCP Sizing for GIR

Baseline Server Specification: 8 CPU Cores and 8 GB RAM

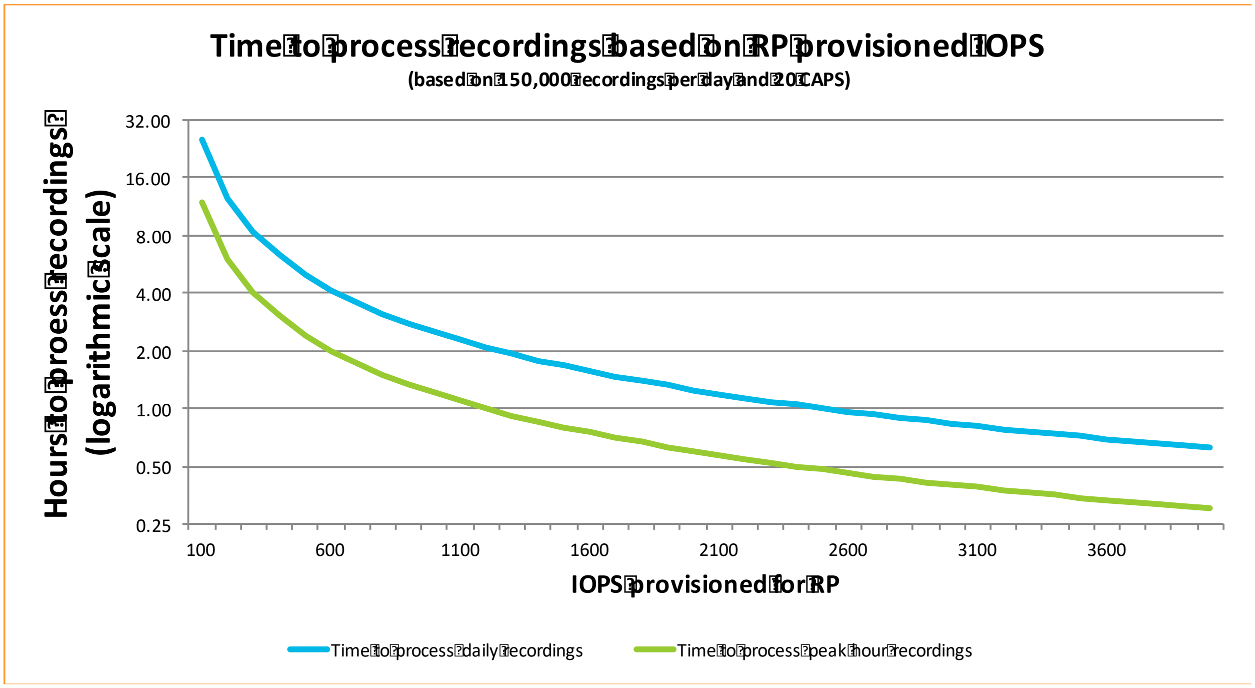
OS	Baseline	Encryption	MP3 + wav	8 kbps
Windows 2008 R2	350	306	210	449
RHEL 6	300	300	180	600

- Assumes G.711, using G.729 reduces capacity by 75% (divide above number by 4)

6.1.1.2 Recording Processor Considerations

Performance impact of recording processor when undersized:

- Reduction of time to processing during peak hour (falling behind)
- If processing backlog is too high, it can create a backlog on MCP side



6.1.2 Storage Sizing

Voice recording storage sizing should be based on:

- Average voice recording size: bitrate times the average file length in seconds divided by 8
  - 16 kbps for mp3 (stereo)
  - 8 kbps for mp3 (mono)
- Average recording size times the retention period

Screen recording storage – refer to the screen recording appendix.

6.1.3 Database Sizing

Database sizing is based on the:

- Number of recording per day
- The retention period for the recordings
- The baseline metrics for the database:

  - Initial size: 1 GB
  - Size increase per 100,000 records (recordings per day times the retention period) 0.1 GB
  - Index size per 100,000 records 0.3 GB

6.1.4 Network Sizing and Readiness

For GIR there are three main considerations for network bandwidth:

- MCP Bandwidth: density of MCP times the bandwidth for bridging
  - The calculated density (number of recording sessions) of each MCP
  - The bandwidth required for G.711 (0.25 Mbps)
- File storage (WebDAV): voice plus screen
  - Voice (Mbps): busy hour total recording size (MB) divided by 3600 times 8
    - Busy hour recording size: busy hour calls time average call duration time bitrate divided by 8 divided by 1024
  - Screen: Depends on a number of factors including the upload window and the video quality. Please refer to the screen recording appendix for details.

6.2 GIA Solution Sizing Guidelines

6.2.1 Solution Sizing Guidelines

This section provides guidelines on sizing the solution components to determine the server requirements. Providing a simple and accurate sizing guideline is difficult as there are many variables, between the number of expected daily calls, the expected turnaround time and the expected usage of the web user interface and generated reports.

The sizing of a GIA solution is most variable in the number of deployed Recognizer Servers as they are the critical path in performing speech analytics on all calls. In addition, sizing around the database and the latency in generating reports is important, however, we do not have sufficient data from engineering to providing tools to aid in sizing this component of the solution.

