Genesys Application Note

Alcatel OXE R11
With Genesys VPS 8.1.7

Version 1.0

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1 Introduction

This **Integration testing of Voice Platform Solution 817 with Alcatel OXE** document presents the integration test results for testing 8.1.7 Voice Platform Solution (VPS) with Alcatel OXE R11 performed by the QA Team as a part of 8.1.7 VPS project. The main objective of testing is to confirm the ability of VPS 8.1.7 work with Alcatel OXE according to requirements.

2 Genesys Software Versions

Table 1: Genesys software versions used

Component	Version
SIP Server	8.1.101.10
Stat Server	8.1.000.08
URS	8.1.200.16
VP Media Control Platform (MCP)	8.1.700.59
VP Resource Manager (RM)	8.1.700.71
Switch (GW)	Alcatel OXE: R11.0-k1.400-25-f-fr-c0s1
VP Supplementary Gateway	8.1.700.02
VP Call Control Platform (CCP)	8.1.700.60
CTI-Connector	8.1.700.54
IVR-TServer	8.1.001.01
DB Server	8.1.000.01
Configuration Server	8.1.000.16
IRD	8.1.200.14
Message Server	8.1.000.01
Composer	8.1.003.73

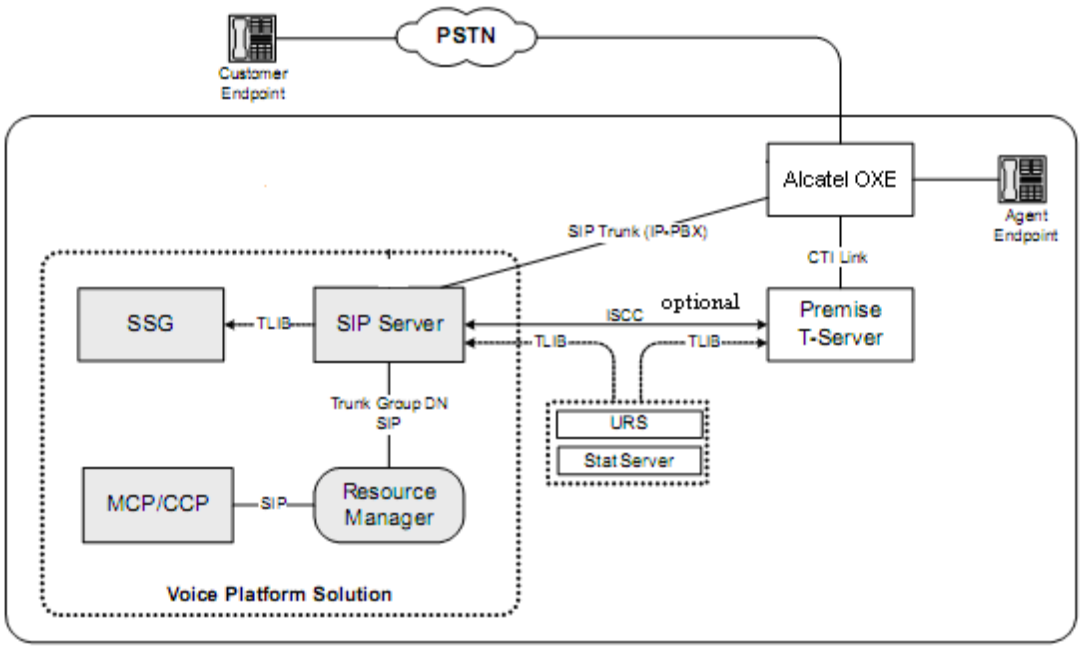
3 Third-Party Components

Alcatel OXE: OmniPCX Enterprise R11.0-k1.400-25-f-fr-c0s1

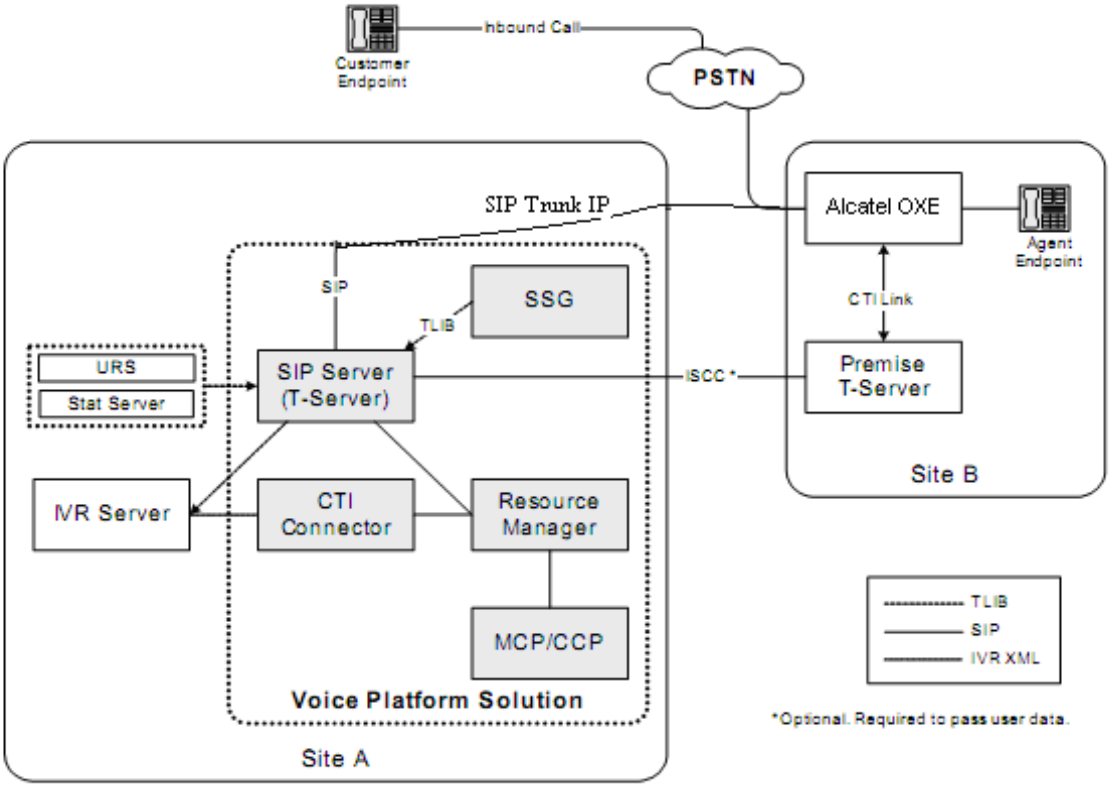
Endpoints: Epi(SIP)Phone, Xlite

4 Test Setup

4.1 Set Up



Deployment without IVR Server

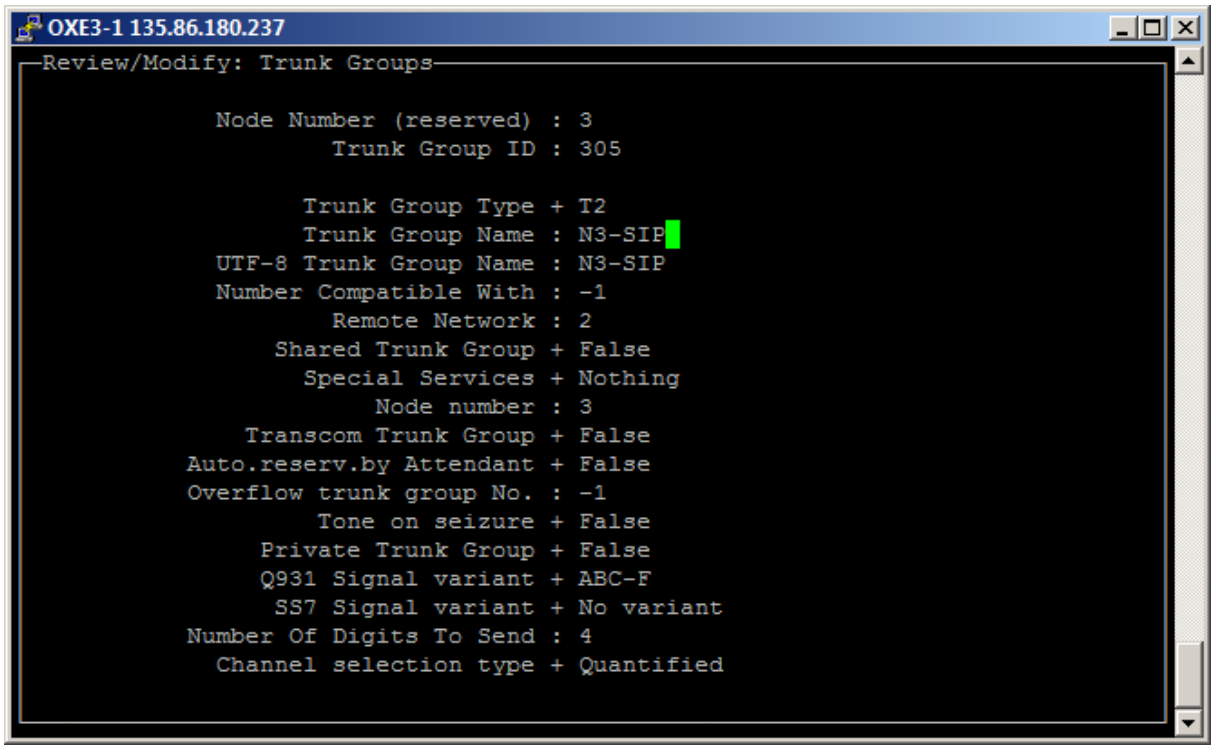


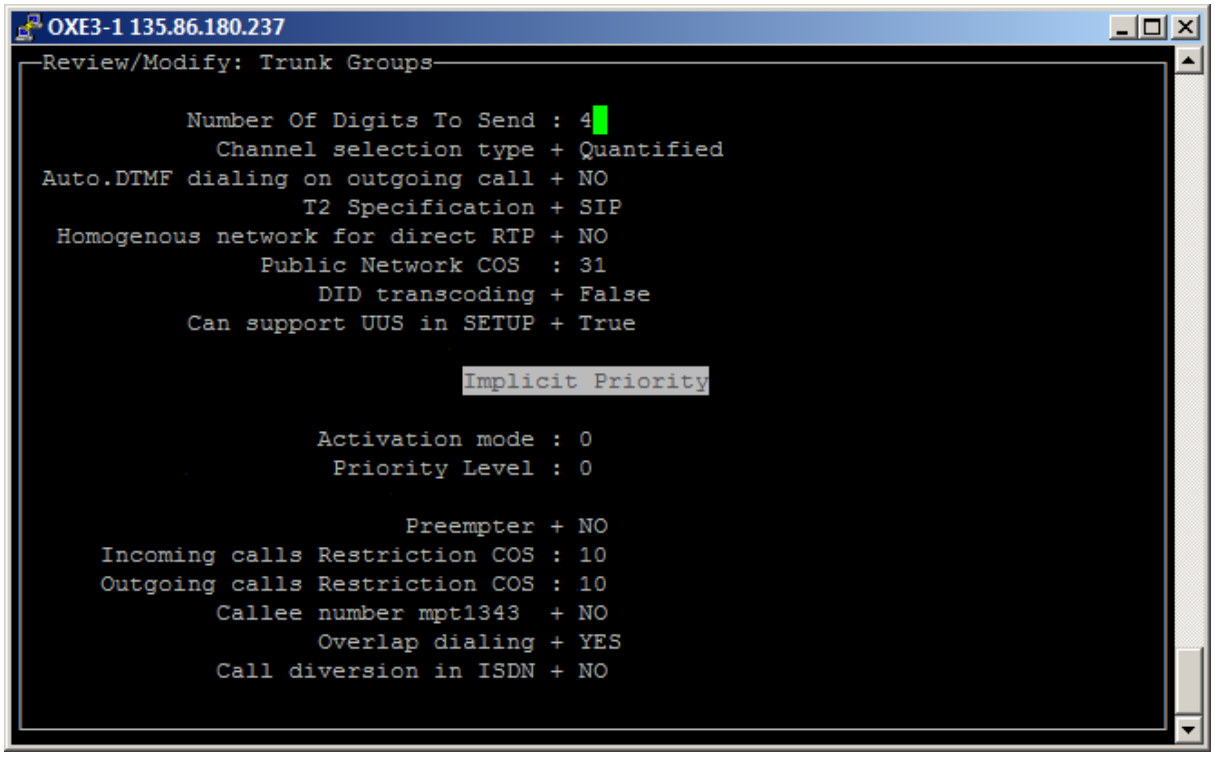
Deployment with IVR Server

4.2 Alcatel OXE Configuration

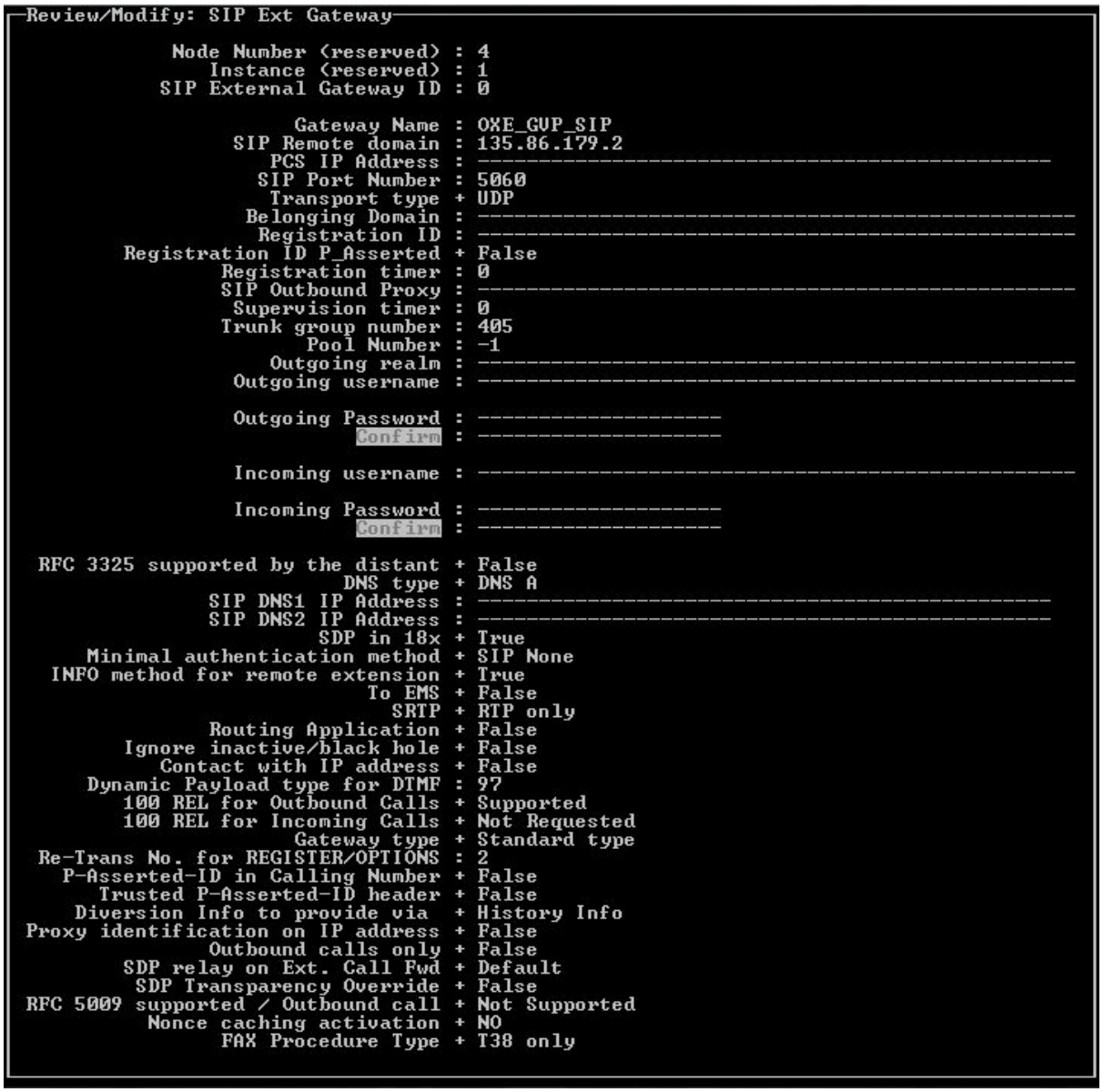
4.2.1 Alcatel OXE Trunk to SIP Server Configuration

- Create a SIP Trunk Group.

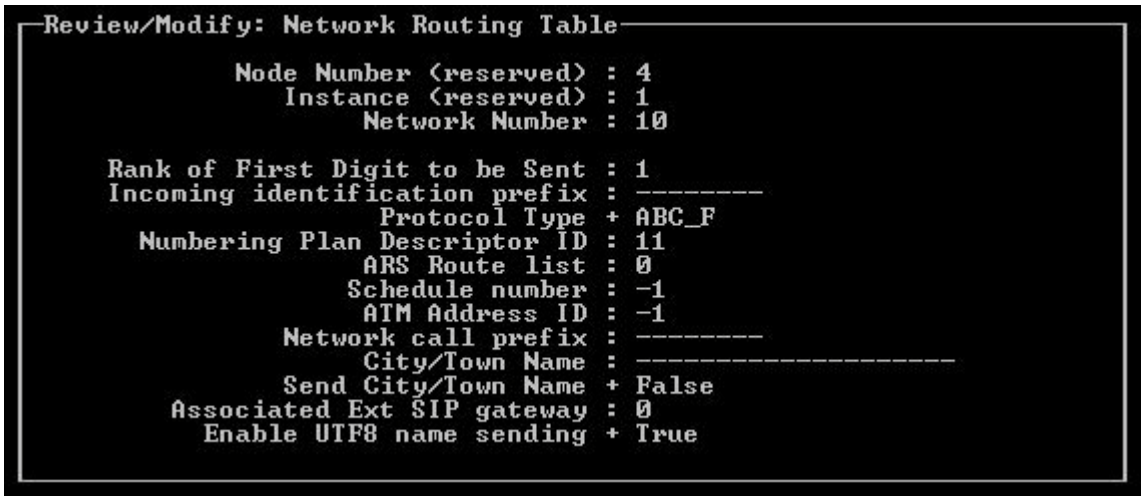




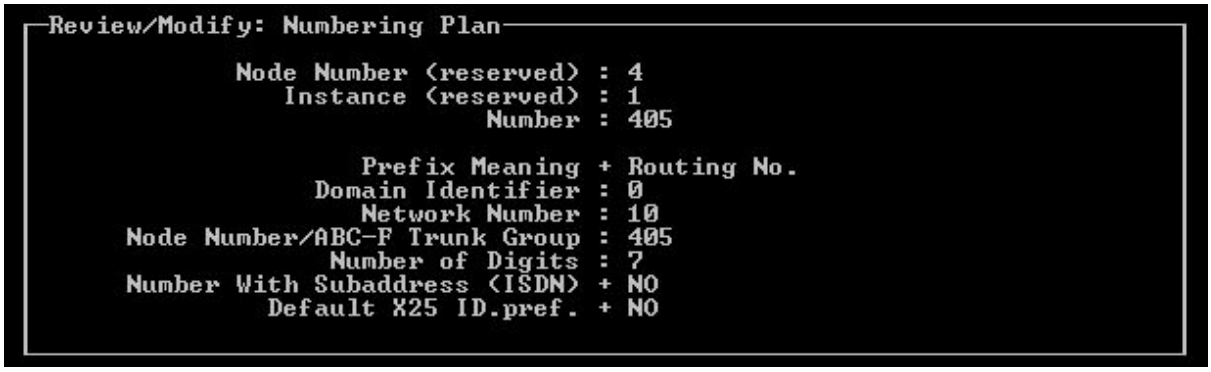
- In MGR, go to SIP->SIP Ext Gateway and point the SIP Trunk Group to the Genesys SIP Server host and port. In the example below we are using SIP External Gateway 0.



- In MGR, go to TRANSLATOR -> Network Routing Table and create an entry that points to the above SIP External Gateway (0 in our case).



- In MGR, go to TRANSLATOR -> Numbering Plan and create a prefix that points to the Trunk Group (405, for example) and the Network Routing Table created above (10, for example).
  The Number will be the prefix required to seize the trunk. The Number of Digits will be the length of the DNIS, including the prefix, provided by the OXE. In our example, we are passing a seven-digit number with the first three digits being formed on the prefix (405). In SIP Server, we can register seven-digit SIP endpoints with prefix 405. Or, in the SIP Server Switch we can have an Inbound Trunk device with prefix 405 replaced by nothing, and have four-digit SIP endpoints.



