GENESYS
GVALIDATED
INTEGRATION

WHITE PAPER

Genesys Gvalidated Application Integration

SANGOMA

09-2009

Contents

1.0 Introduction

Contact center managers everywhere are always looking for means to increase the efficiency and productivity of human capital and technical assets under their control. The recent advent of IP Telephony (IPT) based on open standards such as SIP provides a powerful opportunity to ease and extend communications in the contact center, yielding significant reduction in operating costs and increasing revenue opportunities. However, to achieve the desired results, it is critical that the promise of IPT be delivered in a progressive, non-disruptive manner. Sangoma Technologies and Genesys have partnered together to deliver seamless and highly optimized IP Telephony solutions for the Dynamic Contact Center. This document introduces the architecture, benefits and technical integration of the Sangoma/Genesys solutions.

1.1 About Sangoma

Sangoma is the premium provider of voice and data connectivity components for software-based communication applications. Sangoma's data cards, voice cards, gateways and connectivity software are used in leading PBX, IVR, Contact Center and data communication applications worldwide. The product line represents a comprehensive toolset for deploying cost-effective, powerful, and flexible software communication applications.

Sangoma acquired Paraxip Technologies in 2008, which offers an optimized suite of products that some of the most reputable contact centers in the world are currently using to improve their efficiency.

- IP-PBX: Sangoma supports a large variety of Open Source and commercial IP-PBX software. A number of vendors offer pre-packaged IP-PBX appliances based on these offerings and Sangoma's award-winning voice cards.
- IVR and Call Centers: Sangoma also supports a wide variety of Interactive Voice Response (IVR) and Contact Center applications. Sangoma has acquired Paraxip Technologies in July 2008, which offers the NetBorder suite of products that some of the most reputable contact centers in the world are currently using to improve their efficiency.
- Data Connectivity: Sangoma began designing data communication cards in 1984. Today, leading application vendors worldwide continue to trust and rely on Sangoma's cards for interfacing with X.25, frame relay, ADSL and other types of data networks.

1.2 Partnership With Genesys

With its connectivity software products, Sangoma/Paraxip has been enabling Genesys' SIP-based Contact Center Solutions since 2004. Sangoma adds smart connectivity to IP-based Genesys applications such as inbound, outbound and IVR. As such, it allows Genesys IP Contact Center solutions to connect to legacy PSTN equipment, to be enhanced with superior Call Progress Detection for automating outbound or notification campaigns, to seamlessly integrate remote agents into the IP infrastructure, and/or to have Genesys IP deployed in configurations without a costly softswitch.

1.3 Business Benefits

The benefits of standards-based IP Telephony in the Contact Center are now well known and accepted. However, Genesys IP is not deployed in isolation from other devices and networks, but rather designed to be integrated into the existing environment, offering a seamless, progressive adoption of IP infrastructure that supports the value-added inbound, outbound and IVR applications.

Sangoma's NetBorder solution allows for smart connectivity of Genesys applications to external devices and networks. Below are the main use cases for the integration along with a highlight of their benefits.

1.3.1 NetBorder Call Analyzer for Genesys OCS or Genesys Proactive

Call Progress Analysis (CPA), also called Call Progress Detection (CPD), is a generic term for signal processing algorithms that operate on audio during call setup. The goal of CPA is to determine the nature of the callee or the outcome of call setup to an external network (traditional or IP). Specifically, when a call or session is being established, the caller or initiator is interested in knowing if someone answered, if the line is busy, etc. When the caller is an automated application such as Genesys OCS or Proactive, CPA algorithms are used to perform the classification automatically. The NetBorder Call Analyzer, in combination with Genesys outbound solutions, ensure fast and accurate automated call classification which automatically translate in better efficiency of agents and higher quality customer interactions. The Sangoma/Genesys combined outbound solution can save millions of dollars in yearly annual operating costs to a 500 agent outbound contact center.

1.3.2 NetBorder Agent Bridge for Remote Agents

Contact Center managers are progressively deploying IP Telephony in their contact centers to drive significant operational savings. For concerns of costs and/or quality of service, it is sometimes not possible to provide IP phones to remote agents, at least in the initial phases of a project. Often, customers require the ability to deliver calls to any agent anywhere on the PSTN or behind legacy PBXs, at an agent-provided PSTN phone number.

The Sangoma NetBorder Agent Bridge solution complements Genesys SIP Server to enable remote location behind legacy PBX's or remote agents using regular landline phones to appear as SIP agents inside the customer network. This brings the immediate benefits of virtualization and infrastructure consolidation, but without the immediate requirement to deploy IP endpoints all the way to the far edges. The solution brings the savings of open-standards IP telephony, but without the worries of quality of service or the IP phone roll-out costs.

1.3.3 TDM-to-IP Gateway for Genesys SIP Server without a soft-switch

NetBorder can also be used as a TDM-to-IP gateway, enabling the integration of legacy telephony network or devices with Genesys SIP Server. Of particular importance are Sangoma's proxy capabilities built into its gateways, which allows Genesys to be deployed without a softswitch, therefore bringing the real benefits of IP, but without some of the more costly components.

2.0 Solution Overview

Sangoma's NetBorder product line enables IP telephony applications such as IPBX, IVRs, Routing, Outbound Dialers, etc. to seamlessly and cost effectively connect to external devices and networks. The NetBorder Software Suite is a VoIP Session and Media Controller, a new class of product for VoIP deployments in the enterprise. NetBorder is comprised of three major software subsystems:

- Session Controller and Mediation
- Media Services
- Applications and Network Connectivity

Each software subsystem has its own scripting and OAM components, for increased flexibility and manageability. Figure 1 below depicts the NetBorder System Architecture. In NetBorder, these software subsystems combine to provide the necessary network connectivity, signalling proxy, signalling harmonization, media processing, media conversion and call control scripting to remove the complexity from SIP applications.