Please treat the information in this section as a rule of thumb. It is desirable to get a GIA solution architect involved for any firm customer sizing. More details on sizing can be found here:

https://intranet.genesys.com/display/STA/Sizing+Tool

6.2.2 Sizing Inputs Assumptions

The following table outlines the required input parameters for sizing:

Input Parameter	Description

Calls	Average number of calls expected to be processed daily.
Average Handle Time (AHT)	Average handle time per call, in minutes.
Maximum Turnaround	The maximum allowed time between a file being provided to GIA and it being available in the UI for review. Typically 24 hours.
Retention (RAW Audio)	Number of days to keep original, raw audio. Typically 3 days.
Retention (Compressed Audio)	Number of days to keep compressed audio. Typically 30 Days
Retention (Call Information)	Numbers of days to keep other call related information such as meta data, transcripts, etc. Typically 90 Days

Table 1: GIA Sizing Inputs

The following table outlines assumptions that are made regarding the sizing of the GIA solution:

Input Assumptions	Value
Compressed Audio Bit Rate	32 kbps
Uncompressed Audio Bit Rate	128 kbps
Required DB Size for Call Meta Data	0.7 GB/100K Calls
Required Disk Space for Index	2.0 GB/100K Calls
Required DB Size for Transcripts	1.0 GB/100K Calls
Recognition Speed (English & Spanish)	2 x Real Time
Recognition Speed (Other Languages)	1.3 x Real Time
Busy Hour Calls	20% of Daily Volume

Table 2: GIA Sizing Assumptions

6.2.3 Hardware Considerations

The sizing tool provides guidelines on the class of servers that should be used for a GIA deployment. The class of servers chosen is specific to the functions of that particular server.

For the server that houses the database, it is important to ensure that the machine is not virtualized as maximizing database access speed for a production deployment is important. In addition, the Database Server should be sized with a large amount of RAM and a fast disk for reads/writes.

For Recognition Servers, it is preferable to deploy fewer machines with a larger number of cores. The standard Recognition Server is 32 cores, but for a larger deployment it is preferable to size a machine with a larger number of cores and few servers if possible.

As the system fetches/moves files across the system, the speed of the network attached storage is an important consideration. We do not have firm IOPS information from engineering, however, a fast disk is preferred.

6.2.4 Virtualization Considerations

As noted previously in this document, each server in the GIA solution can be virtualized except for the Database Server, which must be run on a physical machine.

6.2.5 Storage Sizing

The sizing tool provides recommendations on Index and Audio Storage sizing based on the required retention period. In addition, the tool provides recommendation on the size of disk for each server based on the needs of the database or temporary files.

6.3 Configuration Guidelines

6.3.1 Genesys Interaction Recording Configuration Guidelines

6.3.1.1 Genesys Administrator and Genesys Administrator Extension

Audio Tones

To meet the regulatory requirements, some deployments require the system to periodically generate an audio tone to notify the participants in a call that the call is currently being recorded.

Audio tones can be generated either as all-party consent or one-party consent:

- All-party consent requires that all parties in the call being recorded hear the audio tone periodically.
- One-party consent requires that only one of the parties on the call to hears the audio tone. The consent is configurable on Media Server.

There is a difference between all-consent/one-party-consent and applying the beep to certain calls:

- All-consent/one-party-consent setting is a global system setting on the MCP process.

These parameters are configured as Recording Parameters in the IVR profile:

Parameter Name	Description
Recording Alert Tone	The URI of the audio tone. If the URI is set to an empty string, or not defined, or resolves to a bad URI, then no audio tone is applied to the call. No other notifications are generated by Media Server (for example, MSML events) when no audio tone is applied. Only .wav files are supported. DTMF tones and files stored in subdirectories with multiple codecs that are supported by Media Server are not supported. For example, "music/beep" cannot be specified for this option, even though it is valid for other Media Server treatments.
Frequency of Recording Alert Tone	The length of time, in milliseconds, between playing the audio tone. This is a mandatory parameter if the Recording Alert Tone parameter is defined, otherwise no audio tone is applied. The minimum accepted value is 1500 (if a smaller value is specified, 1500 will be used). In addition, if the Frequency of Recording Alert Tone parameter is not present, MCP applies the default value of 30000 instead of not applying tone.

Table 3: Parameters for Audio Tones

Recording Conference Audio Tones

When recording a conference, there are two Media Servers involved in the call:

- One for recording the recording DN.
- One for mixing media for other parties.

The audio tone is generated from the recording Media Server and is propagated to the conferencing Media Server. In order to ensure that all parties get the consent, *set record_recorddnhearstone*, and the *record_otherdnhearstone* options in the conference section of the Media Server application to *true*.

