

VOICEGENIE

VoiceGenie 7 SS7 Connector Users' Guide

April 13th, 2005



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Revision History

Version	Date	Change Summary	Author/Editor
1.0	January 16 th , 2004	Initial release	Henry Lum
2.0	May 3 rd , 2004	Version 1.2 updates	Henry Lum
3.0	June 15 th , 2004	Added redundancy Information	Henry Lum
4.0	October 6 th , 2004	Version 7 updates	Henry Lum
5.0	February 2 nd , 2005	Version 7 updates	Henry Lum
5.1	April 13 th , 2005	Final Revision for VoiceGenie 7 Release	Andrew Ho

1 Introduction

This section contains the following major topics:

Functional Overview – Descriptions and diagrams of the hardware and software interface in the VoiceGenie SS7 Connector. The section explains each VoiceGenie component and network layout of the components.

VoiceGenie Components – Information on how Call Control Platform is integrated into other VoiceGenie components.

Redundancy – Provides a short discussion about what SS7 Connector provides in terms of SS7 redundancy and its configuration.

1.1 Terminology

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

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

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

Historical Terms	New Terms
CCP-SS7/CCPSS7	SS7 Connector
PhoneWeb Software / NeXusPoint 6.4.x Software	VoiceGenie 7 Software
Cluster Management Platform (CMP)	OA&M Framework
Voice Resource Manager (VRM)	Speech Resource Management (SRM)
VoiceGenie Management Console (VMC)	System Management Console (SMC)

1.2 Functional Overview

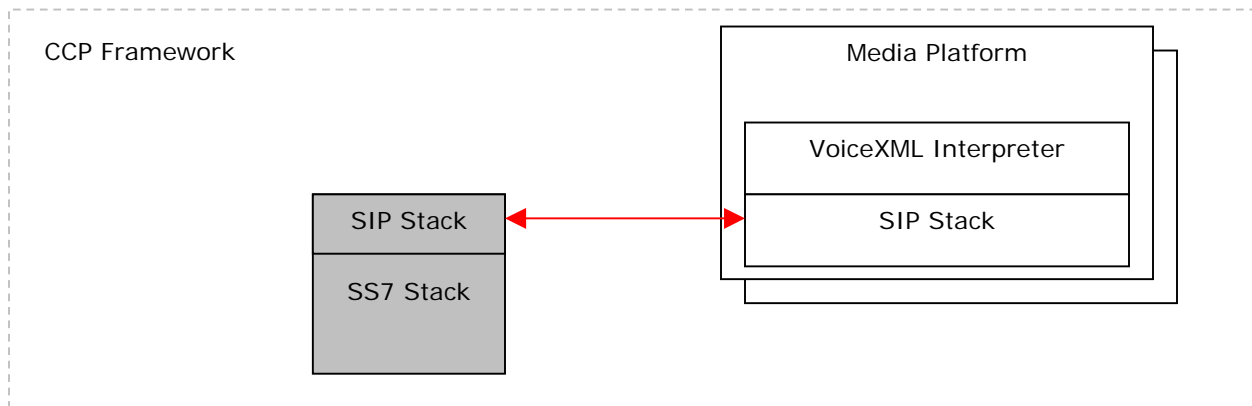
The SS7 signaling protocol is a widely deployed protocol that is used within service provider networks for the purposes of providing call control and enhanced services within the telecommunications network. Unlike other channel-associated signaling, SS7 signaling is independent of the actual voice path by which voice traverses the network.

VoiceGenie SS7 Connector capabilities will be delivered through the VoiceGenie Call Control Platform (CCP), and the description will be explained in the next subsection. In order to support SS7 signaling, VoiceGenie created an SS7-specific protocol module that will reside within the generic CCP framework, allowing support for ISUP call control.

Call Control Platform defines a framework that allows different types of external signaling to be integrated with the media resources within the framework (Media Platform and other media resources). Specifically, VoiceGenie supports SS7/ISUP, and CCP also supports CTI integration and external SIP media gateway.

The Call Control Platform manages the presence of the media platforms and directs SS7 calls into the media platform. The CCP framework allows call control applications such as CCXML (not available yet) to manage the interaction of the call if it needs to modify the route of the call or have the call to access multiple media resources for the duration of the call.

There are multiple components that are managed by the Call Control Platform; a simplified diagram below provides the components for a SS7 configuration:



Lines in red above are SIP communication paths, and the gray components are Call Control Platform components. All of the components above contains a SIP stack and uses SIP as the standard communication protocol between all components within the CCP framework. Media Platform participates in the CCP framework as a media resource to receive incoming ISUP calls and execute VoiceXML pages on the calls. When an ISUP call comes into SS7 stack, it sends a SIP INVITE to the Media Platform.

CCP framework also provides a component called Redundancy Manager (CCP-RM) that provides redundant solution to the SS7 Connector configuration. The CCP-RM is responsible for maintaining synchronization between a pair of SS7 Connector servers and notifies each SS7 Connector server its role

as an active server or standby server. Please refer to Call Control Platform Redundancy Manager for system architecture and configuration.

1.3 VoiceGenie Components

Before explaining the VoiceGenie architecture and provide examples of configurations, the following subsections describe the functionality of each VoiceGenie component.

1.3.1 Media Platform (MP)

Media Platform contains a VoiceXML interpreter that allows incoming/outgoing calls to interact with VoiceXML dialogs. Media Platform integrates media resource boards such as Brooktrout TR-1000 so that it can accept media coming from DS0s or from RTP packets.

1.3.2 Speech Resource Management (SRM)

Speech Resource Management is a server that keeps track of all Automatic Speech Recognition (ASR) and Text-to-Speech (TTS) engines and distributes requests from Media Platform to the engines.

1.3.3 Cluster Management Platform (CMP)

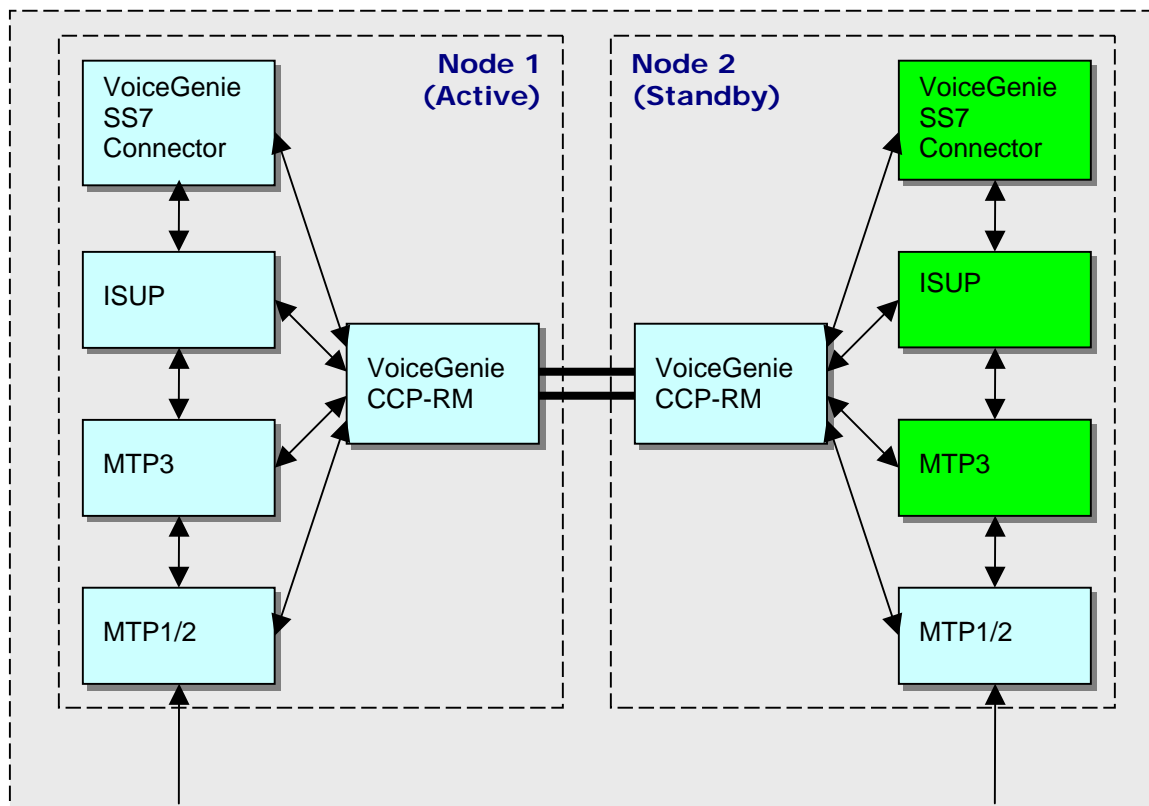
The Cluster Management Platform is a management platform that takes care of operations, administration, and maintenance. It manages the software process of the entire components list above, and provides the following three main functionalities.