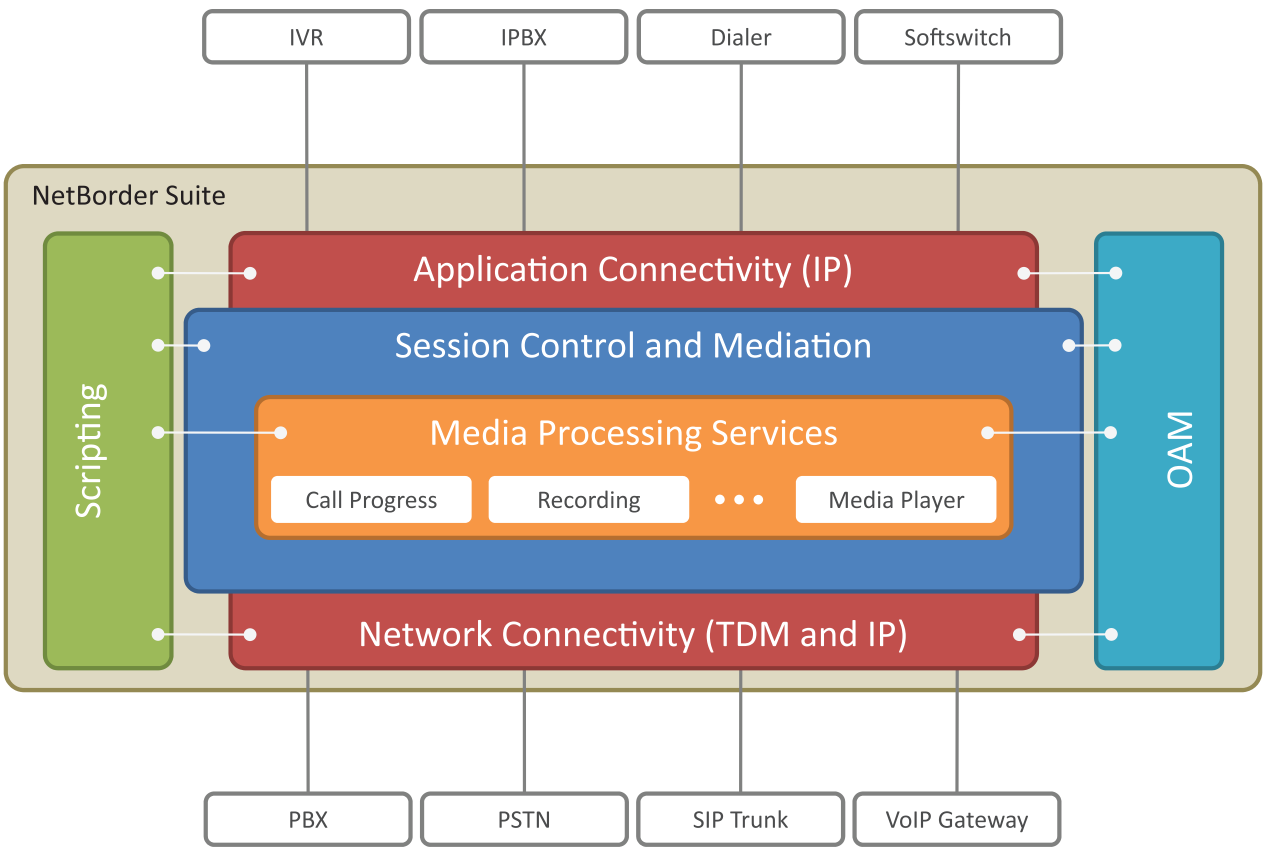

Figure 1 : Sangoma NetBorder Suite System Architecture

The Session Control and Mediation services include functions such as:

- SIP mediation services
- SIP-to-PSTN gateway services
- Call routing and call control admission
- Proxy, registrar, etc.

The media processing services provide value-added functions for the contact center, such as:

- Call Progress Analysis of outbound calls
- Call recording
- Codec harmonization (transcoding)
- Fax relay, etc.

Sangoma's Netborder integrates with the Genesys Suite, through Genesys SIP Server as shown in Figure 2 below. In this case, SIP Server front-ends the Genesys Framework. This integration is done via the Application Connectivity module of NetBorder.

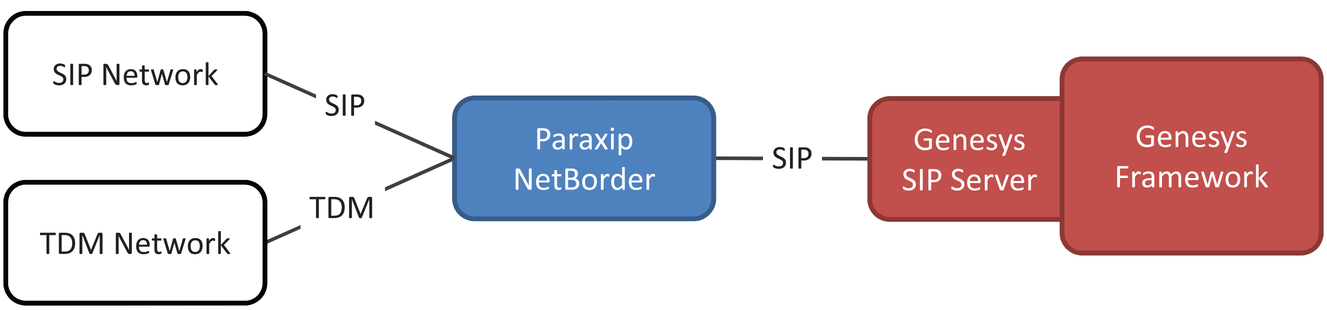

Figure 2 : Sangoma NetBorder integration with Genesys SIP Server

3.0 Application Features And Technical Benefits

3.1 Use Case 1: Outbound Call Progress Analysis

Nowadays, it is crucial for corporations to incorporate proactive or outbound strategies in their Contact Centers in order to actively reach their customer base and prospects, improve customer satisfaction and retention, as well, as the overall performance of the resources at hand.

As Contact Centers are moving to VoIP infrastructure – for instance, by using SIP trunks from service providers, as opposed to costly leased lines from the Telcos – Call Center operators are left without efficient solutions. Not only does NetBorder provide a reliable way to interconnect SIP based outbound dialers, it also provides a much different approach to Call Progress Analysis (the process of automating call classification for the purpose of determining call treatment in and outbound or proactive campaign).

Traditional Call Progress Analysis implementations rely on simple rule-based algorithms which provide sub-optimal accuracy of speed of processing, leading to significant inefficiencies in operations. NetBorder breakthrough approach uses statistical models based on Neural Networks to represent the potential outcomes of an outbound call attempt. The approach identifies much better the call patterns that represent call progress events and provides CPA results with far superior accuracy and flexibility compared to traditional approaches:

- ☑ **Improved Accuracy and Response Time**
  Benchmarks have shown that the statistical approach reduces error rates by up to 70% compared to leading traditional products while reducing the call transfer timing by 40%.

- ☑ **Resilience Against Various Network Conditions**
  This approach provides superior robustness against volume variations, background noises and other network conditions. This is highly important because applications are being now deployed in more heterogeneous conditions than ever before. Also, since contact centers are starting to use automated dialing for a much different set of applications (callback, automated notification proactive campaigns) as opposed to a single application, the CPA platform must allow for per–call optimization of the calling parameters.

- ☑ **Streamlined Tuning Process**
  The statistical approach helps streamline the tuning process. Data can be gathered from a deployment and used to re-train the statistical models. Tuning data from all sites helps to improve the baseline performance over time.

- ☑ **Dynamic Operations**
  One of the major difficulties of call progress analysis is to set the tradeoffs between response times vs. CPA accuracy. The more time that is allocated to the CPA algorithms, the more accurate (on average) the results

will be. Of course, if the response time is too slow, the person called might hang-up, or the system could violate regulations. The statistical approach provides an alternative model (see figure below). It is possible for the automated dialing application to select the particular operation point on a per call basis. This means that the accuracy vs. response time trade-off is locally optimized to provide the best possible results given the specific applications, campaign, customers or any other dynamic condition currently observed in the contact center.

3.2 Use Case 2: Remote Agent Solution

The migration of a Contact Center to an IP infrastructure is often a project that occurs over multiple phases. Typically, one or two sites are identified to be converted first, and then the rollout progresses slowly toward the edge (smaller external sites and home agents). The business case for integrating the smaller sites with existing key systems as well as home agents into the general IP infrastructure is very compelling, but the logistics of deploying quality VoIP networks or endpoints to these locations make it very difficult to move quickly and reliably. The NetBorder Agent Bridge solution enables remote agents (whether they would be at a remote site behind a legacy PBX or at home using their regular phone service) to place a call into the IP Contact Center to be placed into the local IP agent queue. A remote agent appears to the contact center as an IP phone that just registered into the network. Agents can thus be managed identically to IP agents within the contact center's brick walls.

The main benefits of this solution are:

- Seamless and instantaneous extension of the IP Contact Center to any agent, anywhere without having to deploy IP phones and QoS networks to the edge.
- Fast call connection time since agents are 'bridged' in and out of conversations with customers as calls get routed to them. This is of particular importance for outbound or callback applications, but can also have a beneficial effect on inbound call handling time if the agents are home-based.
- Integrated reporting. The remote agents appear the same way as 'brick and mortar' agents, yielding a unified view of the agents in the configuration and reports.

3.3 Use Case 3: TDM-to-IP Gateway in Genesys Configurations Without a Softswitch

Used as a TDM-to-IP gateway, NetBorder seamlessly connects legacy telephony network and devices with SIP-based applications. Sangoma's gateway capabilities are unique in that they are provided as a software application on top of commercial hardware. Furthermore, they are the only gateway solutions in the industry specifically targeted at enabling sophisticated Contact Center environments. Sangoma's gateway solutions deliver a broad set of features, all available and controllable via its SIP interface, including:

- Call Progress Analysis for outbound dialing applications
- Wide range of supported Call Transfers
- Transparent and customizable call routing via XML-based routing engine
- Load balancing and Failover routing
- Speech optimization features such as eVAD and Echo Cancellation
- Open System Solution based on standard, off the shelf computing platforms

Essentially, the gateway's customizable load balancing and failover routing capabilities offer a viable alternative to a costly softswitch configuration, therefore saving significant costs and effort in planning a transition to IP.