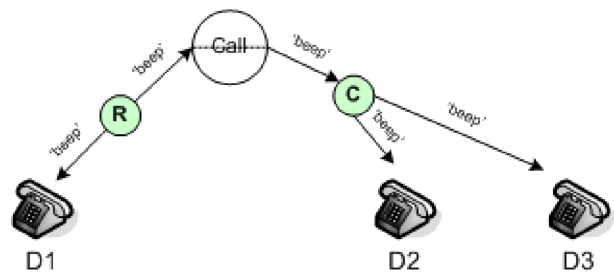


Figure 13: Call Flow and Audio Tones for Conference Recording

When the recording is paused, no audio tone is generated. When the recording is resumed, the audio tone is applied.

6.3.1.2 SIP Server

The core competencies of SIP Server are routing and call control. Also, SIP Server is responsible to initiate call recording by using media control to direct media towards Media Server.

SIP Server provides a T-Lib interface, this interface can be used for various recording related functions. These functions are outlined below.

Enabling Call Recording through T-Lib

The T-Lib interface allows recording to be enabled in three ways:

- Through configuration—Set the record option to true in the DN object to instruct SIP Server to enable full-time recording for this DN. This is an existing feature.
- Set the extension in the TRouteCall event to enable recording on trunk side or agent side. When calling TRouteCall, add the record key in the extension attribute and set the value to source for customer recording or destination for agent recording. This is an existing feature.
- Add a new extension in the RequestPrivateService event to request call recording to be enabled on an existing connection as described in the following table:

Parameters	Description
AttrPrivateMsgID	This parameter is mandatory and must be equal to GSIP_RECORD_START.
AttrThisDN	This parameter is mandatory and is the DN on behalf of which the operation is requested. It must be registered by the T-Client, but not necessarily be a party on the call (for example, the supervisor may request recording of the agent's call).

Parameters	Description
AttrConnectionID	This parameter is mandatory and references the ID for the call to record.
AttrExtensions	Additional request parameters: • record (string)—Set to **source** to record from this DN referenced in this connection. Set to **destination** to record from the other DN referenced in this connection. This parameter is optional, and defaults to source. • Id (string)—Adds a recording identifier to the recording session. This identifier must be globally unique and is passed back in the recording session. This parameter is optional and if not present, Media Server constructs a unique identifier. • Dest (string)—Overrides the default SIP location of the recording server. This parameter is optional. • Dest2 (string)—Overrides a second SIP location of the recording server for duplication of recording. This parameter is optional. • Params (string)—Additional parameters can be passed as generic name-value pairs. These parameters will show up in the recording session.
AttrReasons	The reasons. These are processed the same as for other T-Library requests all

Table 4: Extension for RequestPrivateService for Enabling Call Recording

Runtime Control of Recording through T-Lib

When the recording session is established, the T-Lib interface allows run-time control of the recording for pause, resume, and stop. The following table describes a new extension for RequestPrivateService:

RequestPrivateService—Request services that are supported only by certain T-Servers and that are not covered by general feature requests.

Parameters	Description
AttrPrivateMsgID	Specifies the operation. Choose one of the following values: • GSIP_RECORD_STOP—Stop the recording.

	• GSIP_RECORD_PAUSE—Pause the recording • GSIP_RECORD_RESUME—Resume the recording.
AttrThisDN	The DN on behalf of which the operation is requested. It must be registered by the T-Client, but not necessarily be a party on thecall.
AttrConnectionID	References the ID for the call being recorded.
AttrExtensions	Additional request parameters • Params (string)—Additional parameters can be passed as generic name-value pairs that modify the recording session.
AttrReasons	The reasons. These are processed the same as for all other TLibrary requests.

Table 5: Extension for RequestPrivateService for Runtime Control

Recording Indication

Two mechanisms enable SIP Server to provide recording indication:

- After SIP Server successfully starts recording on Media Server, SIP Server updates the UserData attribute of the call with GSIP_REC_FN using the file name of the recording. A T-Lib client monitoring the call receives an EventAttachedDataChanged with GSIP_REC_FN. This is existing functionality for legacy Stream Manager recording. For clients who only want to know if a recording has been enabled any time during the call, this userdata is sufficient.
- For clients such as, Workspace, who need to render the current recording state for the call, GSIP_REC_FN is not sufficient as a recording indicator. SIP Server provides a new value GSIP_RECORD for the UserData attribute to provide the current state of recording for this call. Whenever SIP Server knows there is a change in recording, SIP Server sends an EventAttachedDataChanged with GSIP_RECORD to update the value of the key. This key has three values:
    - On—Recording is currently in progress.
    - Off—No recording in progress.
    - Paused—Recording is currently in progress but no media is currently captured.

6.3.1.3 GVP Resource Manager

Genesys Voice Platform (GVP) Resource Manager (RM) is a SIP Proxy that manages a pool of Genesys Media Servers (MS) and applies runtime policies such as ensuring call legs to the same conference are pinned to the same Media Server. Resource Manager also generates the call detail record (CDR) that allows correlation with individual call recordings.