Additional Configuration on the OXE switch to allow reINVITE from SIP Server

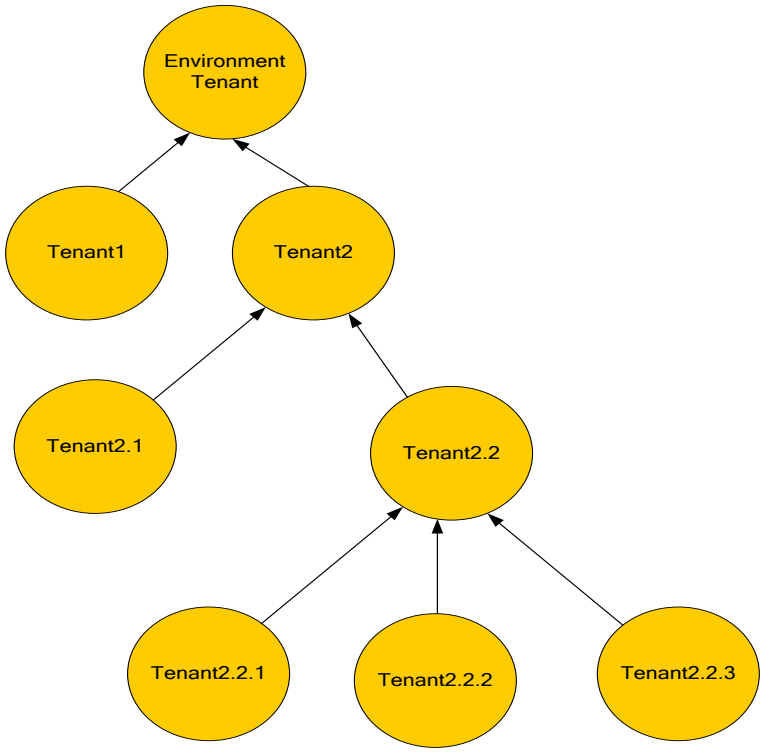
In MGR (manager), set the following options:

- SIP --> SIP Proxy ->Only authenticated incoming calls = False
- system --> Other system param --> SIP Parameters --> Via Header_ Inbound Calls Routing = TRUE

4.3 Hierarchical Multi-Tenancy

The key concept of the tree structure in Management Framework is the introduction of a parent pointer. Each tenant object, other than the root tenant (called Environment), uses the parentTenantDBID to reference its parent tenant object.

Before the introduction of hierarchy multi-tenancy, tenancy is structured as flat tenants and the tenants exist as peers to each other. HMT and flat tenants can co-exist with each other, and multiple hierarchies can be built out of the flat tenants as multiple root-level tenants.

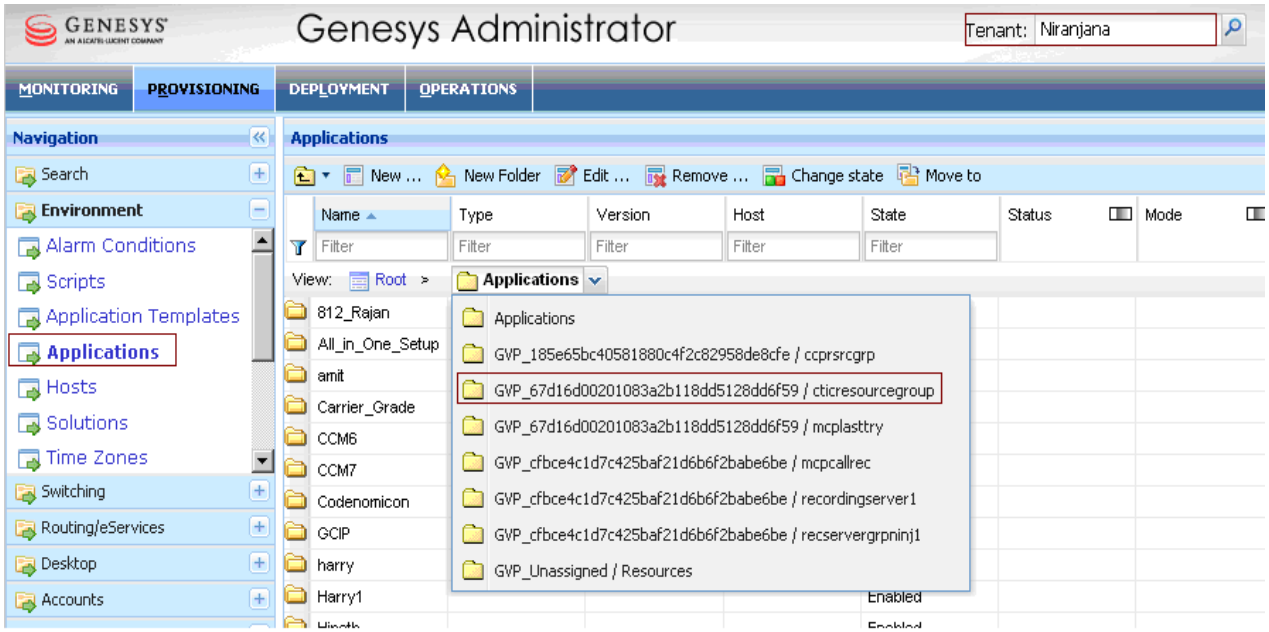


Some of the main characteristics of using a hierarchal model for GVP are as follows:

1. Inheritance of preferences and properties from the parent tenant
2. Allocation of physical resources from parent tenant to child tenants
3. Allocation of DNs from parent tenant to child tenants

While creating the Resource Group, the appropriate tenant under the Resource needs to be allocated and selected in the Tenant selector. Finally, when the Resource Group is configured, the Resource is moved under the Tenant, which can be viewed by selecting the appropriate Resource Group name in the Application window's Drop Down list (highlighted in the figure below.)

Apart from this, Resource can be "exclusively" assigned to either the parent tenant or any of the child tenants by the inclusion of "exclusive" check box during the creation of the resource groups.



4.4 VPS Components Deployment and Configuration

Refer to the GVP Deployment Guide at: http://docs.genesys.com/Documentation/GVP.

4.5 Creating Switch

Refer to the GVP Deployment Guide at: http://docs.genesys.com/Documentation/GVP.

4.6 Creating Resource Group

Create the Resource Groups for the following (refer to the GVP deployment guide if required):