- 1) Monitoring and management agent. A CMP agent is embedded into all VoiceGenie software components to be managed and monitored by the CMP. The CMP agent will define an object-based structure that allows monitoring/management information to be exposed via SNMP. However, the CMP agent will also provide a non-SNMP based connection used by the monitoring and management engine (below) to obtain the real-time status of the system.
- 2) Monitoring and management engine. This part of the CMP will extend the monitoring currently performed by platform pooling to be real-time in nature, using a persistent connection into VoiceGenie software based on the CMP agent. This engine will also be responsible for carrying out requests issued against the platform, such as starting or stopping the software, or performing an upgrade or rollback of nodes within a cluster.
- 3) System Management Console (SMC). This web-based interface will retrieve information from the monitoring/management engine through a database, and will provide a visual interface allowing the current status and configuration of VoiceGenie software to be reviewed. The web-based interface will also allow operations to be scheduled against components or groups of components. The monitoring/management engine will actually perform the operations. A command-line interface also provides most of the functionality offered by the SMC, called the command-line console (CLC). This component runs on each platform that are part of the CMP cluster.

1.4 Redundancy

SS7 Connector software also provides a "hot-standby" mode of operation. In this mode of operation, two SS7 Connector servers will operate in a mated pair, with each SS7 Connector server being aware of the state of the other SS7 Connector server. The SS7 Connector servers will communicate via the dual redundant Ethernet interfaces such that loss of a single Ethernet interface or switch will not cause the loss of communication between the SS7 Connector server pair. Continuous heart-beating will be used to ensure that the failure of one of the servers in the pair is detected quickly.

In this paired configuration, one SS7 Connector server will be "Active"; the other SS7 Connector server will be "Standby". The MTP2 layer will be active on both SS7 Connector servers, but the MTP3, ISUP, and application (SS7 Connector server software) layers on the standby server will be in a standby state – ready to operate, but not connected to the network. Each SS7 Connector server will have at least one active SS7 A-Link to the network. MTP2 SS7 traffic arriving at the standby server will be automatically routed, over the redundant Ethernet interfaces, to the primary SS7 Connector server. The following diagram shows this normal operating condition:



In the above diagram, the CCP Redundancy Manager (CCP-RM) is shown. It is responsible for maintaining synchronization of the pair, and fail-over in the case of a failure of one of the nodes. Active components are shown in cyan; standby components are shown in green.

Section 3 of this document explains the list of failure scenarios and how the hot backup server can recover the failure conditions.

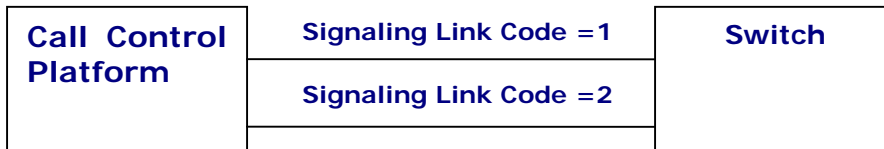
2 Installation

Please see [VoiceGenie 7 SS7 Connector Installation Guide](#) for details.

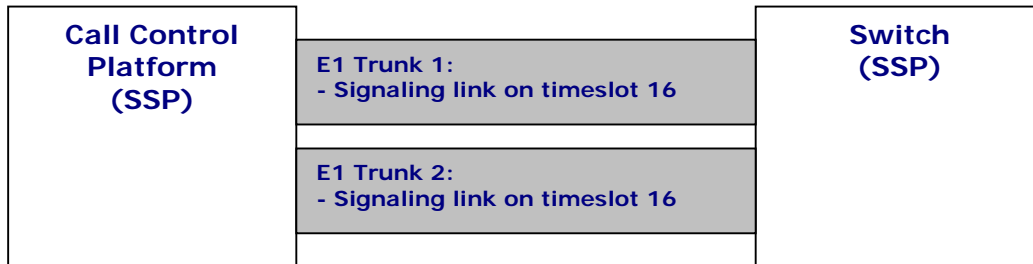
3 System Configuration Overview

3.1 SS7 Signaling Links

The Call Control Platform acts as a local Signaling Point (SP) within the SS7 network and is using a single link-set comprising two signaling channels to communicate with the remote SP through a single route. Here is the basic SS7 link-set diagram of Call Control Platform:



The signaling links are carried on a timeslot two separate T1/E1 trunks. The bearer channels (media) are carried on the rest of the timeslots or on other trunks. The following diagram is a simple signaling link set up:

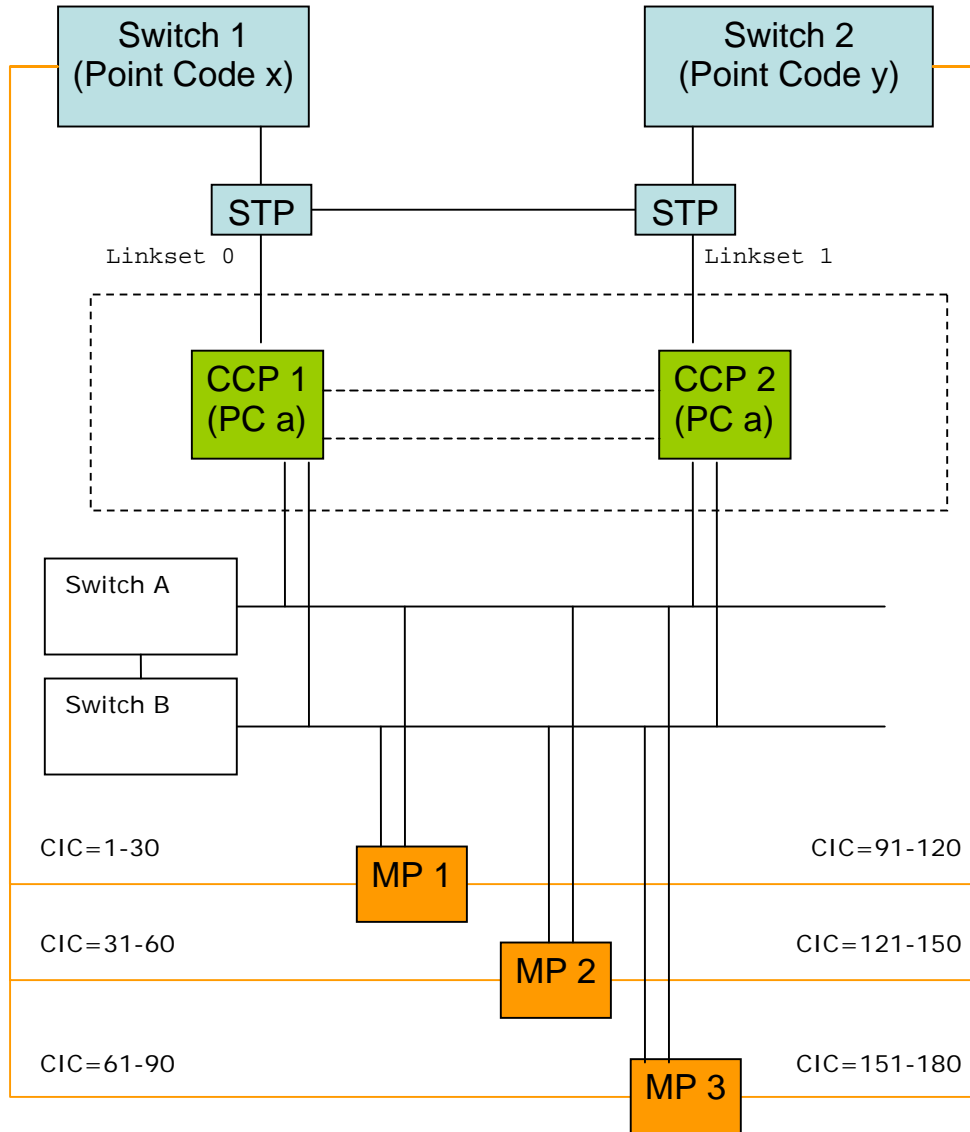


The signaling channels controls the bearer channels through the use of Circuit Identification Codes.

3.2 Redundant Solution – SS7 Configuration

This configuration is a fully distributed, which a pair of CCP and MPs reside on separate boxes. This configuration is logical for cases where signaling links are completely decoupled from the media links and there are redundant signaling links. The following diagram also shows the CMP database is placed on a separate box while there is a pair of redundant CMP engines running on each CCP box. For simplicity, we

will treat MP as the combination of Call Manager, VXML, ASR Manger, and TTS Manager. The pair of SS7 Connector servers will be managed by the Redundancy Manager that is not shown in the diagram below.



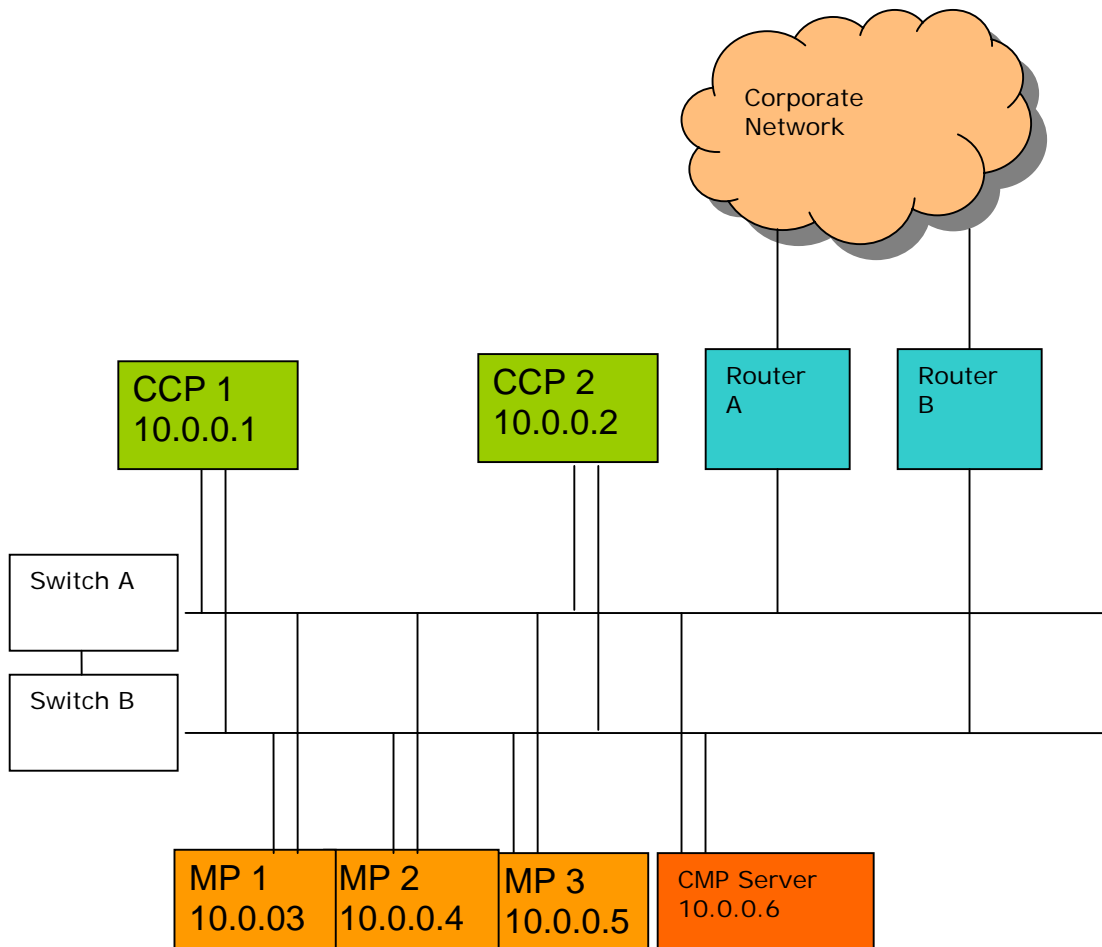
If one of the CCP is down, the surviving CCP will use the remaining route to handle both destinations. No interruptions will occur at the ISUP circuits.

If one of the switch is out of service, one destination will be unavailable and half of the circuits will be out of service.

There are two linksets, Linkset 0 and Linkset 1, each with one signaling link, which connects to an adjacent Signaling Transport Point (STP). The routeset for each destination will include both linksets. If one of the linksets is unavailable, the other linkset can be used as the backup route to send traffic to the destination. There will be no interruptions to the ISUP circuits.

3.3 Redundant Solution - LAN Configuration

The following diagram gives a network view of the configuration that provides redundancy against a single point of network failure.



Each VoiceGenie box will use a bonding Ethernet driver to allow failover of Ethernet device transparently. This means that both Ethernet devices share the same MAC address and IP address. The switches are cascaded to provide protection against power failure of a single Ethernet switch. All of the VoiceGenie use public IP addresses (10.x in this case) that are reachable from the corporate network to allow maintenance operations. In order to provide redundant routes to the corporate network, we will use a

pair of routers with Router B as the hot-backup router in case of failure of Router A. All VoiceGenie machines will reach the corporate network through the gateway provided by either Router A or B, where they will share a single virtual gateway IP address.

3.4 Cluster Management Platform Configuration

Using the above network diagram, we will place CMP components on each machine in the following table:

Machine	CMP Database	CMP Server	CMP Proxy	SNMP	SMC
CMP Server	Y	Primary	Y	Y	Y
CCP1	N	Backup	Y	Y	N
CCP2	N	N	Y	N	N
MP1	N	N	Y	N	N
MP2	N	N	Y	N	N
MP3	N	N	Y	N	N

3.5 System Hardware

3.5.1 Operating System

VoiceGenie qualifies the software on VoiceGenie Linux 3.0 running kernel version 2.4.21-20ELsmp.