4.0 Integration With Genesys

All interactions between Sangoma's edge connectivity software and the Genesys framework are done through Genesys SIP Server, using the standard SIP protocol, with select proprietary extensions as appropriate. Figure 3 below provides a high level overview. This section describes the architecture for the different use cases introduced in earlier sections.

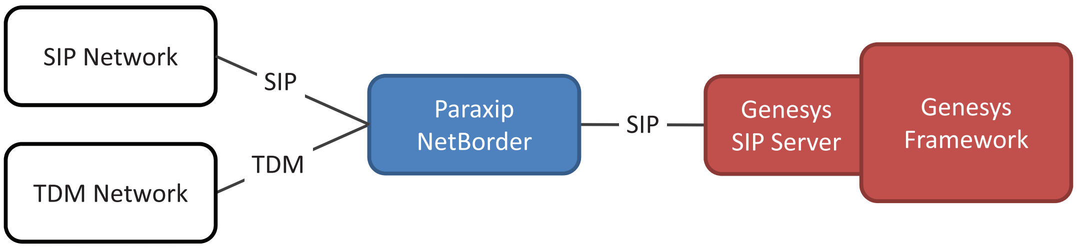

Figure 3 : Sangoma NetBorder integration with Genesys SIP Server

4.1 Architecture and Call Flows

4.1.1 Use Case 1: Outbound Using Sangoma's CPA

Figure 4 below shows the main components involved in outbound dialing. The NetBorder Call Analyzer is configured to be placed between the Genesys OCS application and the outside network.

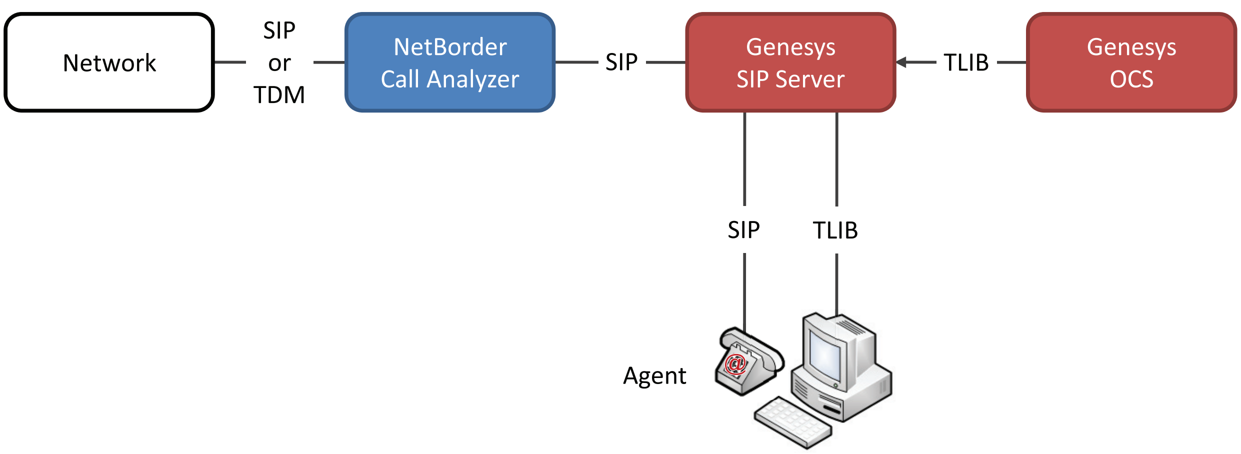

Figure 4: Netborder Call Analyzer with Genesys OCS

The call flow diagram (Figure 5) illustrates the interaction between the Genesys SIP-based outbound solution (the combination of Genesys OCS and SIP Server – here labelled SIP Server for simplicity), NetBorder and a SIP trunking network or gateway.

Call flow description:

1. Agents registers (using SIP REGISTER) to notify SIP Server that it is available to receive calls
2. OCS initiates outbound call via SIP Server
3. NetBorder receives the request and re-originates the outbound call towards the VoIP provider network or local gateway. Note that as soon as media flows, NetBorder CPA engine starts processing call information
4. NetBorder passes CPA results to SIP Server which relays it to OCS. Genesys looks for next available agent SIP Server invites selected agent
5. After proper acknowledgements, NetBorder re-invites the VoIP provider's network or local gateway to point the session towards the selected agent
6. Media flows between agent and customer

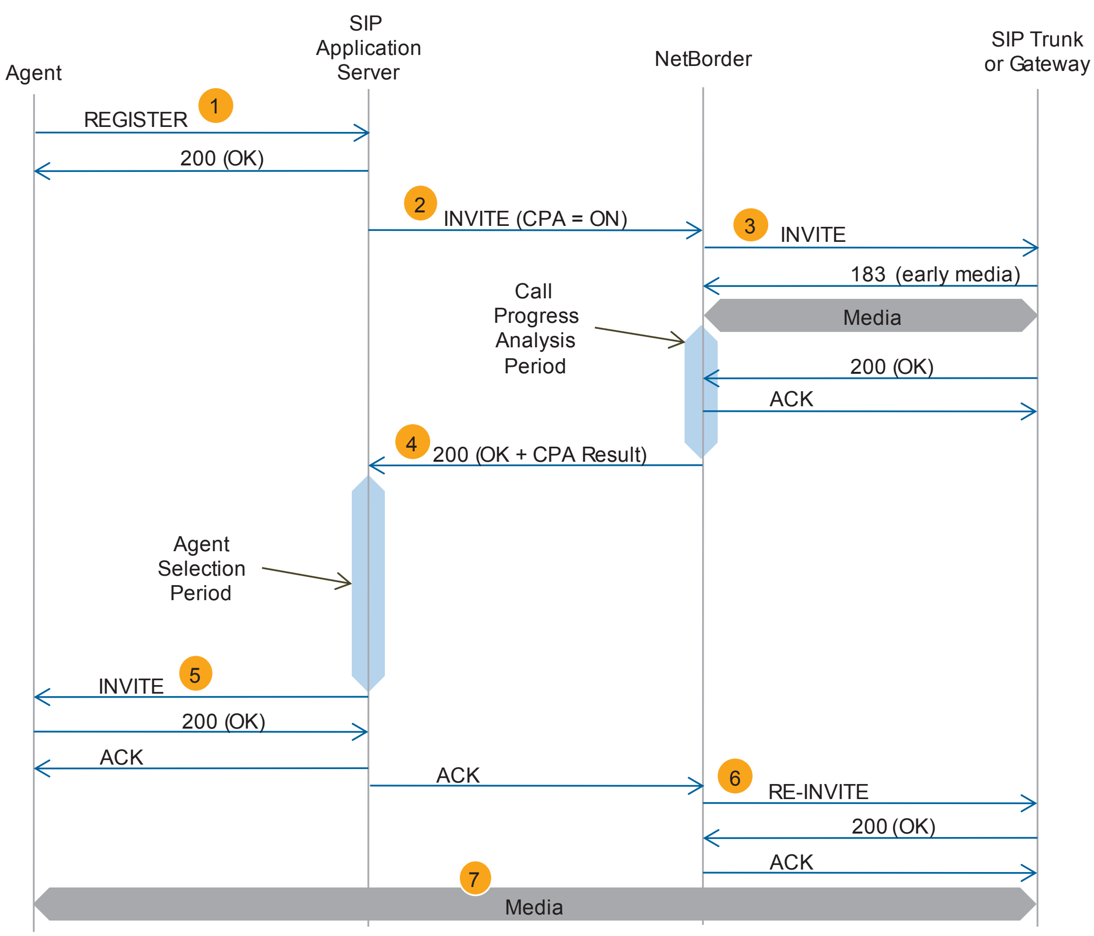

Figure 5: Outbound Call Flow Integration

Throughout the call, NetBorder stays in line with the SIP call control path, but gets out of the loop on the audio stream as soon as call progress is completed. As such, the areas labelled "Call Progress Analysis Period" should be as short as possible – while maintaining proper accuracy –as to meet regulations but as well as avoiding the dreaded dead-air scenario where customers simply hang-up before it is possible to render the service.