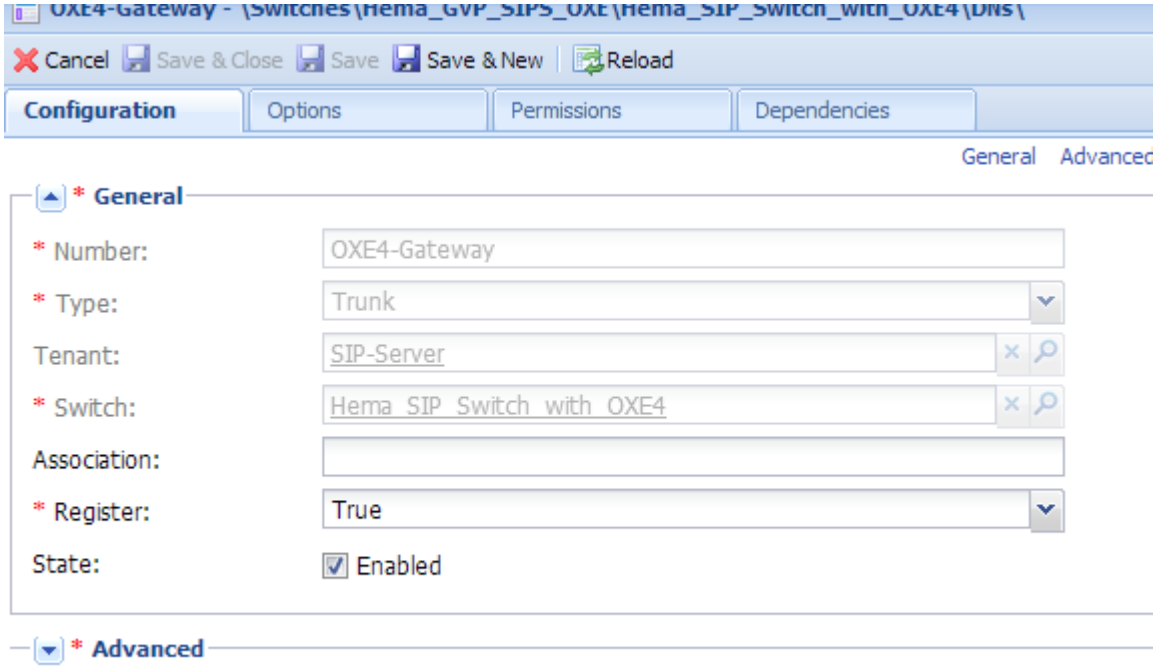
- Media Control Platform
- Call Control Platform
- CTI Connector
- Gateway

4.7 Configuration of DN in SIP Switch, Specific to OXE Interop

4.7.1 Properties of TRUNK DN to point at OXE gateway

a) Provisioning -->Switching --> Switches --> Choose the Switch --> DNs --> New Configuration:

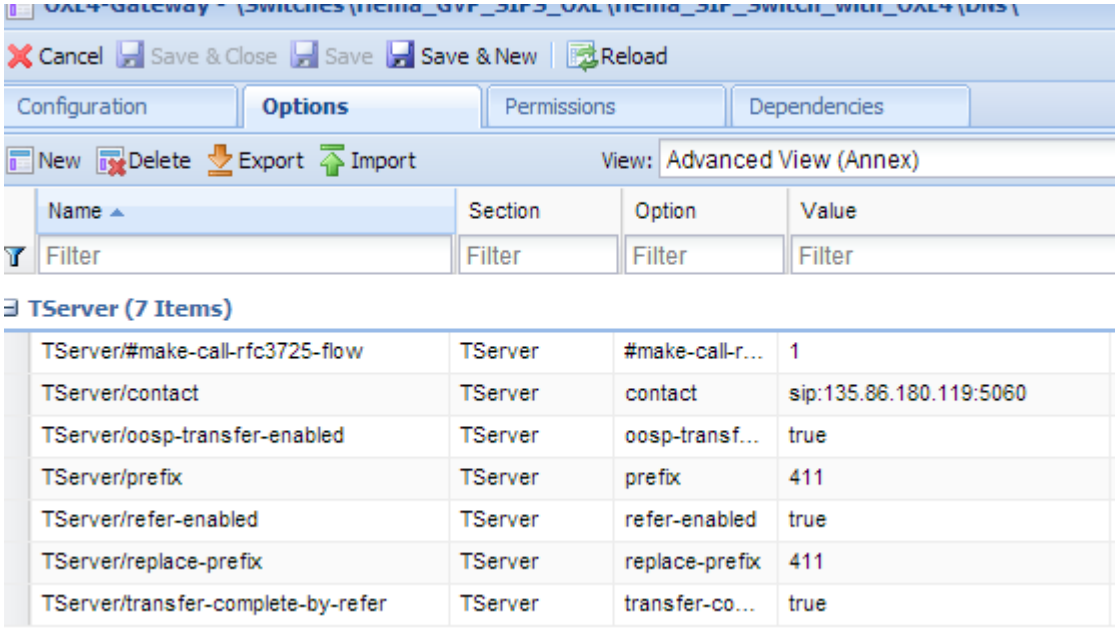
Number: Choose any number
Type: Choose the desired option from the drop down menu
Tenant: Choose the desired Tenant
Switch: The particular SIP Switch is chosen



b) Options --> New

Section:	TServer
contact:	Provide the IP address and SIP port of switch gateway
oosp-transfer-enabled:	Provide the values True/False
prefix:	411
refer-enabled:	Provide the values True/False
replace-prefix:	411

The prefix is set to 411 since the users on OXE have prefix 411.



Note: For Alcatel OXE testing, `TServer\refer-enabled` was set to `true`. However, this can be set to `false` when the user requires SIP Server to remain in the call.

For all other GVP related DNs required for testing, refer to the GVP deployment guide.

5 Test Cases Executed

Test cases written for the solution-level testing comprise primarily of positive test cases that are based on use cases, as well as a few related negative test cases.
Each test case added to the solution testing verifies some aspect of the functionality.

We have tried our best to keep each test case unique so that we can achieve maximum completeness in the testing, per the schedule of VPS 817.

Load and performance test cases are not covered, per the VPS testing.
Jira is the test management system.

5.1 Test Project

Jira Test Project was used to schedule the resources. The actual test cases are covered in test specification document: ODS-OXE-GVP8-SIPServer-TestCases.docm.