The Linux distribution is contained in the VoiceGenie installation disks along with other software installation packages.

3.5.2 System

VoiceGenie supports HP DL380 G3 chassis. The minimum configuration will be:

Dual 2.8 GHz Xeon processors
2GB RAM
30GB hard disk space

Please refer to HP user documents for detailed chassis specifications.

3.5.3 SS7 Hardware

VoiceGenie uses Brooktrout NS700 as the hardware interface to the SS7 network. It is a PCI hardware board with 4 E1/T1 ports and supports H.100 bus. It can support up to 8 signaling channels on each NS700 board. Call Control Platform integrates one board for non-redundant configuration or two boards for redundant configuration.

The following are the SS7 Layers supported by the NS700 board:

- SS7 / C7 (ANSI, CCITT / ITU-T, ETSI, variants)
- MTP L2 & L3 solutions
- ISUP (ITU-T / ETSI V1 & V2)

- ANSI ISUP
- UK ISUP
- BTNUP/IUP
- Chinese TUP
- SCCP
- TCAP (ANSI, ITU-T)
- GSM Short Message Service
- MAP (GSM, 3GPP)
- ETSI INAP

The following is a list of ISUP variants supported by the ISUP stack:

Document	Protocol	Date/Version
ANSI T1. 113	ISUP	10/2000 Ver 1
ITUT Recommendation Q763, 764	ISUP	03/1993
ITUT Recommendation Q767	ISUP	02/1991
ETSI ETS 300 335	ISUP V1	ed.1 (1994-07)
ETSI ETS 300 356-1	ISUP V2	ed.1 (1995-02)
BAPT223 ZV3	German ISUP V2	Version 3
220-250-732	Brazilian ISUP	Standard Issue 3 1997
PTT692.08	Swiss ISUP	Edition 1 Heft V111E
NOM-112-SCT1-1995	Mexican ISUP	1-1995
PNO-ISC Specification 006 - BSI PD6645	IUP (BT National User Part)	05/1999

3.5.4 Media Processing Hardware

VoiceGenie uses Brooktrout TR1000 as the media processing hardware running on the Media Platform to receive audio from DS0s. It is a PCI hardware board with 2 T1/E1 ports and supports H.100 bus.

4 Redundancy

As described in the system configuration diagram, SS7 Connector provides redundancy to handle failures in SS7 links or SS7 Connector processes. This section discusses the failover scenarios in detail and describes how CCP handles these scenarios.

4.1 Hot Standby CCP and Call Fail-over

The system configuration diagram shows that there is a pair of SS7 Connector servers. One of the SS7 Connector servers will be acting as the hot-standby server, preparing to take over when the active server dies. MTP3 and ISUP components make use of check-pointed state information to ensure circuit states are maintained after a standby change-over. Note that signaling links connected to the hot-standby server will be active; the actual process of the signaling information will be transported to the active server.

Calls that are in connected state will be check-pointed by ISUP component. A call is considered in connected state when the ANM message is received by the caller. Check-pointing happens immediately after the call is in connected state on the active server and the standby server sets the call to connected as soon as it receives the check-point information,

Calls that are establishing (before receiving the ANM message) will not be handed over. These calls will be timed out by the caller and releases the call by sending REL. New calls can arrive on this circuit normally after the call is dropped. Note that a call is in establishing state when playing early media or making answer transfer; call failover is not available until the call is connected (or ANM is sent).

When the standby SS7 Connector server takes over as the active server, the standby system is capable of taking over with full awareness of the calls that are currently in connected state. This allows connected calls to be dropped or transferred successfully.

Call fail-over is achieved with IP multicast; the active and standby nodes both listen for SIP traffic on the same underlying multicast IP address. All addressing information – such as the SIP Via header, Contact header, and so forth, reflect the multicast IP address, not the separate unicast IP addresses that the active and standby SS7 Connector servers have. When acting as a standby, a SS7 Connector server will ignore all SIP traffic; when it becomes active, after synchronizing state, it simply starts processing messages that it receives instead of ignoring them. The system configuration places both SS7 Connector servers and the MP on the same LAN segment so that multicast packets will only be received by machines on this LAN and not on the corporate network.

4.2 Failure Scenarios

The following table summarizes the failure scenarios and the corresponding recovery for each of the two releases.

Failure Scenario	Result
Link Failure on Active Plane	MTP changeover; no interruption
Link Failure on Standby Plane	MTP changeover; no interruption
Card Failure on Active Plane	MTP changeover; no interruption
Card Failure on Standby Plane	MTP changeover; no interruption

Power Failure on Active Plane	Connected calls stay connected; no service interruption to connected calls. Establishing calls will be lost.
Power Failure on Standby Plane	MTP changeover; no interruption
Process Failure on Active Plane	Connected calls stay connected; no service interruption to connected calls. Establishing calls will be lost.
Process Failure on Standby Plane	MTP changeover; no interruption
CCP-RM Failure on Active Plane	Connected calls stay connected; no service interruption to connected calls. Establishing calls will be lost.
CCP-RM Failure on Standby Plane	MTP changeover; no interruption
Switch1 down	Circuits served by Switch1 will be lost. (All calls lost and no new calls can arrive to these circuits) No service interruption to circuits on Switch2. Outbound calls will only be placed on Switch2's circuits.
Switch2 down	Circuits served by Switch2 will be lost. (All calls lost and no new calls can arrive to these circuits) No service interruption to circuits on Switch1. Outbound calls will only be placed on Switch1's circuits.
Linkset 0 down	Take alternative route to both Switches through linkset 1; no service interruption
Linkset 1 down	Take alternative route to both Switches through linkset 0; no service interruption
LAN Switch A down	Small delay (under 5 seconds) in LAN traffic to switch over to LAN B. No service interruptions.
LAN Switch B down	Small delay (under 5 seconds) in LAN traffic to switch over to LAN B. No service interruptions.
Router A down	Router B will take over for external network traffic.
Router B down	Router A will take over for external network traffic.
LAN disconnection on VoiceGenie machine	Small delay (under 5 seconds) in LAN traffic to and from this machine to switch over to other Ethernet device. No service interruptions.
CMP Server down	No logs written to database until CMP server recovers.
MP LAN failure	Local block circuits associated with this MP.
MP failure	Outstanding calls on this Media Platforms will be lost; Local block circuits associated with this MP.

Scenarios that are not listed in the above table are dual-failures; they are require manual intervention to recover service (ie. both links down).

5 Provisioning

Cluster Management Platform provides provisioning data for SS7 deployments. Provisioning data can be modified during runtime through MC. There are two types of provisioning data introduced with SS7 and they are explained in the following subsections.

5.1 Application Provisioning

Application provisioning determines the mode of operation for all ISUP inbound calls. It looks up the Called Party Number of the incoming calls to select the mode of call and the URL of the application. The application URL is an optional parameter in this provision page, as this can be provided through Media Platform's DNIS-URL mapping.