4.1.2 Use Case 2: Agent Bridge for Remote Agents

This section provides more details with respect to the interaction of the Sangoma NetBorder Agent Bridge and other SIP end-points in the hosted infrastructure. Figure 6 below shows the main components. The next sub-sections will use this diagram to illustrate the interaction between NetBorder and the rest of the architecture.

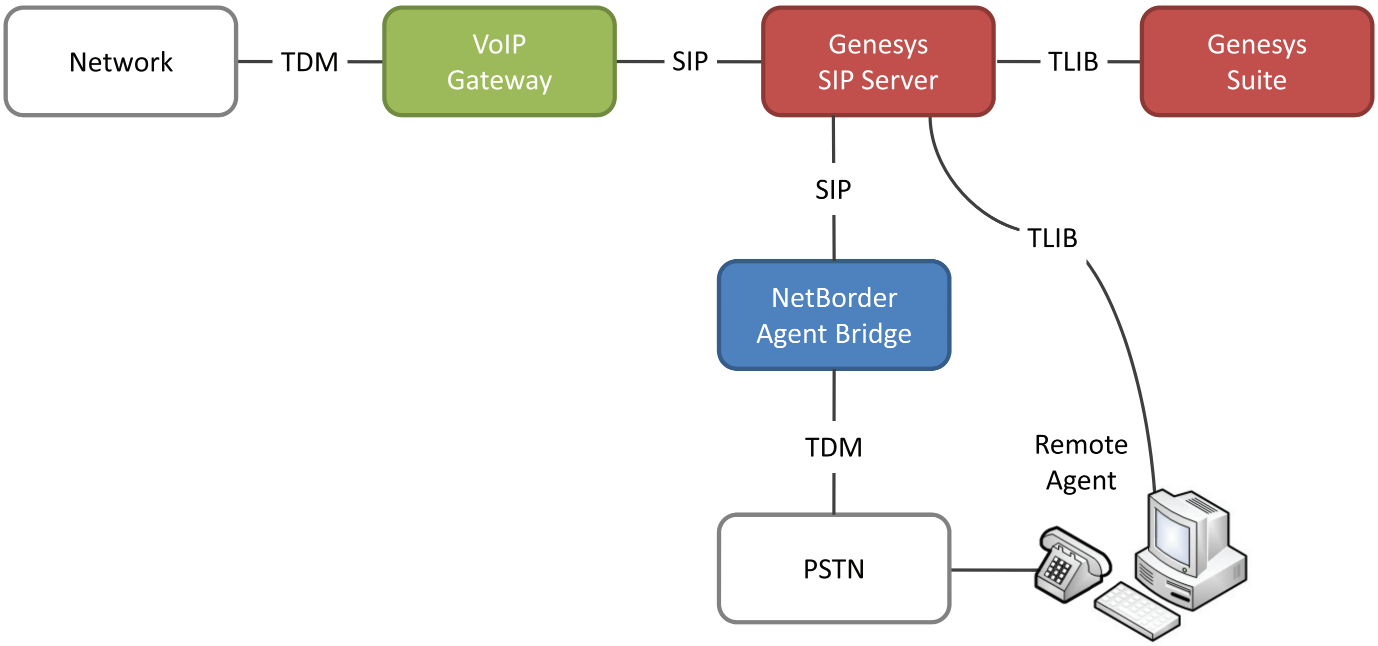

Figure 6: NetBorder Agent Bridge Integration with Genesys SIP Server

4.1.2.1 Remote Agent Login

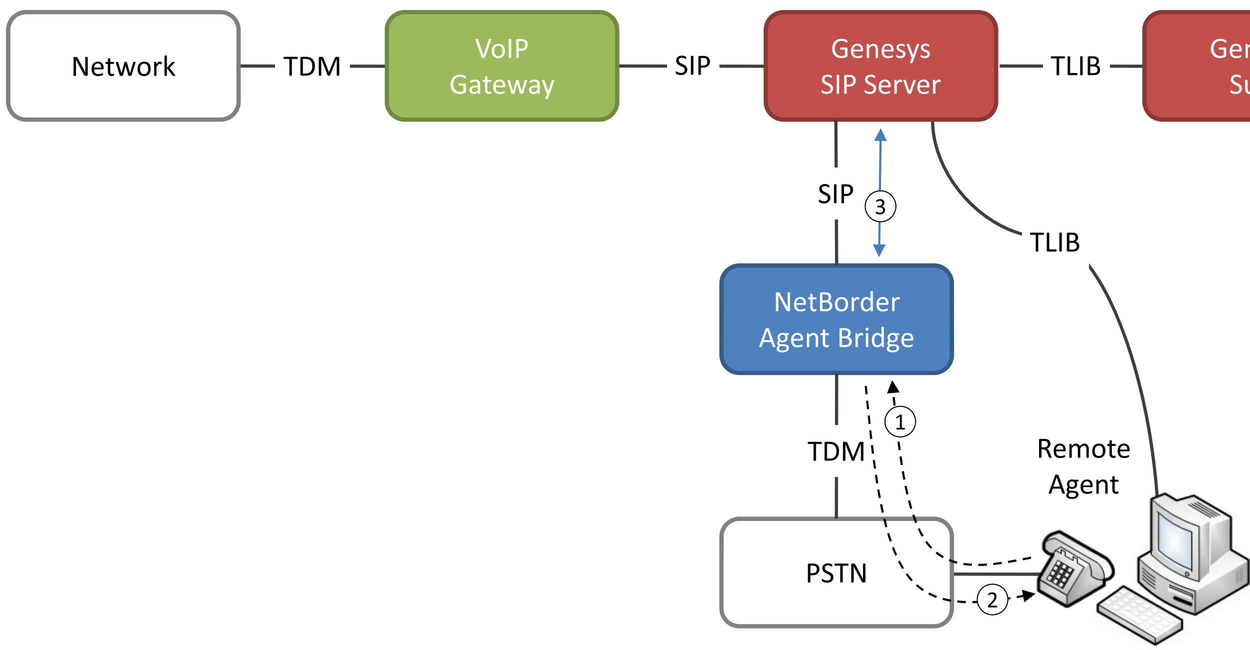

Figure 7 : Remote Agent Log-in

Call Flow Description:

1. Remote Agent calls designated phone number and is connected to NetBorder Agent Bridge.
2. NetBorder accepts call and plays a prompt for the Agent identification digits. Agent enters digits.
3. NetBorder sends SIP REGISTER message to SIP Server, registering the Agent with the ID that was entered in step [2].

Notes:

- NetBorder supports the HTTP digest authentication method
- NetBorder does not hold end-user data to perform authentication
- The TDM leg of the call is parked (or nailed-up) permanently between NetBorder and the remote Agent.