5.2 Transfer Scenarios

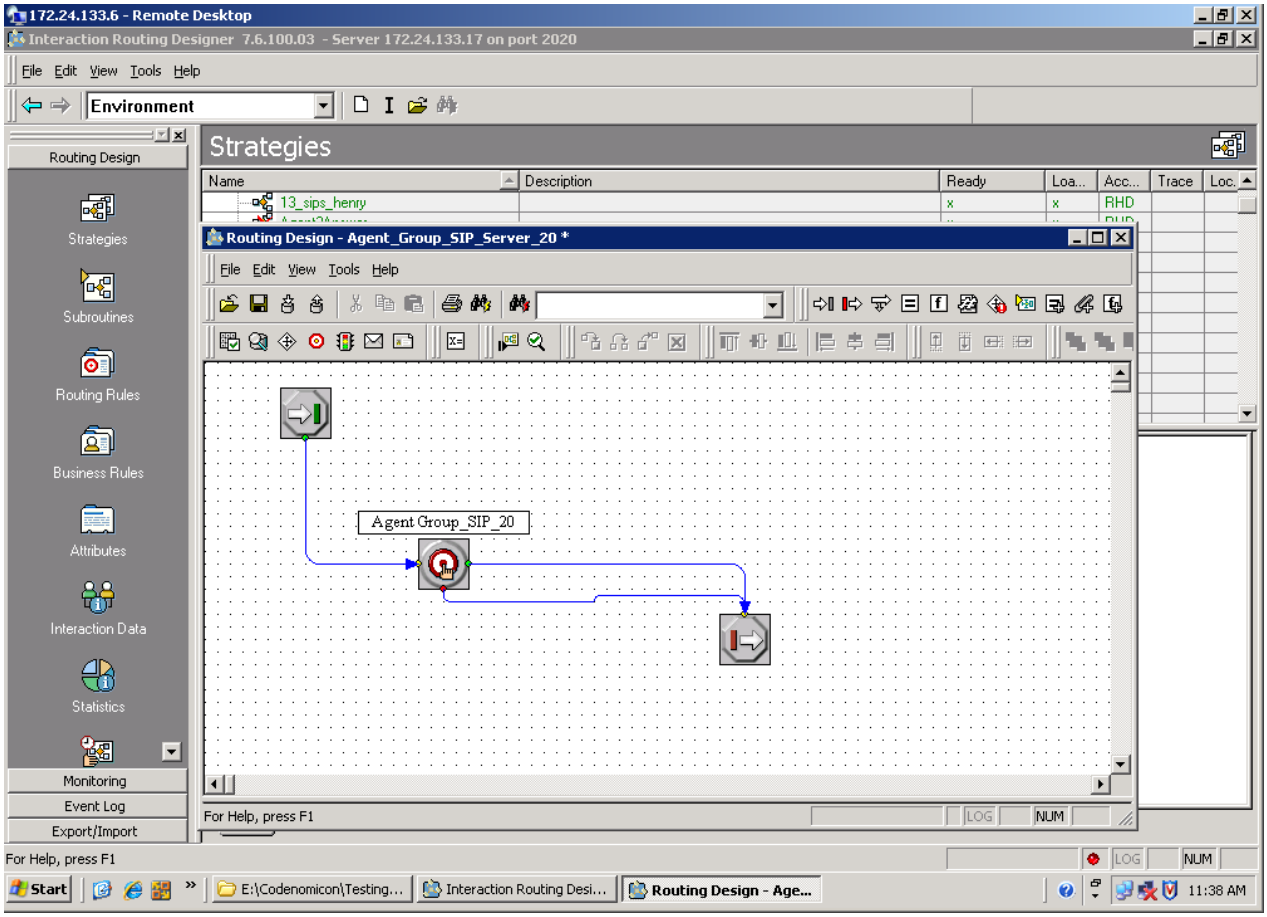
VXML Bridge, Blind, and Consultation transfers were tested. Transfers were tested to both RP, a direct extension on SIP Server and to an extension on the OXE switch.

5.3 IVR Centric Scenarios

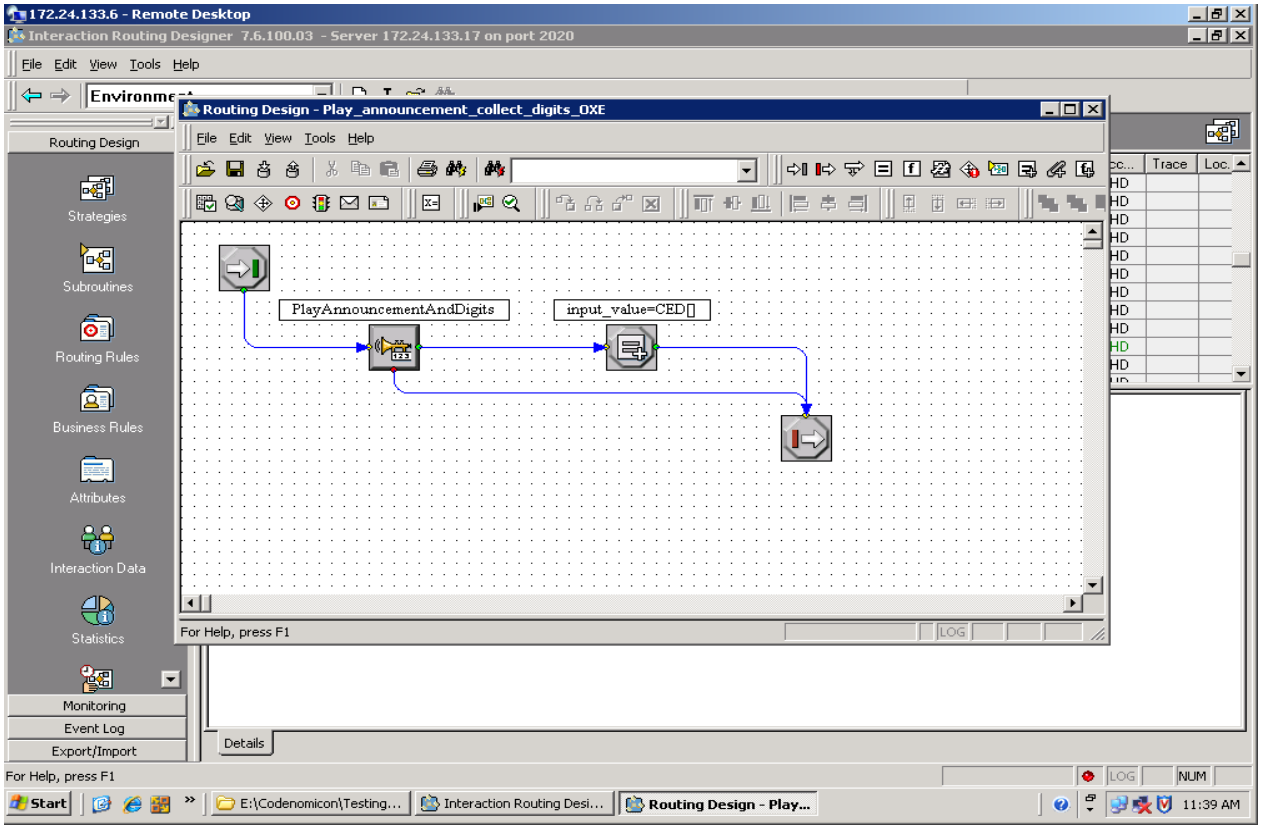
In this configuration, the IVR Server is involved in Behind-Mode with respect to SIP Server; simple inbound calls and playing of supported treatments were tested. SIP Server is added to the connections of I-Server for IVR centric scenarios.