Genesys Voice Platform (GVP) provides various media services in addition to IVR, and depending on the usage, GVP can be configured as various different DN types on SIP Server. The following table shows the list of DN types and whether recording is supported for each DN type:

DN Type	Usage	Can be recorded by SIP Server
VoIP Service	Media services such as music-on-hold, conferencing, call parking, recording	No
Voice Treatment Port	Legacy IVR ports for both inbound and outbound IVR calls	Yes
Trunk Group DN	Call Progress Detection (CPD) and proactive notification (outbound GVP IVR calls)	Yes
Trunk DN	Inbound GVP IVR calls	Yes

Table 6: DN Types for GVP

6.3.1.4 Recording Processor

This component builds and sends call recording metadata to the Web Services API and SpeechMiner servers. The component builds the recording metadata from ICON, and from the metadata from MCP for each recording file. The overall structure is in JSON format with the following mandatory core properties:

- id—The recording identifier.
- callerPhoneNumber—The caller DN (ANI).
- dialedPhoneNumber—The dialed number (DNIS).
- mediaFile—A list of recording files associated with this recording.
- eventHistory—A list of call events associated with this recording.

mediaFile Properties

The following table describes the mediaFile properties:

Property	Data Type	Description	Required
mediaDescriptor	object—Contains the path and storage property.	Specifies the path and storage location of the media file.	Yes
startTime	datetime	Specifies the start time of the media file.	Yes
stopTime	datetime	Specifies the stop time of the media file. If MCP fails, this value will be the same as the startTime.	Yes

Property	Data Type	Description	Required
mediaID	string	Specifies the media file name for the media file that is used by clients to refer to the same media file. MCP ensures that this value is globally unique.	Yes
type	string	Specifies the MIME type of themedia file.	Yes
duration	time	Specifies the time duration of the media file.	No
size	number	Specifies the size, in bytes, of the media file.	No
tenant	string	Specifies the tenant that the recording belongs to.	Yes
ivrprofile	string	Specifies the IVR Profile name that serviced the recording.	Yes
parameters	object—The properties are parameters	Specifies the list of additional metadata information provided by SIP Server and the client applications. The properties are: sipsAppName, ani, dnis, dateTime, calluuid, connid, agentId, recordDn, record	Yes
masks	array of objects—Each object contains the time and type property	Specifies the time stamps of the pause/resume periods if the recording is masked by a client application.	No
pkcs7	string	Specifies the PKCS7 envelope (in PEM, base 64 string format) if the media file is encrypted	No

Property	Data Type	Description	Required
certAlias	array of strings	Specifies a list of aliases to the encryption certificates if the media file is encrypted.	No (Yes if the pkcs7 property is present)
partitions	array of strings	Specifies a list of partition names for the media file.	Yes
accessgroups	array of strings	Specifies the access groups identified agent associated with the recording	Yes

Table 7: GIR Metadata mediaFile Properties

eventHistory Properties

The following table describes the eventHistory properties:

Property	DataType	Description	Required
occurredAt	datetime	Specifies the start time of the event.	Yes
calluuid	string	Specifies the call UUID that the event belongs to.	Yes
event	string	Specifies the event type: Joined, Left, data	Yes
contact	object	Specifies the contact information of the caller who joined or left the recording if the event is Joined or Left.	No
data	object	The attached data included in the recording if the event is data.	No

Table 8: GIR Metadata eventHistory Properties

6.3.1.5 Recording Crypto Server

This component provides the Key Management System for the Genesys Interaction Recording solution. Further details on its function can be found in the Security section.

The Recording Crypto Server (RCS) is also responsible for archiving. By default, the component is scheduled to generate an archive file every day based on the retention period. Each time the archival process is run, the output file is saved to a compressed file. The compressed file contains the recording files and the metadata. The following directory structure is the result of the expanded compressed file:

```
/{contact-center-id}
    /{tenant-name}
        /{year-month-day}
            /{recording-id}
                /recording.json
                /{mediaFiles}
```

Where each item is defined as:

- contact-center-id—The unique identifier of the contact center that is generated by Genesys Web Services.
- tenant-name—The name of the tenant.
- year-month-day—The directory where the recording files are stored. There is a separate directory generated for each calendar day.
- recording-id—The unique recording identifier for each recording.
- recording.json—The recording metadata.
- mediaFiles—One or more recording files for the recording. The file name is referenced in the metadata. If encryption is enabled for this tenant, the recording is also encrypted in PKCS7 format.

6.3.2 Genesys Interaction Analytics Configuration Guidelines

<insert GIA guidelines here>

6.4 Security

Protecting the customer's infrastructure should be imperative for any solution deployment. Genesys components can typically be deployed in a secure manner. Many customers have their own security procedures that our solution needs to conform to. The following are guidelines for some of the requirements that may be encountered or should be recommended.