4.1.2.2 Inbound Call Routing to Remote Agent

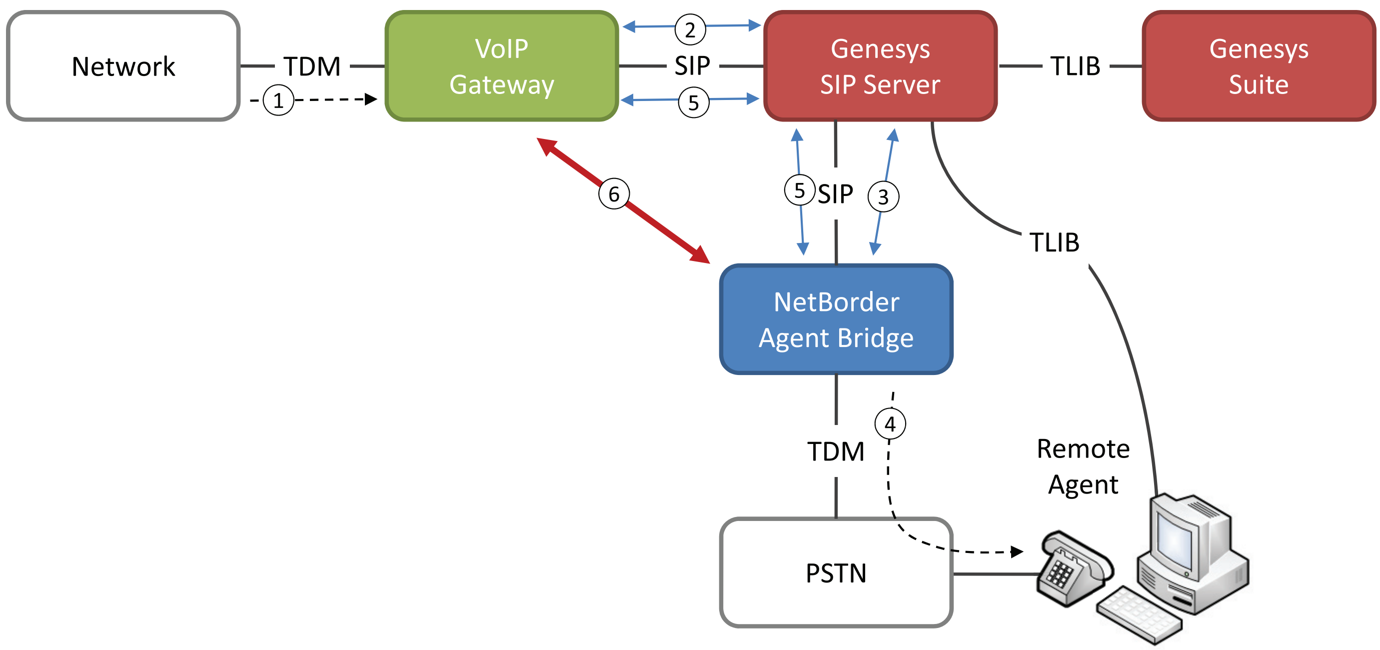

Figure 8 : Inbound Call Routing to Remote Agent

Call Flow Description:

1. Inbound call is initiated
2. On-premise VoIP GW SIP INVITEs Genesys SIP Server
3. SIP Server INVITEs NetBorder to connect to designated remote agent
4. NetBorder plays an audio cue to notify agent that a call is incoming
5. NetBorder re-INVITEs VoIP GW (via SIP Server) with new media endpoint
6. Audio is established between the VoIP GW and NetBorder

Notes:

- At this point, SIP Server remains in the SIP signaling loop. Media (voice) flows directly across the LAN/WAN between the PBX and NetBorder TDM Gateway

4.1.2.3 Caller Hangs-up

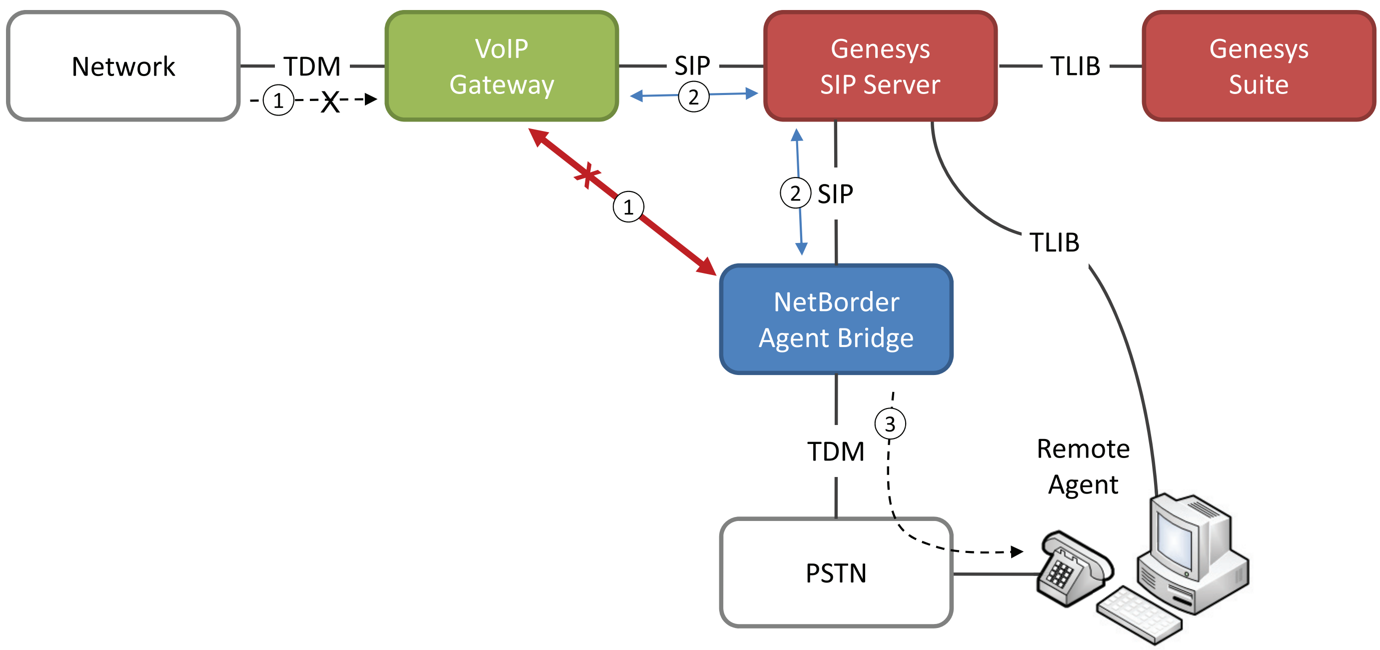

Figure 9 : Caller Hangs-up

Call Flow Description:

1. Caller Hangs-up. Media is terminated
2. NetBorder receives notification that the call is terminating (SIP BYE from VoIP GW via SIP Server)
3. NetBorder plays an audio cue to indicate to the remote agent that it is back in queue to take more calls; Remote Agent is then parked with the NetBorder Agent Bridge, ready to take more calls