5.4 Routing Strategy Used For Testing

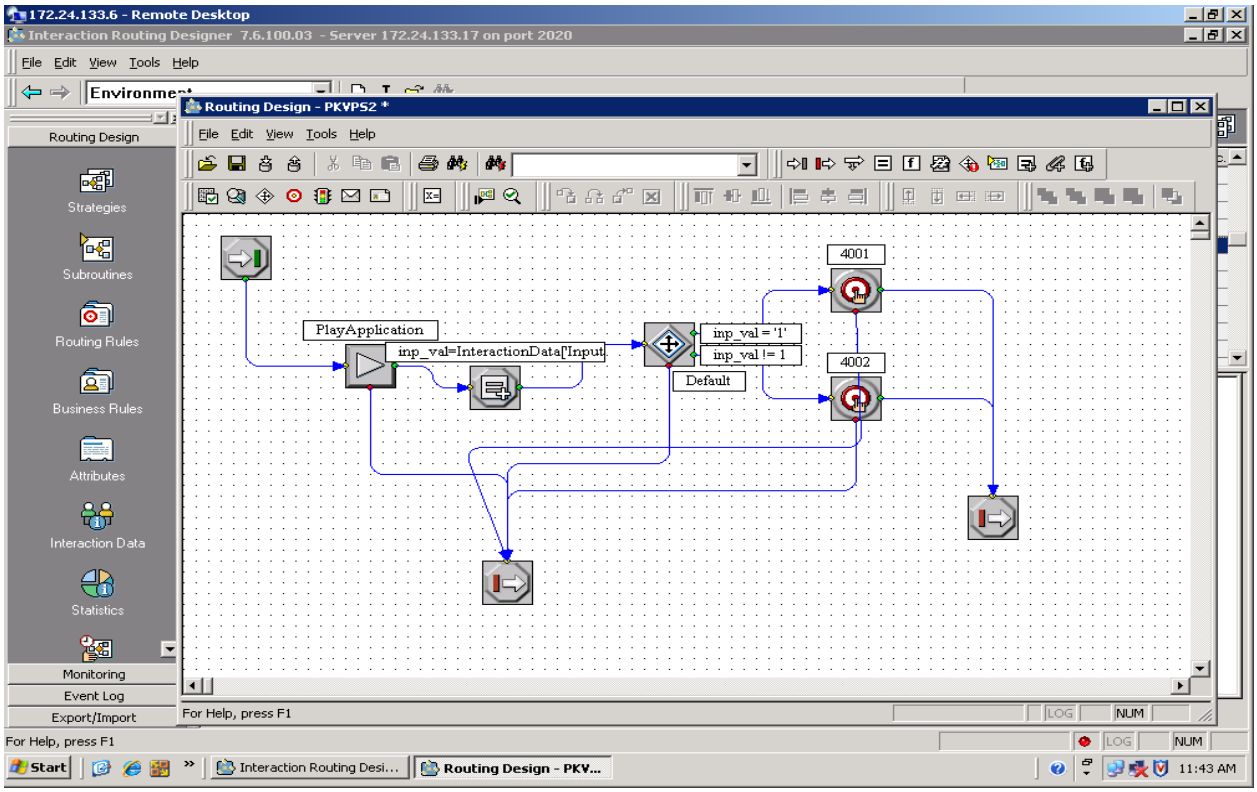
5.4.1 Routing to Agent Group on SIP Server



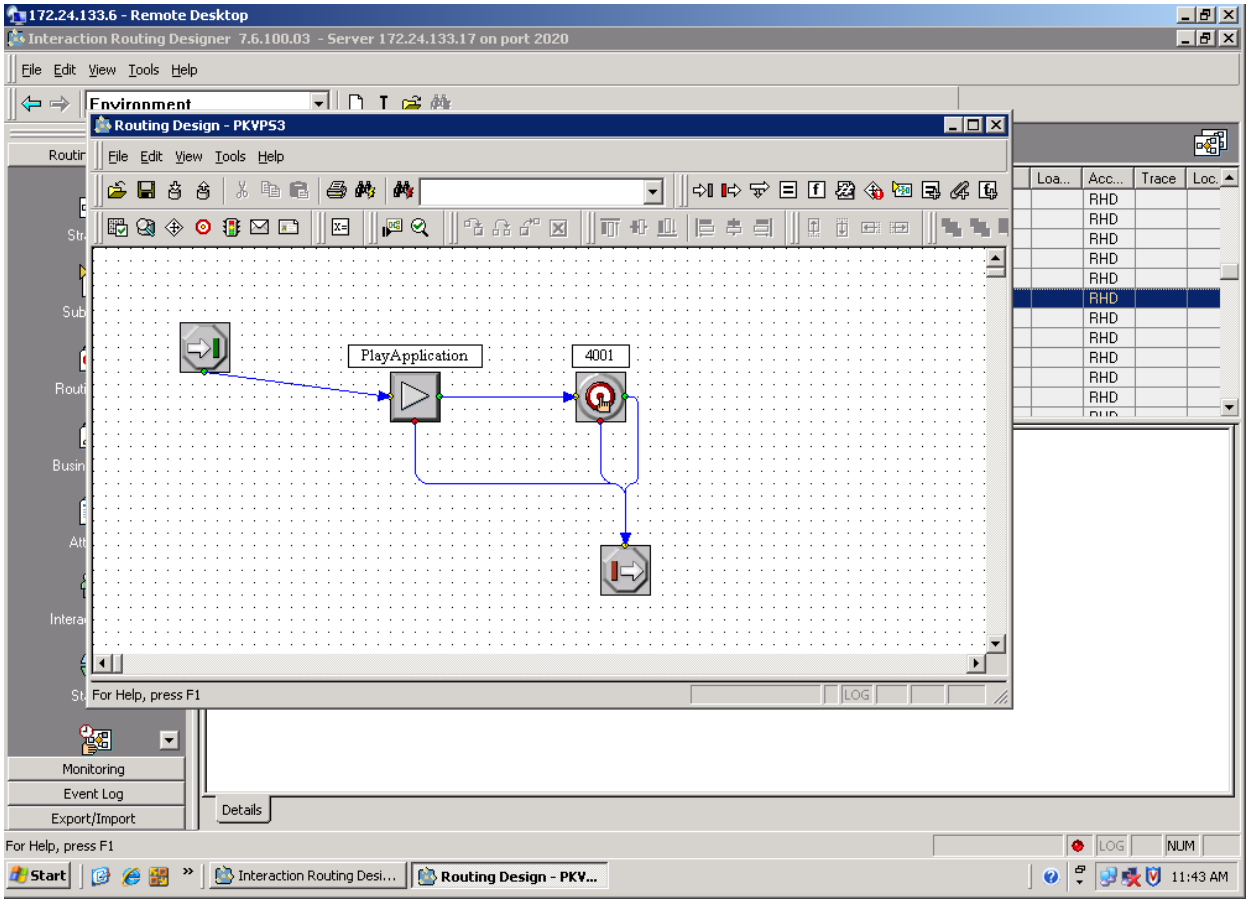
5.4.2 Play Announcement and Collect Digits