6.4.1 Secure Connections

Connections between components, especially those external to the solution (see 0

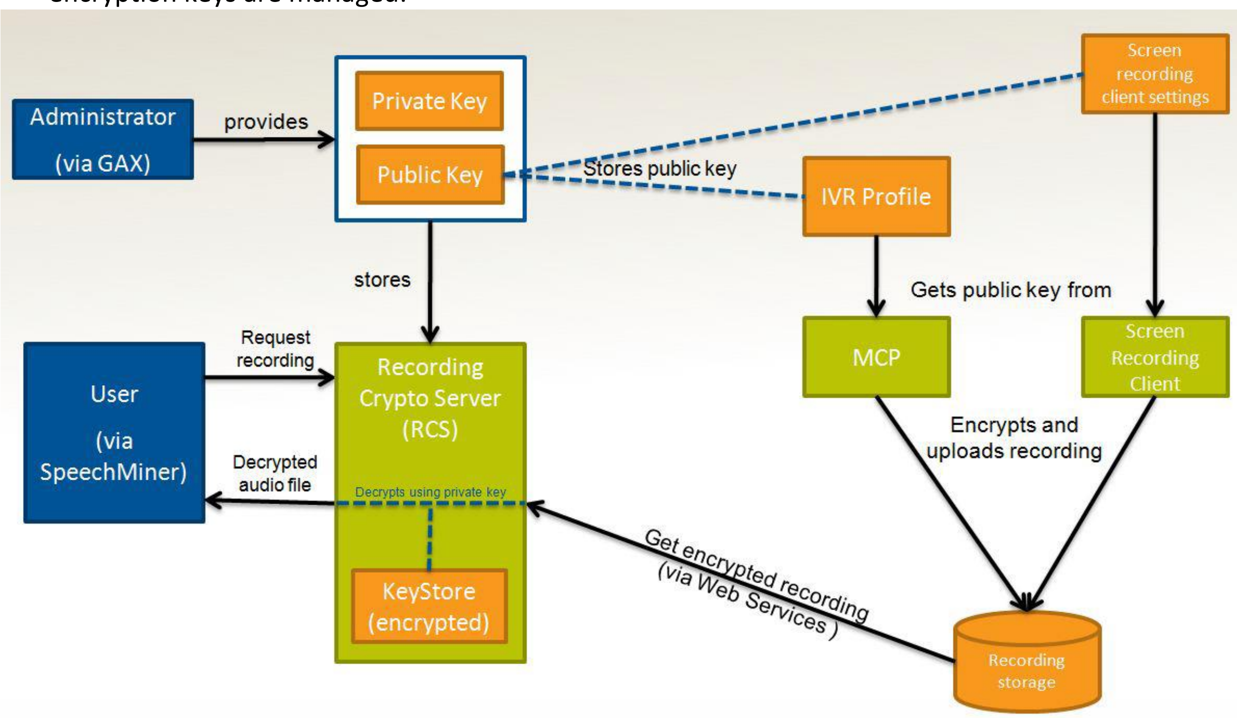
External Interfaces) should be secured. Where possible use SSL or HTTPS.

Typically customers will insist on firewalls to protect HTTP traffic from the wild internet. In a similar fashion Media Gateways or Session Border Controllers need to be configured to protect VoIP traffic.

6.4.2 Data Security Considerations

6.4.2.1 GIR Encryption

- **Security and Encryption** - A key management system allows users to decrypt call and screen recordings for playback purposes. The following diagram provides a quick overview on how the encryption keys are managed:



- **Security Keys and Storage -** The administrator for the customer tenant provides two keys:
  - Private key—for decrypting audio files.
  - Public key—for encrypting audio files.

- **Private Key Storage** - The Recording Crypto Server (RCS) uses Java Cryptography Architecture (JCA) to store private keys. Private keys are accessed through the Java KeyStore and rely on third-party providers to provide a secure KeyStore solution.

- **Key Size Limit** - The default Java installation limits key sizes to 128 bits. Larger key sizes can be enabled by installing the Java Cryptography Extension (JCE) Unlimited Strength Jurisdiction Policy.

- **Public Key Storage** - Public keys for voice recordings are stored in the IVR profile for the tenant. The Media Control Platform (MCP) reads the certificate from this IVR Profile. A customer may include multiple private/public keys for a tenant; however, private keys per user are not necessary. Since MCP is shared across multiple tenants, there must be a single place to store all certificates that are accessible. Configuration Server stores all certificates (text format) as parameters in the IVR Profile for call recording. MCP loads all call recording IVR Profiles and receives updates for the IVR Profiles. When Resource Manager forwards a recording request to MCP, it inserts the IVR Profile DBID so MCP can look up the list of certificates to perform encryption. The certificates are stored in the IVR Profile Annex tab in the recording-certificates section. Public keys for screen recordings are stored in the screen-recording-encryption settings of Web Services.

- **Encrypting Call Recordings -** MCP encrypts call recordings in two parts:
    - The audio file itself is encrypted with a session key (for example, AES encryption).
    - The encrypted session key uses public keys provisioned for the tenant. MCP posts the encrypted audio data to the S3 cloud, or to the disk, and provides metadata and the encrypted session key in a PKCS7 envelope in the metadata in JSON content in the pkcs7 property.