4.1.2.4 Remote Agent hangs-up the call via Genesys Agent Desktop

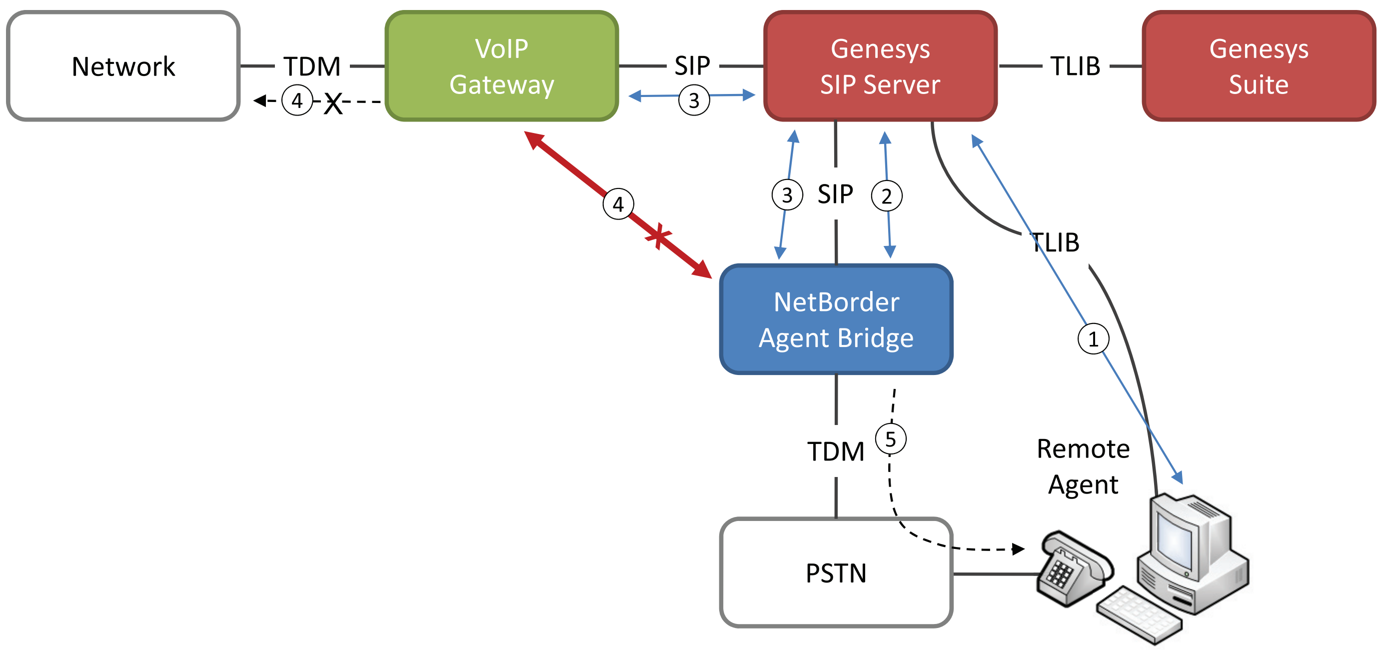

Figure 10 : Remote Agent Hangs-up via Agent Desktop Software

Call Flow Description:

1. Genesys SIP Server receives notification that the agent is hanging-up
2. SIP Server sends SIP BYE to NetBorder
3. NetBorder sends SIP BYE to VoIP GW via SIP Server
4. Media is terminated
5. NetBorder plays audio cue to Remote Agent to indicate it is back in queue; the call is now parked in the NetBorder TDM Gateway and the Remote Agent is ready to take calls

4.1.2.5 Remote Agent hangs-up the phone

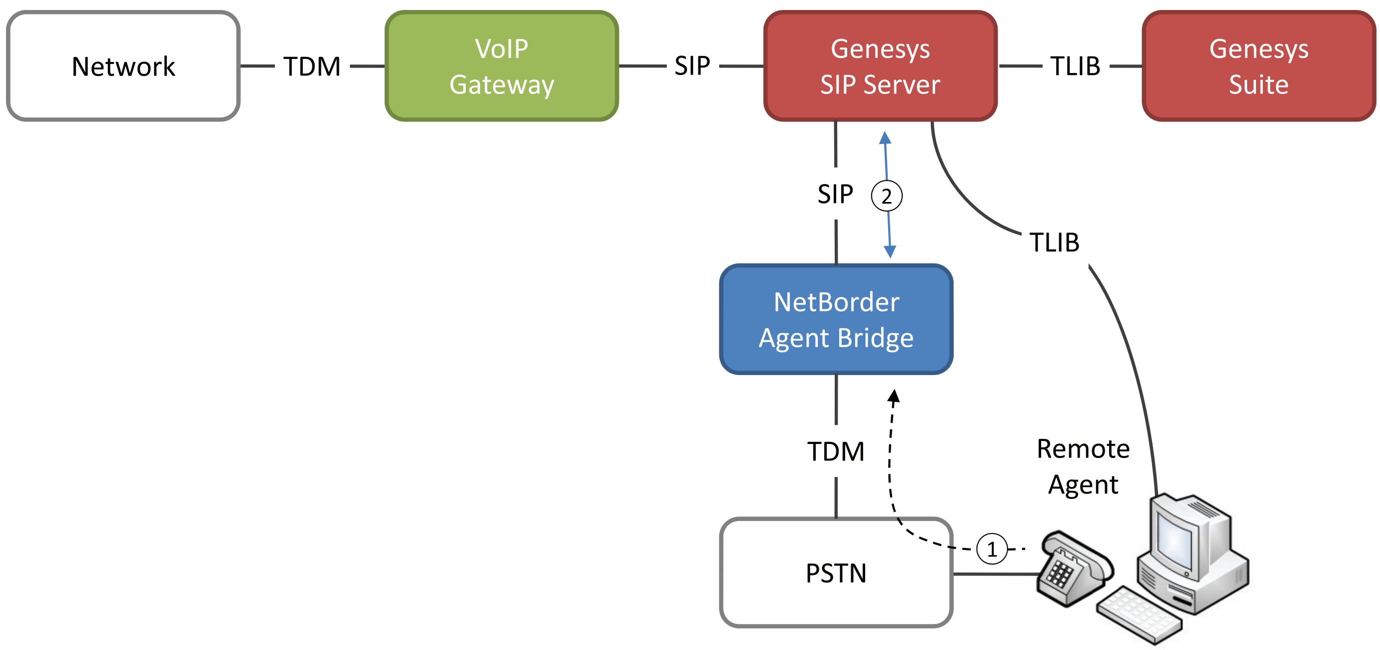

Figure 11 : Remote Agent Hangs-up the phone

Call Flow Description:

1. Agent Hangs-up phone. NetBorder is notified by PRI events from the PSTN
2. NetBorder de-REGISTERs the Remote Agent with SIP Server

4.1.3 Use Case 3: TDM-to-IP Gateway

Sangoma's NetBorder Express gateway solutions, with built-in proxy capabilities such as SIP load balancing and failover routing, provide seamless connectivity to external telephony networks while at the same time perform some of the capabilities found in softswitches. As shown in Figure 12 below, the gateway can be set to load balance traffic among two or more Genesys SIP Servers.

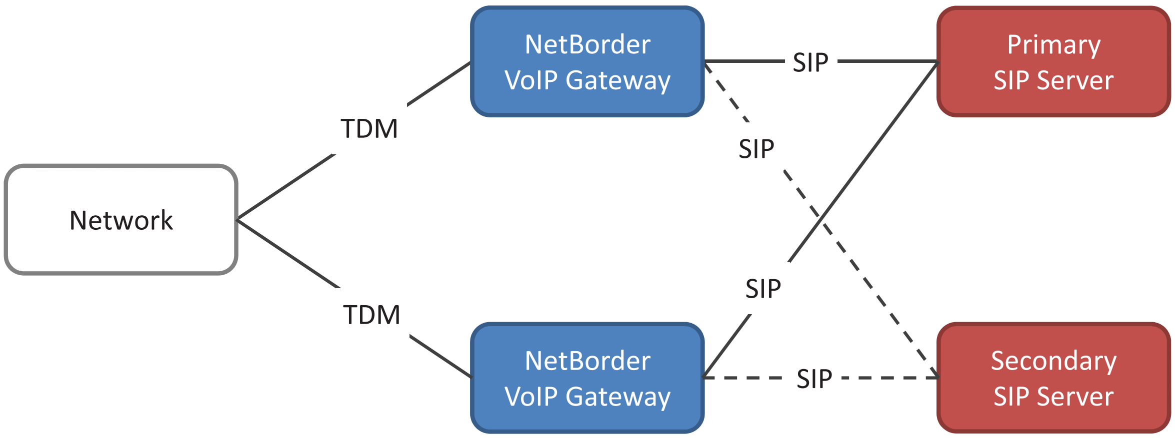

Figure 12 : Failover routing with redundant NetBorder Gateway pair

As for failover, the call flow in Figure 13 below illustrates the situation where one of the SIP Servers would be completely unavailable. In this case, the SIP request would timeout and the gateway would hunt over to a backup SIP destination to establish the session, as per its routing configuration.

5.0 Deployment Considerations

This section addresses some of the more detailed considerations for deployment.

5.1 Platform Requirements And Dependencies

Sangoma's NetBorder software runs on standard computing platforms. The software requires a Windows 2003 Server. The specific processing and memory requirements are dependent on the use case / call flows and the required capacity of the solution. Optional TDM connectivity (connection to a traditional telephony network) is achieved through Sangoma's AFT telephony interface cards, which require a half-length PCI Express or PCI-X slot in the server. The AFT card series can support 1, 2, 4 or 8 T1/E1 interfaces per card, depending on the model. Additional capacity can be delivered by adding cards to the server.