System Management Console
VoiceGenie
A PRODUCT OF

Monitoring
Operations
Configuration
Administration

vienna.voicegenie.com | Connected to CMP Proxy
User Name: pw | v7.0.0 Cluster

Concise Config View ■

OA&M Framework

- CMP Server ■
- CMP Proxy ■
- Command Line Console ■
- System Mgmt Console ■
- VG SNMP ■

Media Platform

- Call Manager ■
- VoiceXML Interpreter ■
- Web Proxy ■

DNIS - URL Mapping

- Dialing Rules ■
- Hunt Groups ■
- Partition Definition ■
- Speech Resources ■

CCP SS7

- SS7 Call Control ■
- SS7 DNIS Mapping ■
- SS7 Circuits Mapping ■

CCP RM

- Redundancy Manager ■

SS7 DNIS - URL Mapping Configuration

To create a new DNIS Mapping record enter the DNIS, Call Type, and optionally URL. Then click on **Create**.

DNIS:

Call Type:

URL (optional):

The following entries already exist.

To update an entry change the value in the text box and click on **Update**.
Click on **Delete** to delete an entry.
To view and select targets for an entry click on **Select Target**.

SS7 DNIS - URL Mapping Entry ID: 11

DNIS:

Call Type:

URL (optional):

Enter the DNIS (Called Party Number), Call Type, and optional URL for the entry and click "Create" button. The entry will be added to the bottom of the list. Click "Select Target" to select the CCP server to target the provisioning entry.

There are three call types: normal, early media, and answer transfer. Please refer to ISUP mode of operation for details.

Suffix wildcarding is supported for application provisioning, and "*" is the wildcard character. Ordering of the entries does not matter as wildcarding is limited. A more specific DNIS string has higher precedence over less specific strings. Here is an example:

```
<entry id="1" type="103">  
* NORMAL  
</entry>  
<entry id="2" type="103">  
1416736* EARLYMEDIA http://www.voicegenie.com/earlymedia.vxml  
</entry>  
<entry id="3" type="103">  
14167360000 ANSWERXFER http://www.voicegenie.com/transfer.vxml  
</entry>
```

If the incoming call is 14167360000, id=3 is selected.

If the incoming call is 141673612345678, id=2 is selected.

If the incoming call is 14167351234, id=1 is selected with no specific VoiceXML dialog.

5.2 Voice Circuits Provisioning

Voice trunks are connected to the Media Platforms and the CCP servers must know the mapping of trunks into ISUP circuits. The SS7 Circuit ID/Media Platform Mapping Configuration page allows the appropriate mapping to be entered:

System Management Console

Monitoring Operations Configuration Administration

vienna.voicegenie.com | Connected to CMP Proxy User Name: pw | v7.0.0 Cluster

Concise Config View ■

OA&M Framework

- CMP Server ■
- CMP Proxy ■
- Command Line Console ■
- System Mgmt Console ■
- VG SNMP ■

Media Platform

- Call Manager ■
- VoiceXML Interpreter ■
- Web Proxy ■

DNIS - URL Mapping

- Dialing Rules ■
- Hunt Groups ■
- Partition Definition ■
- Speech Resources ■

CCP SS7

- SS7 Call Control ■
- SS7 DNIS Mapping ■
- SS7 Circuits Mapping ■

SS7 Circuit ID/Media Platform Mapping Configuration

The following table lists all platforms with or without Circuits Provisioning entry. To add a new SS7 Circuits Provisioning entry enter the trunk ID, number of circuits and start CIC. Then click on *Add*.

To update an entry change the value in the text box and click on *Update*.

To delete an entry click on *Delete*.

To view and select targets for an entry click on *Select Target*.

Entry ID: 6			
Network ID: 19	Platform: 10.0.0.87		
Trunk ID	# Circuits	Start CIC	
<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="button" value="Add"/>
2	30	31	<input type="button" value="Remove"/>
1	30	1	<input type="button" value="Remove"/>
<input type="button" value="Update"/>		<input type="button" value="Delete"/>	
<input type="button" value="Select Target"/>			

This page only shows a list of all available Media Platforms within the CMP cluster. Enter the voice trunk number connected to the Media Platform, number of circuits (24 for T1 and 30 for E1), and the start CIC into the text boxes and click "Add" button. Trunks can be removed from a call manager if necessary. To modify the entries, click "Update". Click "Select Target" to ensure the provisioning data is propagated to the CCP server.

5.3 RM Cluster Mapping Provisioning

An RM Cluster defines a group of CCP-RM processes running within a CMP cluster. Each CCP-RM determines the type of redundant service that is installed on each machine; it finds the redundant services by viewing the by looking through the list of server components on configured on its server.

Note that one CCP-RM can only join a single RM cluster. In the case of SS7 redundancy, only one RM cluster should be used. For each CCP-RM installed, add them into the same RM cluster and create a member string to identify the member on the network. The following diagram shows that the RM resources use its IP address and port 9801 as the member string. The port number is equivalent to "RMComm.TCPBondingLocalPort = 9801" from the CCP-RM configuration.

System Management Console



Monitoring Operations Configuration Administration

vienna.voicegenie.com | Connected to CMP Proxy

User Name: pw |v7.0.0 Cluster

Concise Config View ■

OA&M Framework

- CMP Server ■
- CMP Proxy ■

- Command Line Console ■
- System Mgmt Console ■
- VG SNMP ■

Media Platform

- Call Manager ■
- VoiceXML Interpreter ■
- Web Proxy ■

- DNIS - URL Mapping ■
- Dialing Rules ■
- Hunt Groups ■
- Partition Definition ■
- Speech Resources ■

CCP SS7

- SS7 Call Control ■
- SS7 DNIS Mapping ■
- SS7 Circuits Mapping ■

GCP RM

- Redundancy Manager ■
- RM Cluster Mapping ■

RM Cluster Mapping Configuration

To create a new Redundancy Manager Cluster, click on *Create RM Cluster*.

To add a member to an entry, choose the Redundancy Manager from the drop down list, entry member string and click on Add.
Click on Remove to remove a member from a RM Cluster entry.
Click on Delete to delete an entry.

Redundancy Manager Cluster ID: 5			
Cluster Members			
RM:	Member ID	Member String	
10.0.0.104 (13)	1	10.0.0.104:9801	<input type="button" value="Remove"/>
10.0.0.169 (10)	2	10.0.0.169:9801	<input type="button" value="Remove"/>
<input type="button" value="Delete"/>			

Whenever a member is added to a cluster, all members will be updated automatically and will be targeted automatically to all members defined in the cluster.

6 Operation, Administration, and Maintenance

6.1 Starting CCP/SS7

There are three ways to start CCP/SS7:

- 1) type "vgstart" at command prompt
- 2) Via the Command Line Console (CLC) -- type "clc" and then type "start"
- 3) Go to the Operations tab in SMC and find "Start/Stop Server" menu items. Select the SS7 server(s) and click "Start"

System Management Console

Monitoring Operations Configuration Administration

vienna.voicegenie.com | Connected to CMP Proxy User Name: pw |v7.0.0 Cluster

- Operations**
- Start/Stop Software
- Get Platform Info
- Web Cache**
- Start/Stop Cache
- View Cache
- Cache List Manager
- Perform Preload/Purge
- View Access Log
- View Event Log
- Licensing**
- Platform Licensing
- Logout

Start/Stop Server

This page allows you to *start*, *stop*, *suspend*, or *resume* services on a set of VoiceGenie servers.

Please select the Clusters or Nodes or Components that you would like to perform the operation on. Then click Start, Stop, Suspend, or Resume to perform the desired operation.