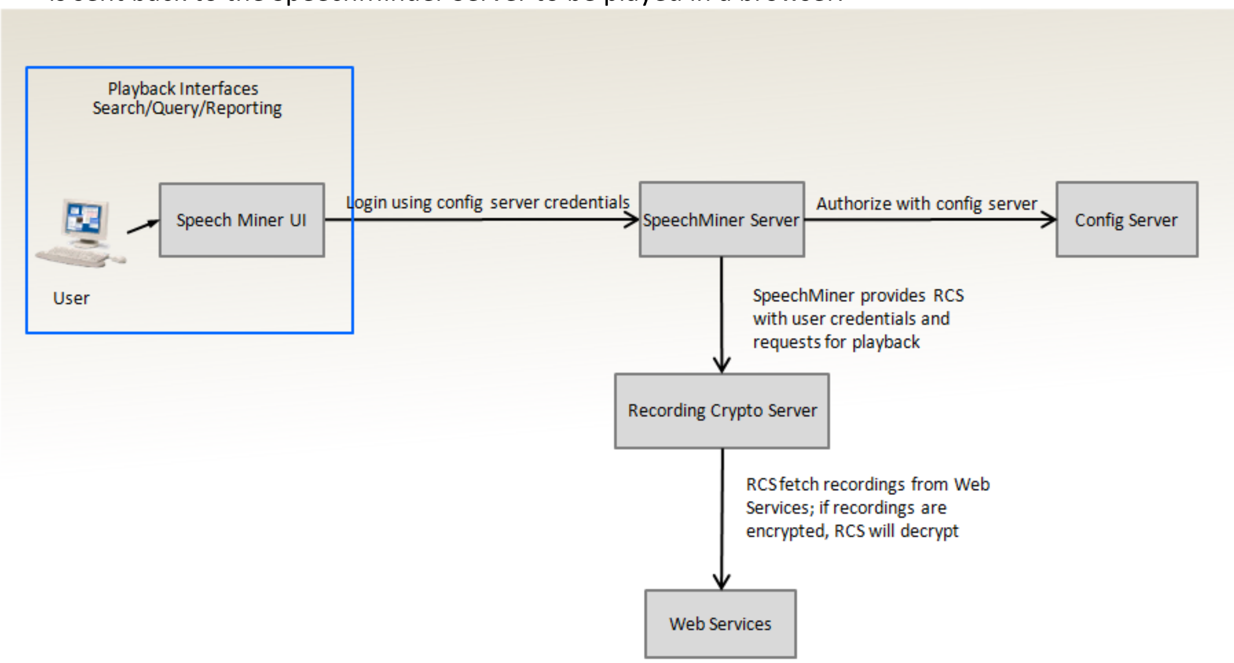
  When Web Services receives the encrypted symmetric keys in the PKCS7 envelope, Web Services can provide key rotation of the public keys without needing to modify the encrypted audio content. Whenever a call recording is requested, Web Services concatenates the encrypted symmetric key and the encrypted audio data as a single PKCS#7 content response.

- **File Format -** The Genesys Interaction Recording solution supports RSA certificates and keys for the recording encryption. Certificates and keys are uploaded from files in PEM format. Certificates files must be X.509 PEM format. This is the default format generated by the Openssl. The following formats are supported for private key files:
    - Openssl RSA private key format. The PEM file must start with -----BEGIN RSA PRIVATE KEY-----.
    - PKCS#8 private key format. The PEM must start with -----BEGIN PRIVATE KEY-----.
    - PKCS#8 encrypted private key format. The PEM file must start with -----BEGIN ENCRYPTED PRIVATE KEY-----.

  For an example of encryption using OpenSSL, see Sample Encryption Using OpenSSL.

- **Certificate Validation** - Certificate validation is performed when certificates are uploaded to the system and upon each use of a certificate to encrypt a recording. The validation consists of checking the certificate signatures from the recording certificate up to the root CA certificate, and checking that the certificate "not valid before" date has passed and "current" date is within the "not valid before" and "not valid after" dates. A certificate that is not yet valid can be imported, but an error will occur if it used for a recording.

- **Decrypting Call Recordings -** If the user wants to play back a specific recorded call, the SpeechMiner Server checks for the user's permission and makes a request to the Reporting Crypto Server to fetch and decrypt the recording. The Recording Crytpo Server retrieves the call metadata and fetches the recording from Web Services. The recording files are decrypted using the private key, and the audio is sent back to the SpeechMinder Server to be played in a browser.



- **Playback of Archived Files -** The archiving function provides a zip file containing multiple encrypted recording files. Each encrypted file is in the PKCS#7 format. Each encrypted file can be played back with the following Openssl command: `openssl smime -decrypt -inform DER -in encryptedFile -inkey pemkeyfile`

6.4.3 VM and OS hardening

Operating Systems are often pre-configured for ease of use and development and not necessarily security. If the O/S is being installed or is part of a set of VMs being delivered, that O/S should be hardened to ensure that typical security holes are addressed.

The following document provides recommendations that can be used to harden the solution VMs and the OS.