Let's consider an example configuration that supports up to 16 E1s of basic connectivity to Genesys SIP Server. In such a case, the recommended hardware specification would be as follows:

- Intel Xeon Quad-Core (or equivalent), 2 x 4 MB L2 Cache (no AMD processors)
- 2 GB of memory
- 120 GB HDD
- Windows Server 2003 SE
- 2 x Sangoma A108DX (8-span TDM interface card).

Please consult Sangoma sales for assistance in determining the hardware configuration that is right for your particular operating conditions and target density.

The NetBorder platform is compatible with Genesys SIP Server 7.5.000.65 or later.

5.2 Configuration

Here are some of the high-level configuration items for each of the use cases.

5.2.1 Use Case 1: Outbound Using Sangoma's CPA

When using Genesys OCS in predictive or preview dialing mode with SIP Server, the latter will use extensions to the SIP call flow to indicate that CPA is required on the outbound trunk on appropriate call attempts. The outbound route is configured by a DN of type 'Trunk' within SIP Server Switch configuration object. The name choice for that DN is arbitrary.

- contact: address + port 5602 of Netborder installation
- request-uri: must be empty
- dual-dialog-enabled: false
- refer-enabled: false
- call-rfc3725-flow: 1
- auto-redirect-enabled: true

Options needed for this 'Trunk' DN are summarized below (Annex tab of DN configuration):
Other trunk configurations should be left with the default settings.
As no outbound proxy can be set in SIP Server for the outbound SIP trunk, the address and port of the target IP address (usually, the IP address and port of the VoIP gateway reaching out to the PSTN or of the carrier's SIP trunk) must be set in the NetBorder Call Analyzer:

This can be changed in the file:

In **python-scripts\user\InitialRequestHandlerCpaFollowingRequestParam.py** and **python-scripts\user\InitialRequestHandlerBase.py**, set **sip.request.relay.server.address/port** to the address and port of the SIP user agent able to dial out (SIP Server in your case)

"sip.request.relay.server.address":"x.x.x.x"
"sip.request.relay.server.port":<port>

If NetBorder is providing the PSTN interface (not an external 3rd party gateway or SIP trunk from the network), then the relay server address should be set to 'localhost' and the relay server port should be set to 5061.
More information can be found in the User Guide of the NetBorder Call Analyzer.

5.2.2 Use Case 2: Agent Bridge For Remote Agents

The NetBorder Agent Bridge makes remote agents appear to the Genesys framework as if they were IP agents inside the IP Contact Center. When an agent calls into the bridge, he/she is 'registered' to Genesys (directly to the SIP Server or to an external registrar). One must specify in Genesys SIP Server which component acts as the registrar. This is done in the 'Options' tab of the SIP Server configuration.

For an external registrar, one must set the 'external-registrar' parameter to the SIP URI of the registrar (e.g. sip:192.168.8.100:5090; transport=tcp).

If Genesys SIP Server is used as the registrar, then the following parameters must be set:

- *'internal-registrar-domains'* is set to the list of domains from endpoints which shall be handled by the internal registrar
- *'internal-registrar-enabled'* is set to 'true'
- *'internal-registrar-persistent'* is set to 'false'

For the agents managed through the NetBorder Agent Bridge, the extension DNs need to be configured as follows (Annex tab of DN configuration):

- contact: address + port 5602 of Netborder installation
- request-uri: must be empty
- dual-dialog-enabled: false
- refer-enabled: false
- call-rfc3725-flow: 1
- auto-redirect-enabled: true

Other extension configurations should be left with the default settings.

Here are some SIP message examples of the interaction between the Agent Bridge and SIP Server:

Example #1: Agent bridge (172.25.58.10:5065) registering contact information for Agent 7893020 with SIP Server at 172.25.58.6

```
[1]   REGISTER sip:172.25.58.6:5060 SIP/2.0
[2]   Via: SIP/2.0/UDP 172.25.58.10:5065;branch=z9hG4bK-d87543-111d7e1c316fe475-1---d87543-
      ;rport
[3]   Max-Forwards: 70
[4]   Contact: <sip:7893020@172.25.58.10:5065>
[5]   To: <sip:7893020@172.25.58.6:5060>
[6]   From: <sip:7893020@172.25.58.6:5060>;tag=7e41ee44
[7]   Call-ID: NWFjNjA4NWIyNzNhYTFlNWE3YTdhNmY2ZTJjZmU4ZDA
[8]   CSeq: 1 REGISTER
[9]   Expires: 70
[10]  Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, INFO
[11]  Content-Length: 0
```

Example #2: Response from SIP Server (registration is valid for 70 sec.)

```
[1]   SIP/2.0 200 OK
[2]   Via: SIP/2.0/UDP 172.25.58.10:5065;branch=z9hG4bK-d87543-111d7e1c316fe475-1---d87543-
      ;rport;received=172.25.58.10
[3]   To: <sip:7893020@172.25.58.6:5060>;tag=A5A0865D-8CF4-46CF-813D-EF09F7135D7B-23
[4]   From: <sip:7893020@172.25.58.6:5060>;tag=7e41ee44
[5]   Call-ID: NWFjNjA4NWIyNzNhYTFlNWE3YTdhNmY2ZTJjZmU4ZDA
[6]   CSeq: 1 REGISTER
[7]   Expires: 70
[8]   Contact: <sip:7893020@172.25.58.10:5065>;expires=70
[9]   Content-Length: 0
```

Example #3: Agent bridge de-registering the agent (“expires=0”) once it leaves the bridge

```
[1]   REGISTER sip:172.25.58.6:5060 SIP/2.0
[2]   Via: SIP/2.0/UDP 172.25.58.10:5065;branch=z9hG4bK-d87543-7338db3cd115b213-1---d87543-
      ;rport
[3]   Max-Forwards: 70
[4]   Contact: <sip:7893020@172.25.58.10:5065>;expires=0
[5]   To: <sip:7893020@172.25.58.6:5060>
[6]   From: <sip:7893020@172.25.58.6:5060>;tag=7e41ee44
[7]   Call-ID: NWFjNjA4NWIyNzNhYTFlNWE3YTdhNmY2ZTJjZmU4ZDA
[8]   CSeq: 2 REGISTER
[9]   Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, INFO
[10]  Content-Length: 0
```

More details on routing can be found in the NetBorder Agent Bridge User Guide.

5.2.3 Use Case 3: TDM-to-IP Gateway

In the case the Sangoma platform is used as a TDM-to-IP gateway, the following guidelines would apply to implement inbound failover routing over two Genesys SIP Servers.

When multiple routing rules are triggered in Sangoma's solution, the Gateway will apply the rule with the highest priority ('q-value'). Multiple SIP destinations with different priorities could thus be targeted for a single PSTN call. This way, if the Gateway is not able to connect with the highest priority destination, it will try the second highest priority, and so on, according to the maximum number of rules that can be used to establish a call (as per the ***netborder.gw.maxRoutingRulesMatches*** global configuration property in [NETBORDER_HOME]/config/gw.properties). When working with a primary and a secondary SIP Server, this property should be set to '2'.