Network

- 10.0.0.104 [properties]
 - Command Line Console
 - Redundancy Manager
 - SS7 Call Control ⚠
 - System Management Console
 - VG SNMP Agent
- 10.0.0.169 [properties]
 - Command Line Console
 - Redundancy Manager
 - SS7 Call Control ⚠
 - System Management Console
 - VG SNMP Agent
- 10.0.0.87 [properties]
 - Call Manager
 - Command Line Console
 - VoiceXML Interpreter
 - Web Proxy

Start Stop Suspend Resume



Note: Server should not be stopped while the server is initializing or loading the firmware if there is a redundant system running. This will cause a redundant system to go out of sync and requires restarting both servers to recover active state.

6.2 SS7 Health Status and SNMP

Health Status string gets polled at a regular interval and the health status string shows the current statistics of the SS7 component. However, it does not show the circuit states of each individual ISUP circuits (Please see ISUP circuit snapshot for details). The health status message should look like the following:

```
Health for SS7 Call Control (ss7) on 10.0.0.94
Started: 2004-06-15/16:03:25.307
CCPSS7 status: Active
SIP Status: Active
SIP @ 5060
Calls (#IB/#OB): Current 0:0 Peak 0:0 Total 0:0
Local Point Code: 124
Destination: 123:up
Route: 123:up
Firmware status: B0:Running B4:Running
Trunk: B0:T0:up B0:T1:up B4:T0:up B4:T1:up B4:T2:down B4:T3:up
Link: B0:L0:up-uninhibited B4:L0:up-uninhibited
```

Destination, Route, Link, and Trunk above can have more than one item and new items are appended on the same line.

Redundancy status is also printed on the third and fourth line of the health status. The status represents the state of this process. It can have the following state names:

- Unavailable – service is unavailable for processing
- Unknown – service state is unknown; happens before initialization is completed
- Initialized – service is initialized; waiting to become active or standby
- Active – service is active and ready to accept calls (ISUP and SIP)
- Standby – service is on standby and can take over when another active element fails.

The health status parameters are available through SNMP get. The complete list of parameters and description is listed in Appendix A. Please also refer to the OA&M Framework User Guide for details on retrieving the information.

6.3 Redundancy Manager Health Status

CCP-RM provides the following health status string:

```
Health for Redundancy Mgr (ccprm) on 10.0.0.190
Started: 2004-05-17/18:26:08.881
My Server ID: 1
Master Server ID: 3
Slave IDs: 1 2
Running Service Managers:
CCPSS7:active Service Agent:Connected Subservice:CCPSS7:active
CCure:active SIP:active
```

The first line reports the member ID of this server within the RM cluster;

The second line reports the master server within the RM cluster; this server owns the view of all the CCP-RM that is running;

The third line reports the IDs of the slaves that are running within the RM cluster;

Each line thereafter about provides state for each redundancy service that CCP-RM provides. There are a few states available:

- UNKNOWN - Service state is not known yet.
- UNAVAILABLE – Service is unavailable.
- INITIALIZED – Service is initialized; waiting to be in either active or standby state
- GOING_ACTIVE – state of becoming active; this is a brief state while check pointing information is being uploaded before becoming active.
- ACTIVE – The service is active on this machine and handling external network traffic
- STANDBY – The service is on standby ready to become active in case of failure

The health status parameters are available through SNMP get. The complete list of parameters and description is listed in Appendix A. Please also refer to the OA&M Framework User Guide for details on retrieving the information.

6.4 Logging and SNMP Traps

Log file for SS7 Connector is located in /usr/local/ccp/logs/CMP.log.ccps7 and CMP.log.ccprm for CCP-RM. The file is rotated based on CMP Log Rotation Parameters in the configuration. A short summary is shown below; please refer to the CMP User Manual for configuration details.

Parameter	Description
cmp.log_file	This full path to the log file of the SS7 Connector Default: /usr/local/ccp-ss7/logs/CMP.log.ccps7 Rollover all log files by size or by time
cmp.size_option	Possible values: FALSE, TRUE Default: TRUE
cmp.rollover_size	The size limit, in MB, for rollover when rolling over by size Default: 10
cmp.num_rollover_files	The number of files to roll through before they are overwritten when rolling over by size Default: 5
cmp.rollover_mins	The interval of time, in minutes, between rollover when rolling over by time Default: 1440
cmp.rollover_time	The time at which the log files are rolled over when rolling over by time Default: 4:00
Email parameters	
cmp.email	If the EMAIL sink is specified, the email address be used

Logging Service parameters	Default: name@domain.com
cmp.log_sinks	Sinks that will be used by this component, possible sinks are: FILE, UPSTREAM, SYSLOG, SNMP, EMAIL
cmp.trace_flag	Default: FILE UPSTREAM Determines if logs at level log_5 (tracing/debugging) should be logged
cmp.pid_option	Possible values: FALSE, TRUE Default: FALSE Appends PID of the process to the name of the trace file so that they are not overwritten when the process restarts
cmp.metrics	Possible values: FALSE, TRUE Default: FALSE log mask for metrics data
cmp.log_0	Default: 0 1 Log mask for data logged at log level 0 Default: 11 11 11 11 111111111111 11 11 11 11

- CCP logs all major events for SS7 signaling such as state changes in:
- Physical trunks
 - Signaling Links
 - Routes
 - Destination
 - User request through command line console
 - Media Platform goes online/offline
 - Voice trunks

Logging information is generated using the VoiceGenie OA&M Framework. All logs of level Critical (LOG_0), Error (LOG_1), and Warning (LOG_2) sent upstream as SNMP Traps and to the log file. Please refer to Appendix B and C for a complete list of SNMP traps generated by SS7 Connector and CCP-RM.

Log levels for Notice (LOG_3) and Information (LOG_4) are stored in the log file.



Trace logs (LOG_5) is disabled by default. Trace is not recommended for deployment environment as trace will flood the trace files quickly and decreases system performance.

To enable trace, go to SS7 Call Control Configuration and select false for cmp.trace_flag. Click Update to submit the configuration change.

6.5 Billing

Billing records contains records of calls including transfer. These records are stored as Media Platform billing entries. Please refer to the VoiceGenie 7 Media Platform System Reference Manual for details.

6.6 Command Line Console (CLC) Operations

6.6.1 ISUP Circuit Snapshot

To provide a view of the states for each circuit through CMP, the generic command interface is used as a polling mechanism. This is chosen over the health status message because health status is polling every 20 seconds and generic command is on demand. Printing the states of each circuit uses a large string and there is no need to do periodically. Any state changes to the following output will result in a LOG_4 log entry, giving the CMP engine log it into the database to provide the users with a history of circuit state changes.

The format of the generic command is:

```
sccpss7 [service] <host>
```

This command takes no arguments. The return value will be:

```
Local Point Code: 0x123
Destination: 0x124:up
Route: 0x124:up
Link: 0:up 1:up 2:down
Trunk 1:up 2:up 3:down 4:down
ISUP Circuit Operation Status:
X=Unknown, L=Local Block, R=Remote Block,
B=Local Remote Block, U=Unblocked
CIC(1-30)=UUUUU UUUUU UUUUU UUUUU UUUUU UUUUU
CIC(31-60)=LLLLL LLLLL LLLLL LLLLL LLLLL LLLLL
ISUP Circuit Administration Status:
I=Inbound only, O=Outbound only,
D=Duplex, N=None
CIC(1-30)=DDDDD DDDDD DDDDD DDDDD DDDDD DDDDD
CIC(31-60)=OOOOO OOOOO OOOOO OOOOO OOOOO OOOOO
```

Each line of the CIC above represents a voice trunk provisioned in the Media Platform (see above section Voice Circuits Provisioning); it prints the administrative and operation state of the circuit with a single character.