5.4.3 Play Application and Routing to Agent depending on the Input Value

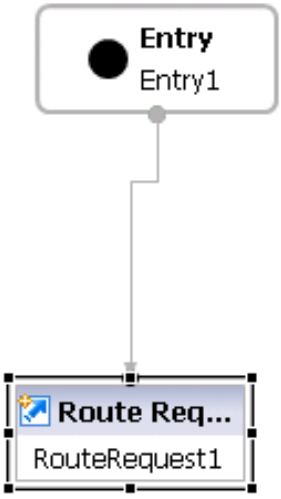


5.4.4 Play Application and Route to Agent



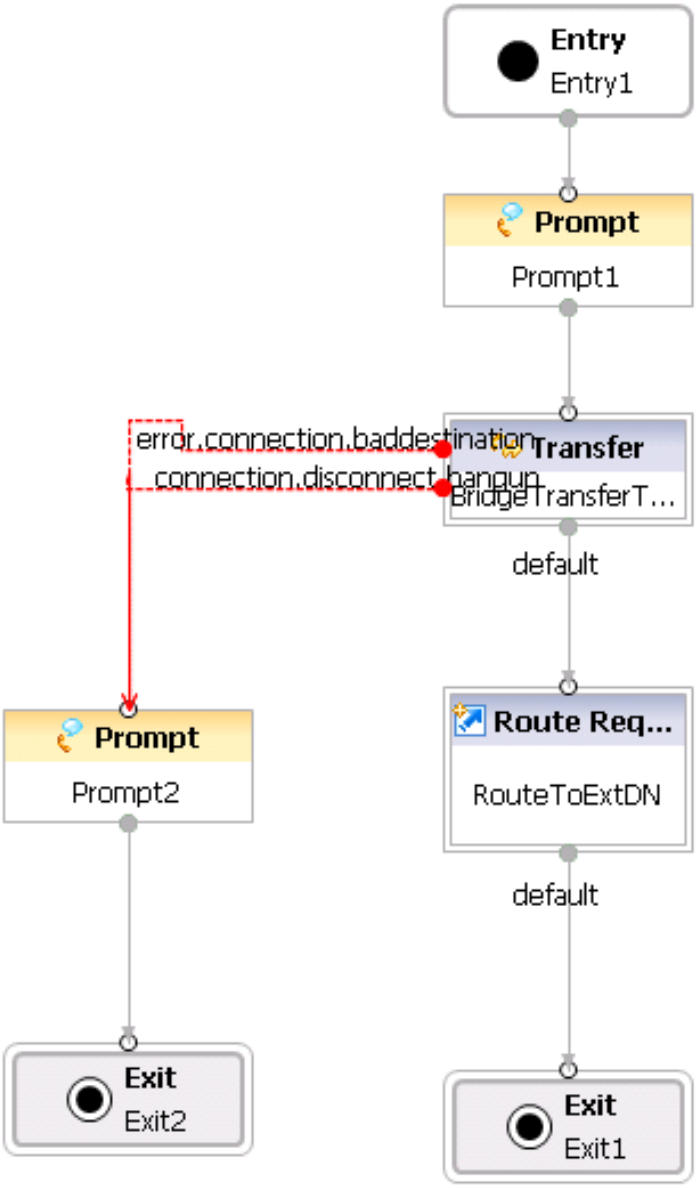
5.5 VXML Application Used For Testing

5.5.1 Blind Transfer Application with Route Request block designed using Composer



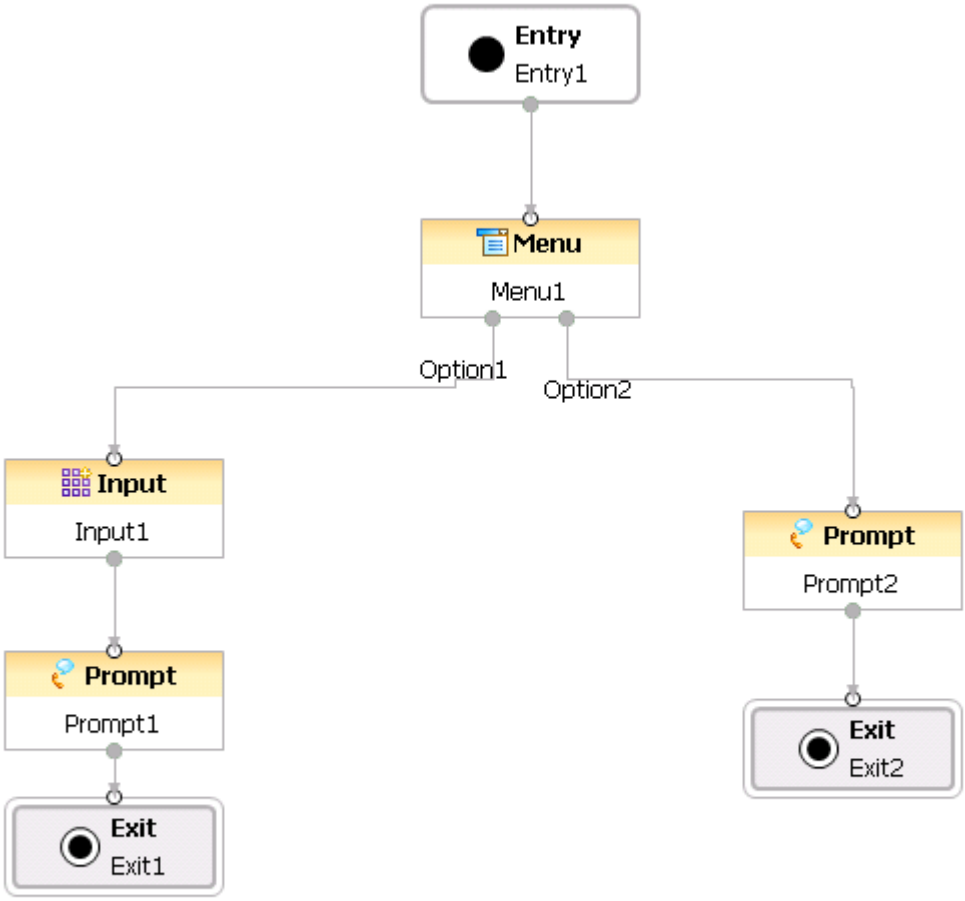
Description: The VXML application consists of Entry1 block, Prompt1 block with a text message and Route Request1 block with the destination provisioned as Route Point at SIP Server, Transfer Type=Blind, Method=REFER.

5.5.2 Bridge Transfer Application with RouteRequest block designed using Composer



Description: The VXML application consists of Entry1 block, Prompt1 block with a text message and Route Request1 block with the destination provisioned as Route Point at SIP Server, Transfer Type=Bridge, Method=Bridge, Prompt2 block with a text message and Exit block.

5.5.3 DTMF Application with Input block designed using Composer



Description: The VXML application consists of Entry1 block; Prompt1 block with a text message, Input1 block prompts the user to enter the DTMF Input, Prompt2 block with a text message and Exit1 block.

6 Troubleshooting

6.1 Capturing SIP Traces on OXE

- CH options
  tuner km ctr
  trc init
  mtracer –a-u-g >> *<to a log trace file>*
  tuner +cpu +cpl +at +xtr +s +tr hybrid=on
  actbdg all=off fct=on sip=on abcf=on rtp=on voip=on ei=on cnx=on comp=on acdv2=on mtracer –aug

- Sipmotor trace
  motortrace 6
  traced >> *<to a log trace file>*

7 Glossary and Acronyms

Term	Definition
CCP	Genesys GVP Call Control Platform Application
CTI	Computer Telephony Integration
DNIS	Dialed Number Identification Service
DTMF	Dual Tone Multie Frequency
GVP	Genesys Voice Portal Suite
GW	Gateway
HTTP	Hypertext Transfer Protocol
IP	Internet Protocol
IRD	Genesys Interaction Routing Designer Application
MCP	Genesys GVP Media Control Platform
PSTN	Public System Telephone Network
RM	Genesys GVP Resource Manager
RP	Genesys Routing Point Device
RTP	Real-Time Transport Protocol
SDP	Session Description Protocol
SIP	Session Initiation Protocol
SIP-S	Genesys SIP Server Application
SS	Genesys Stat Server Application
SSG	Genesys GVP Supplementary Services Gateway Application
URS	Genesys Universal Routing Server Application