Below are sample routing rules that could be used to provide application failover. In this example, the address ***sip:5050@genesys-primary.com*** is targeted in priority (qvalue is set to 0.2). The value 5050 would be the target DN on the inbound routing. If this application does not respond, then the second rule (qvalue set to 0.1) will be executed and attempt to contact ***'sip:5050@genesys-backup.com'***.

```
<rule name=”sip_out_to_primary_server” outbound_interface=”sip” qvalue=”0.2”>
<condition param=”pstn.in.channelName” expr=”.*” />
<!-- Retrieving incoming DNIS and storing in %0 -->
<condition param=”pstn.in.dnis” expr=”(.*)” />
<!-- Retrieving incoming ANI and storing in %1 -->
<condition param=”pstn.in.ani” expr=”(.*)” />
<out_leg name=”” media_type=”sendrecv”>
<param name=”sip.out.requestUri” expr=”sip:5050@genesys-primary:5060” />
<param name=”sip.out.from.uri” expr=”sip:%1@GW_HOST_IP:GW_SIP_PORT” />
<param name=”sip.out.to.uri” expr=”sip:%0@genesys-primary:5060” />
<param name=”sip.out.from.displayName” expr=”Paraxip Gateway” />
<!-- Using a 5 sec ring timeout to detect that the Genesys Primary SIP server is down -->
<param name=”netborder.gw.ringTimeoutMs” expr=”5000” />
</out_leg>
</rule>

<rule name=”sip_out_to_backup_server” outbound_interface=”sip” qvalue=”0.1”>
<condition param=”pstn.in.channelName” expr=”.*” />
<!-- Retrieving incoming DNIS and storing in %0  -->
<condition param=”pstn.in.dnis” expr=”(.*)” />
<!-- Retrieving incoming ANI and storing in %1 -->
<condition param=”pstn.in.ani” expr=”(.*)” />
<out_leg name=”” media_type=”sendrecv”>
<param name=”sip.out.requestUri” expr=”sip:5050@genesys-backup:5060” />
<param name=”sip.out.from.uri” expr=”sip:%1@GW_HOST_IP:GW_SIP_PORT”/>
<param name=”sip.out.to.uri” expr=”sip:%0@genesys-backup:5060” />
<param name=”sip.out.from.displayName” expr=”NetBorder Gateway” />
</out_leg>
</rule>
```

Load balancing across two or more SIP Servers readily be added by having multiple, equal priority routing rules that select a different primary SIP Server depending on some condition. For instance, a simple way to load balance between two SIP targets is to have all calls coming in on even ports go to one target and all calls coming in odd ports go to the other. The conditions to be added to detect even or odd port number in the respective routing rules are:

```
<condition param="pstn.in.channelName" expr=".*[0,2,4,6,8]$" />
```

for channels that finish by a 0,2, 4, 6 or 8.

```
<condition param="pstn.in.channelName" expr=".*[1,3,5,7,9]$" />
```

for channels that finish by a 1, 3, 5, 7 or 9.

More details on routing can be found in the NetBorder Express Gateway User Guide.

6.0 Performance

The performance of Call Progress Analysis for an outbound application is driven by two key metrics: accuracy and response time. Reporting accuracy of a statistical process is always a challenging exercise, as the results will be highly dependent on the task, size of sample, traffic mix, etc. The more challenging the task (for instance operating in wireless conditions with a lot of background noise, or making outbound campaigns to reach enterprises which have a different set of greetings than home users, etc), the best one can appreciate the differences between different technologies. What is important in any performance assessment of statistical results is to highlight the relative performance of different systems, under the same operating conditions.

Figure 14 below represents NetBorder's actual performance on a large set of live calls, for a traffic mix of 50% human responses and 50% of automated system responses. Because the NetBorder Call Analyzer operates with statistical models and makes use of confidence measures to determine when to connect or drop the outbound call, one can easily select the operating point on the accuracy vs. response time tradeoff by selecting the Threshold 'T' of confidence at which a CPA decision would be rendered to OCS for processing. Figure 14 illustrates the NetBorder accuracy vs. response time performance as a function of the Threshold 'T', and compares against the performance of a 'tuned' system with a leading traditional call progress technology.

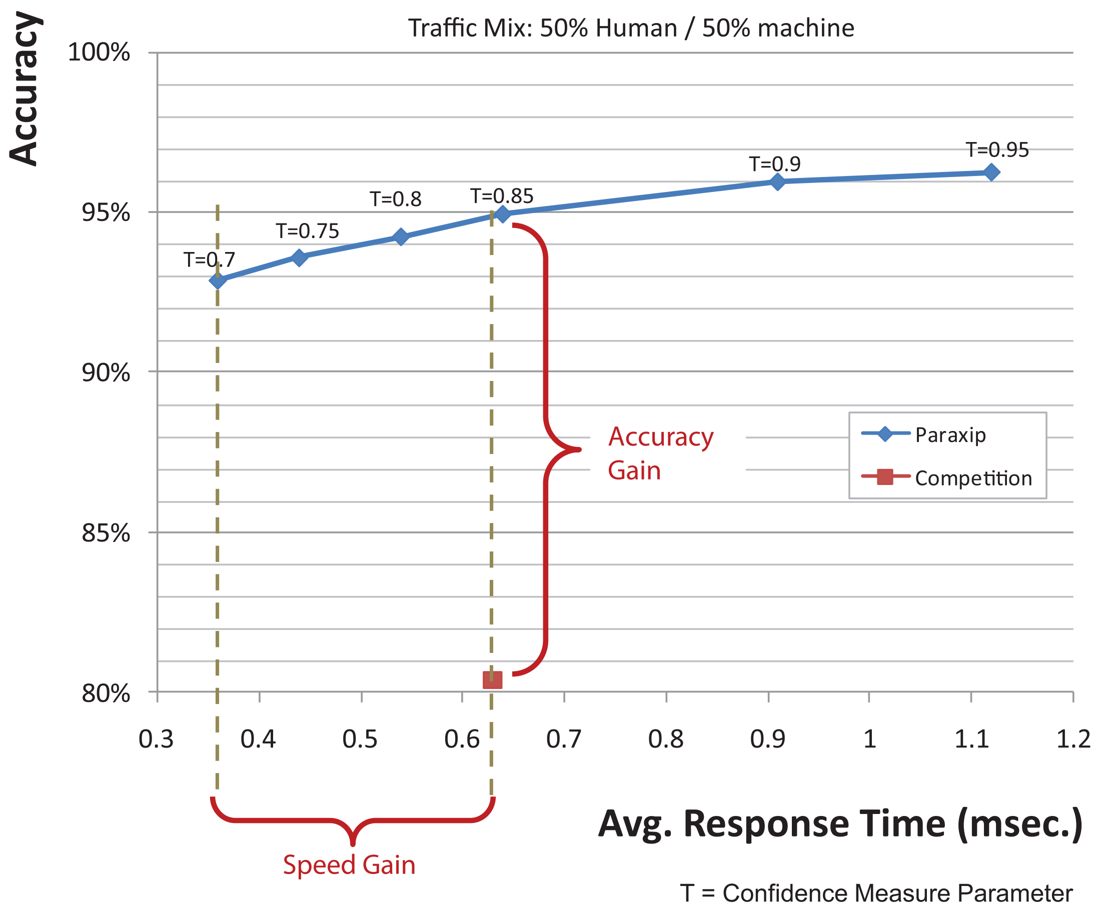

Figure 14 : NetBorder Call Analyzer Performance

One can observe that the NetBorder Call Analyzer can provide significant accuracy gain at T=0.85 (reducing the error rate from about 1 call out of 5 for ‘traditional’ technologies to a mere 1 call out of 20 with NetBorder) for an equivalent average response time. Alternatively, customers can choose to slightly reduce the accuracy gain of NetBorder (compared to traditional technologies) by selecting a lower confidence threshold and thus improving the CPA decision response time by over 40%.

Conclusion

Sangoma adds 'smart connectivity' to IP-based Genesys applications such as inbound, outbound and IVR. As such, it allows Genesys IP Contact Center solutions to connect to legacy PSTN equipment, to be enhanced with superior Call Progress Detection for automating outbound or notification campaigns, to seamlessly integrate remote agents into the IP infrastructure, and/or to have Genesys IP deployed in configurations without a costly softswitch. Sangoma's validated integration with Genesys 7.5 is available today. Please contact sales@sangoma.com for further information.

About Sangoma

Sangoma is the premium provider of voice and data connectivity components for software-based communication applications. Sangoma's data cards, voice cards, gateways and connectivity software are used in leading PBX, IVR, Contact Center and data communication applications worldwide. The product line represents a comprehensive toolset for deploying cost-effective, powerful, and flexible software communication applications. Founded in 1984, Sangoma Technologies Corporation is publicly traded on the TSX Venture Exchange (TSX VENTURE:STC). For more information, please visit: www.sangoma.com