6.6.2 MTP Management

Command format:

```
ss7mtpmgt [service] <host> <command> <args>
```

With the following combinations of <command> and <args>

<command>	<args>	Description
ActivateLink	LinkID	Activates datalink
DeactivateLink	LinkID	Deactivates datalink
InhibitLink	LinkID	Inhibits link
UninhibitLink	LinkID	Uninhibits link

If the <command> is not one of the commands, CCP does nothing and returns an error message.

If LinkID is not a valid link ID, CCP does nothing and returns an error message.

6.6.3 ISUP Circuit management

There will be a generic command to allow the user to block, unlock, and reset ISUP circuits:

```
ss7isupmgt [service] <host> <command> CICs
```

<command> can be one of block, unblock, reset

CICs can be a single circuit or a range of circuits of this format: 1-30 (no space)

Examples:

```
ss7isupmgt ccps7 10.0.0.169 block 1-30
```

```
ss7isupmgt ccps7 10.0.0.169 reset 34
```

If the <command> is not one of the commands, CCP does nothing and returns an error message.

If the CICs is not in a range of valid number, CCP does nothing and returns an error message.

7 Brooktrout NS700

7.1 Processes

There are three SS7 stack processes running on the CCP host. They must be running in order for CCP to run properly. The processes are monitored by CMP-Proxy and will respawn the processes in case of failure.

Process	Description
lininmpac	Layer 1 and Layer 2 configuration and maintenance
lininmtp3	MTP3 stack
lininisup	ISUP stack

7.2 Runtime debug logs

There are three levels of trace logs that can be turned on during execution of CCP.

Please note that the following logs are NOT officially supported and may change without notice.

7.2.1 API Trace

API Trace prints all Brooktrout API calls into the CCP trace file, located in /usr/local/ccp/logs/TRACE.ccp.*. Start CLC and run the following command to change the trace level:

```
tracelevel [service] <host> <lev>
```

```
where [service] is ss7  
      <host> is the host of CCP  
      <level> is trace level
```

Set the trace level to 0xf0099999 for full trace. To disable, set the level to 0x00000000.

7.2.2 MTP Trace

Log on to the CCP machine and type:

```
telnet localhost 6666  
debug 255  
quit
```

A file will be generated in /opt/TDAPI/bin/linux/mtp_dbg.txt.log. This log contains all MTP3 messages going through the MTP3 stack. To disable MTP trace telnet back to the debug port and type:

```
debug 0  
quit
```

7.2.3 ISUP Trace

Log on to the CCP machine and telnet to port 7500 for ISUP debugging:

```
debug 255  
quit
```

A file will be generated in /opt/TDAPI/bin/linux/0_isup_dbg.txt. This log contains all ISUP messages going through the ISUP stack. To disable ISUP trace, type

```
debug 0  
quit
```

7.3 Utilities

7.3.1 ISUP

/usr/local/ccp-ss7/util/isup provides a simple utility to configure a single link-set with one link. The following is the list of MTP configuration parameters:

- Local Point Code: 0x123
- Adjacent and Destination Point Code: 0x124
- Trunk 0 (closest of the LEDs on the board): E1, 1200hms, HDB3 framing, CRC is turned on
- Signaling link is on timeslot 16 of trunk 0
- One route to adjacent point code 0x124
- 60 ISUP circuits from circuit ID 1 to 60

To execute the example, run /usr/local/ccp-ss7/util/isup/runall. You should see a series of messages indicating the progress of the configuration and startup procedure.

By typing H <enter>, you can display a menu:

Key	Function
D	Allows the user to set various bit values to be used in the outgoing IAM message when making a call.
Q	Sets the handling options for incoming calls 0 = Ignore – no action taken on call arrival 1 = Reject – a call is rejected immediately by sending REL 2 = Accept – on receiving a call, an ACM is sent 3 = Answer – on receiving a call, an ACM and ANM is sent
M	Initiate an outgoing call – the user is prompted for circuit number and called party number.
C	Clear call – the user is prompted for circuit number and release cause
A	Answer Call
P	Send CPG
B	Block a circuit
U	Unblock a circuit
L	Activate an SS7 link
K	Deactivate an SS7 link
S	Send SUS message
R	Send RES message
O	Send COT message
Z	Send RSC message
H	Display this help menu
X	Exit the program

8 ISUP

There are three modes of operations. The mode is selected depending on the Called Party Number. The lookup table can be provisioned through CMP.

8.1 Normal

The Media Platform will accept the call based on the ISUP circuit identifier (CIC) and executes the VoiceXML page on the voice circuit mapped to the CIC.

8.2 Early Media

An Early Media call allows the Media Platform to start playing voice prompts before the call is connected on SS7. In terms of ISUP, audio can be played by Media Platform after ACM is sent but before ANM is sent.

There are two scenarios where Early Media can be used:

- Play audio before connect and release the call after audio is played
- Play audio before connect and answer call after prompt is completed

8.3 Answer Transfer

Answer Transfer is another special inbound call scenario to allow calling another party before the inbound call is accepted. Essentially this is an extension of the Early Media mode and will only connect the inbound call whenever the outbound call is accepted. If the outbound call rejected, the REL release cause will be transferred to the inbound call.

8.4 Transfers

8.4.1 Bridge

Bridge transfer represents the voice paths of two individual call legs that are bridged together on the Media Platform. VoiceGenie supports the <transfer> tag on VoiceXML

8.4.2 Release Link Trunk (RLT) Transfer

RLT is proprietary messaging for Nortel's DMS MSC. This allows two call legs on the SS7 Connector to be bridged on the MSC and release both calls legs from the CCP. This saves two ports on the VoiceGenie SS7 Connector when the calls are bridged on the switch.

RLT can work in answer transfer mode as well, before the inbound call is connected.

8.4.3 Ringback Transfer Mode

Ringback Transfer is a special case of Answer Transfer where VoiceGenie is acting as a media resource to play customizable ringback tone to a caller while dialing to a callee. The difference between Ringback transfer mode and answer transfer mode is the handling of ACM and CPG messages. Ringback transfer passes ACM and CPG message from the callee to the caller before the caller leg is established. The caller's backward call indicators is set to no indication until the indicator is set on the callee to ensure the audio path for the ringback tone is not played until the callee subscriber is free. If the callee rejects the call, the REL release cause will be transferred back to the caller.

9 VoiceXML Extensions

9.1 Session variables for ISUP Call Parameters

VoiceGenie defines the follow session variables for all SS7 incoming calls. Session variables are available to all VoiceXML applications within a particular session. They are declared by the interpreter and are read-only.

There are two types of values: numeric value and binary string. Numeric values are numbers as appeared in the ISUP messages. Binary strings are encoded as URL encoded strings. All variables are prefix with session.connection.protocol.isup.