Microsoft Word 97 - 2003 Document

6.5 Localization and Internationalization

Localization and Internationalization are topics for numerous Genesys components, especially user interfaces and reporting. Within the GIA and GIR Solutions, the main components to pay particular attention are:

- Speech Analytics Processing
- Web Based Administration & Operation Management User Interfaces

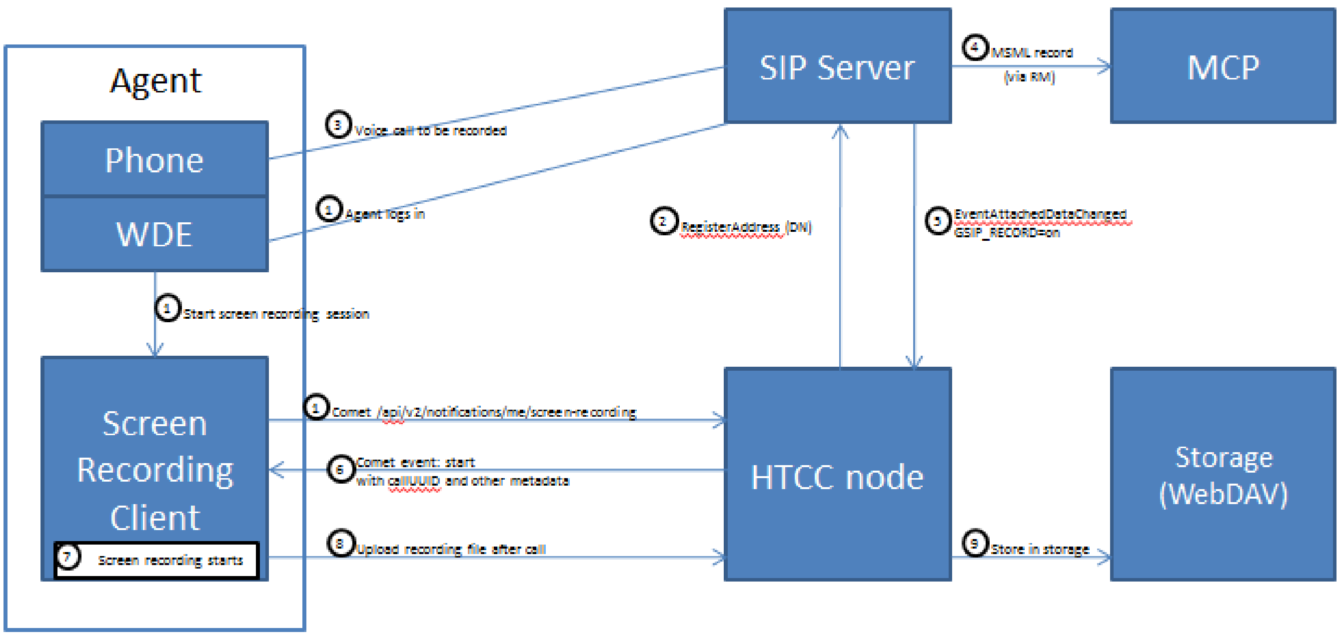
For Speech Analytics processing carried out by the Recognizer Server, only a limited set of languages is supported. Please refer to the following intranet site for the latest information on which languages are supported:

https://intranet.genesys.com/display/~david.ollinger@genesys.com/Language+Enablement

Appendix A Screen Recording Client

The Screen Recording client is currently integrated with Genesys Workspace Desktop Edition (WDE) and Genesys Web Services and Applications.

The following diagram illustrates Screen Recording architecture in a WDE integration:



1. The Screen Recording Client runs on the Agent's PC as a Windows Service. The Screen Recording Service communicates to Genesys Web Services (node or load balancer) to enable the registration of the agent DN.
2. Genesys Web Services registers for events from SIP Server for the Agent's DN.
3. When the Agent is in a call that is being recorded, Media Control Platform (MCP) records the voice portion.
4. Web Services receives the voice recording event.
5. Web Services determines that the Agent's screen is to be recorded and notifies the Screen Recording Client.
6. The client records the Agent's screen.
7. When the call ends, the Screen Recording Client uploads the recording to Web Services.
8. Web Services stores the recording in the recording storage location.

A.1. Limitations

- Interaction-based screen recording

  - Means: single searchable and playable screen recording file per handled interaction
  - Is tied to voice only, and can be based on percentage of recorded calls, and allows QM
  - Is not possible with non-voice channels (e-mail, chat, social, etc.), therefore no QM
- Blended agents are not supported
  - Agent that has configured, and logged into, a voice DN, only gets screen rec for voice
  - Agent that logs into voice and non-voice at the same time, does not get screen rec
  - Agent that logs into non-voice only, and has no voice DN configured, gets screen rec from the moment they login, until they logout. (continuous screen recording)
- Non-voice screen recording has no metadata, except agent-ID and date/time

A.2. Deployment Considerations

A.2.1. Screen Resolution

The Screen Recording Service works with most screen resolution settings. However, Genesys has tested the Screen Recording Service under the following recommended screen resolutions:

Single Monitor:

- 1024 x 768
- 1280 x 720
- 1600 x 1200
- 1920 x 1080

Dual Monitor:

- Side-by-side 1024 x 768 + 1024 x 768
- Side-by-side 1280 x 720 + 1280 x 720
- Side-by-side 1600 x 1200 + 1600 x 1200
- Side-by-side 1920 x 1080 + 1920 x 1080

To use the Screen Recording Service SRS on a computer with different screen resolution than listed above, a field validation of the Screen Recording Service against the desired screen resolution should be performed.

A.2.2. Security Certificates

It is advisable to use a security certificate for the communication between the Screen Recording Service and the Genesys Web Services. There are the following options available:

1. A self-sighed security certificate is created and populated into the Windows certificate store at installation
2. A self-signed security certificate can be created post installation and populated into the Windows certificate store.
3. A installation specific certificate can be used.

A.2.3. Network Bandwidth

The Screen Recording Service sends the captured recording file to Genesys Interaction Recording solution via the Genesys Web Services solution. The following considerations need to be taken:

The file size is based on the recording quality:

Quality	**Bitrate - Kbps**	**Size Per Minute – KB/min**
Low – Greyscale	77	577.5
Standard – Greyscale	88	660
High – Greyscale	91	682.5
Low – Color	103	772.5
Standard – Color	150	1125
High – Color	170	1275

By default, the recording is sent immediately to Genesys Web Services upon completion of the interaction.

A.3. Common Component – Genesys Web Services

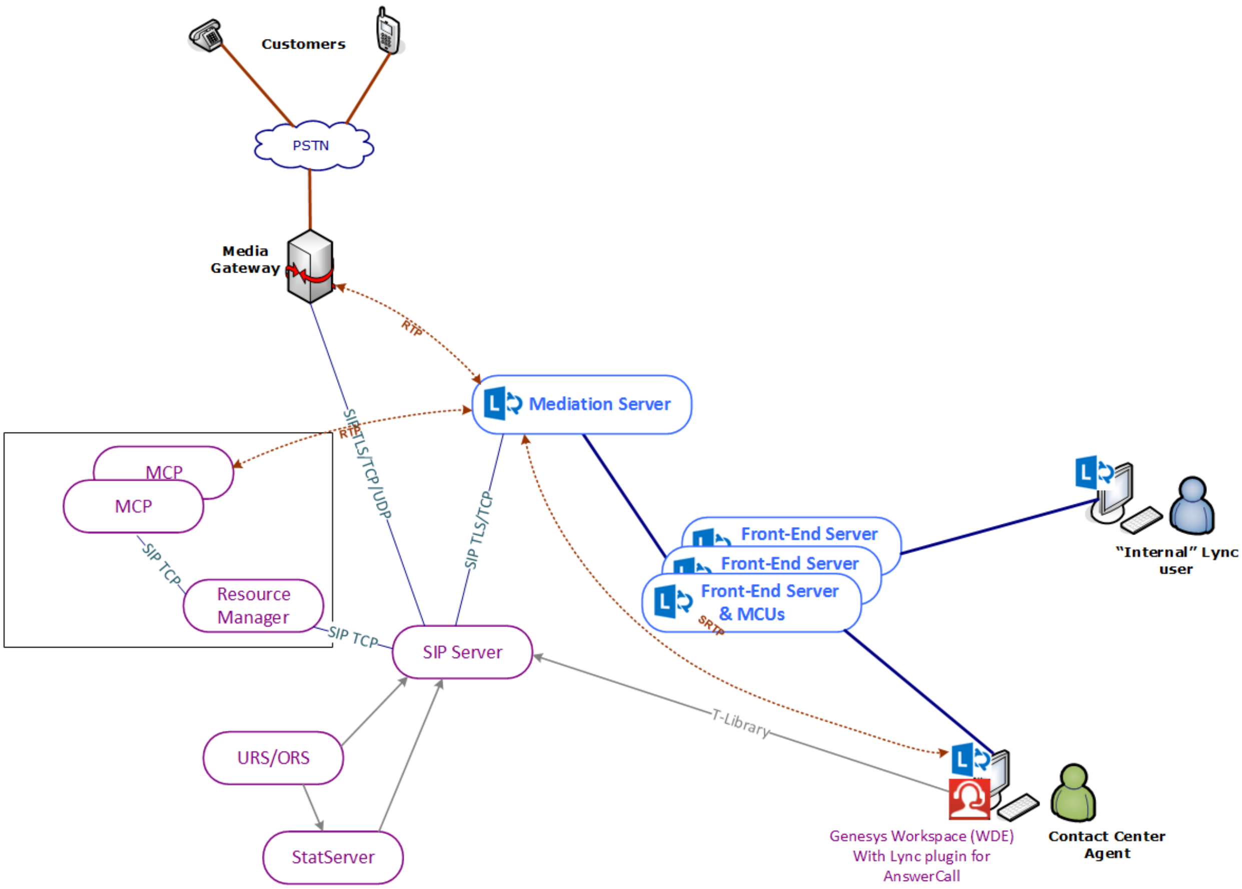
As with call recordings, Genesys Web Services requires a specific configuration for GIR ***screen*** recordings to work correctly.

Appendix B Recording - SIP In front – Skype for Business

Although SIP in front is not a supported architecture an exception has been made when implementation is within a Skype for Business environment.

It MUST be noted that GIR is only supported with the SIP Integration Skype for Business architecture and not the native Skype for Business (Lync) integration via UCMA.

An overview of the SIP integration solution is highlighted in the following diagram.



Appendix C Migration Strategies

TBD