Variable (Prefix: session.connection.protocol.isup)	Description (Refer to Q.763 for value details)	Type
natureofconnection.si	Nature of Connection Indicator – Satellite Indicator	Numeric
natureofconnection.cci	Nature of Connection Indicator – Continuity Check indicator	Numeric
natureofconnection.ec	Nature of Connection Indicator – Echo Control Device Indicator	Numeric
session.telephone.uui	User-to-User Information	Binary String
originalcallednumber.num	Original Called Number	Binary String
originalcallednumber.nai	Original Called Number – Nature of Address Indicator	Numeric
originalcallednumber.numberplan	Original Called Number – Numbering Plan	Numeric
originalcallednumber.pri	Original Called Number – Address Presentation Restriction Indicator	Numeric
redirectingumber.num	Redirecting Number	Binary String
redirectingnumber.nai	Redirecting Number – Nature of Address Indicator	Numeric
redirectingnumber.numberplan	Redirecting Number – Numbering Plan	Numeric
redirectingnumber.pri	Redirecting Number – Address Presentation Restriction Indicator	Numeric
redirectioninfo.ri	Redirection Information – Redirection Indicator	Numeric
redirectioninfo.orr	Redirection Information – Original Redirection Reason	Numeric
redirectioninfo.rc	Redirection Information – Redirection Counter	Numeric
redirectioninfo.rr	Redirection Information – Redirecting Reason	Numeric
session.telephone.dnis	Called Party Number	Binary String
calledpartynumber.nai	Called Party Number – Nature of Address Indicator	Numeric
calledpartynumber.numberplan	Called Party Number – Numbering Plan	Numeric
session.telephone.ani	Calling Party Number	Binary String
callingpartynumber.nai	Calling Party Number – Nature of Address Indicator	Numeric
callingpartynumber.numberplan	Calling Party Number – Numbering Plan	Numeric

callingpartynumber.si	Calling Party Number – Screening Indicator	Numeric
callingpartynumber.pri	Calling Party Number – Presentation Restriction Indicator	Numeric
callingpartycategory	Calling Party Category	Numeric
tmr	Transmission Medium Requirement	Numeric
accesstransport	Access Transport	Binary String

9.2 Early Media application

Early Media application allows the VoiceXML application to play prompts before choosing to answer the call by sending ANM message. Please refer to "ISUP mode of operations" – "Early Media" Section of the [VoiceGenie 7 SS7 Connector System Reference Guide](#) for the actual call flow. Use the <log dest="callmgr"> tag and set the content to VG/CCPSS7 EMANSWER to answer the call. The following is an example to play a prompt and then connect the call to play connected.vox.

```
<?xml version="1.0"?>
<vxml version="1.0">
<meta name="application" content="Early Media Application"/>
<form id="Welcome">
  <property name="bargein" value="false"/>
  <property name="timeout" value="3s"/>
  <block name="early media">
    <audio src="builtin:prompts/sting.vox"/>
    <audio src="builtin:prompts/welcome_to_voicegenie.vox"/>
  </block>
  <field name="connect" type="digits?minlength=8;maxlength=8">
    <noinput>
      <log dest="callmgr">VG/CCPSS7 EMANSWER</log>
      <goto nextitem="begin"/>
    </noinput>
    <filled>
      <log dest="callmgr">VG/CCPSS7 EMANSWER</log>
    </filled>
  </field>

  <field name="begin">
    <audio src="builtin:prompts/connected.vox"/>
  </field>
</form>
</vxml>
```

Note that the switch will only a one-way cut through of audio for VoiceGenie to play prompts but not receiving audio. No ASR or DTMF will be received until the call is connected with the <log> tag. It is important to build a dummy <field> that has a short no input timeout and then execute connect/disconnect. The reason is because of prompt queuing optimizations VoiceGenie provide and will run executable items before the prompts are played; having a <field> requires the application to wait for the prompt to complete before executing connect/disconnect.

9.3 Transfers

VoiceXML application can make outbound calls with the <transfer> tag. The application can provide a set of ISUP call parameters with the signalvar attribute. This attribute read from an object containing call parameters as properties of the object. The following is a simple example:

```
<var name="callvars" expr="new Object()"/>
<block>
  <assign name="callvars.callingpartycat" expr="10"/>
  <assign name="callvars.calledpartynumber.nai" expr="1"/>
</block>
<transfer signalvar="callvars" .../>
```

Similar to session variables, there are two types of values: numeric value and binary string. Numeric values are numbers as appeared in the ISUP messages. If the provided numeric value is out of the range of the size of value, then a bitmask a performed on the size of the value and enter the value into the ISUP message. Binary strings are URL encoded strings and will be decoded to be inserted into the ISUP message.

Signalvar Variable (case-insensitive)	Description (Refer to Q.763 for value details)	Type
natureofconnection.si	Nature of Connection Indicator – Satellite Indicator	Numeric
natureofconnection.ec	Nature of Connection Indicator – Echo Control Device Indicator	Numeric
callingpartycategory	Calling Party Category	Numeric
tmr	Transmission Medium Requirement	Numeric
DNIS	Called Party Number	Binary String
CalledPartyNumber.NAI	Called Party Number – Nature of Address Indicator	Numeric
CalledPartyNumber.numberplan	Called Party Number – Numbering Plan	Numeric
ANI	Calling Party Number	Binary String
CallingPartyNumber.NAI	Calling Party Number – Nature of Address Indicator	Numeric
CallingPartyNumber.numberplan	Calling Party Number – Numbering Plan	Numeric
CallingPartyNumber.SI	Calling Party Number – Screening Indicator	Numeric
CallingPartyNumber.PRI	Calling Party Number – Presentation Restriction Indicator	Numeric
redirectingnumber.num	Redirecting Number	Binary String
redirectingnumber.nai	Redirecting Number – Nature of Address Indicator	Numeric
redirectingnumber.numberplan	Redirecting Number – Numbering Plan	Numeric

redirectingnumber.pri	Redirecting Number – Address Presentation Restriction Indicator	Numeric
RedirectionInfo.RI	Redirection Information – Redirection Indicator	Numeric
RedirectionInfo.ORB	Redirection Information – Original Redirection Reason	Numeric
RedirectionInfo.RC	Redirection Information – Redirection Counter	Numeric
RedirectionInfo.RR	Redirection Information – Redirecting Reason	Numeric
OriginalCalledNumber.num	Original Called Number	Binary String
OriginalCalledNumber.NAI	Original Called Number – Nature of Address Indicator	Numeric
OriginalCalledNumber.numberplan	Original Called Number – Numbering Plan	Numeric
OriginalCalledNumber.pri	Original Called Number – Address Presentation Restriction Indicator	Numeric
AccessTransport	Access Transport	Binary String
uudata	User-to-User Information	Binary String
GenericNumber.num	Generic Number	Binary String
GenericNumber.NQI	Generic Number – Number Qualifier Indicator	Numeric
GenericNumber.NAI	Generic Number – Nature of Address Indicator	Numeric
GenericNumber.SI	Generic Number – Screening Indicator	Numeric
GenericNumber.PRI	Generic Number – Address Presentation Restriction Indicator	Numeric
GenericNumber.numberplan	Generic Number – Numbering Plan	Numeric
GenericNumber.NI	Generic Number – Number Incomplete	Numeric

9.4 RLT Transfer

To initiate an RLT transfer, use the <transfer> tag with the following attributes set:
bridge=false
type=supervised

9.5 Disconnecting the call with cause code

To disconnect the call with a specific cause code, use the reasonexpr attribute of the <disconnect> tag.

The following example disconnects the call with a specific cause of code after announcement is played.

```
<?xml version="1.0"?>
<vxml version="1.0">
<meta name="application" content="Early Media Application"/>
  <form id="Welcome">
    <property name="bargein" value="false"/>
    <property name="timeout" value="3s"/>
    <var name="cause" value="16"/>
    <block name="early media">
      <audio src="networkannouncement.vox"/>
    </block>
    <field name="disconnectfield" type="digits?minlength=8;maxlength=8">
      <noinput>
        <disconnect reasonexpr="cause"/>
      </noinput>
      <filled>
        <disconnect reasonexpr="cause"/>
      </filled>
    </field>
  </form>
</vxml>
```