Genesys Application Note

AT&T IP Toll Free (IPTF) and IP Transfer Connect (IPXC) for SIP Server and GVP with Sonus SBC GSX/PSX

Document Version 2.2

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1. Introduction

This white paper describes the interoperability of GVP 8.1.4 release to support AT&T IP Toll free (IPTF) and IP Transfer connect (IPXC) service, where calls are routed to GVP via Genesys SIP Server and Sonus NBS.
GVP is playing role of call-handling device (IVR) used by AT&T's subscribers to answer calls. GVP is able to detect/send touch-tones as Dual Tone Multi-Frequency (DTMF) signals and play announcements to callers so calls may be properly handled without human intervention

A SIP call is routed from AT&T's defined Public IP over internet which will terminate on SONUS NBS. Sonus NBS will anchor SIP and RTP between GVP and AT&T network.

1. SIP end point(s) send SIP REGISTER to Sonus NBS external Interface.
2. Sonus NBS forwards these SIP REGISTER requests to SIP Server through its internal interface. The SIP REGISTER request forwarded to SIP Server has the contact as internal interface of Sonus.
3. Sonus internal interface peers with SIP Server.
4. Sonus external interface peers with AT&T IP toll free and IP transfer connect service.
5. At a high level, a PSTN call is placed and converted to an IP toll free call by AT&T media gateway and is routed to SIP Server through Sonus external and internal interface. The DNIS of the dialed call is matched to a Trunk (Which has Resource Manager as contact) on SIP switch.
6. SIP Server sends an INVITE to the Resource Manager. When the call is received by Resource Manager, it fetches the IVR profile associated with the DNIS and chooses the MCP resource to route the call to for self service.
7. A VXML application is then executed which plays self service prompts to the caller, collect user input and eventually transfer the call to an agent or disconnect the caller after playing appropriate prompts related to the self service VXML application.

Please refer to section 3.2 Integration Logic section which describes in detail various IP Toll free and IP transfer connect call flows.

Important Notes:
The test plan concentrates on using GVP is used as a self service platform.
SIP Server behaves more like a contact center application than as an IP PBX in an AT&T-Sonus-Genesys environment, hence only test cases related to contact center application were executed.
Polycom IP sound point 550 phones were used throughout the testing. This test plan was executed using 1pcc call mode.

2. Software and Hardware Versions

The following equipment and software/firmware were used for the sample configuration provided.

2.1. Genesys Components

Note: Only final versions listed if not specifically mentioned.

Component Name	Version
Configuration Manager	8.0.200.05
Interaction Routing Designer	8.0.000.03
Message Server	8.0.200.02
Solution Control Server	8.0.200.01
Solution Control Interface	8.0.200.03
SIP Server	8.0.400.36
Voice Platform MCP	Initial Test Version=8.1.310.08 Final Test Version=8.1.401.66
Voice Platform RM	8.1.310.02
Genesys Administrator	8.0.300.18

2.2. Non Genesys Components

Component Name	Version
Polycom Sound Point IP 550	3.2.3.1734
Sonus GSX 4000	V07.03.05 R000
EMS	V07.03.05R000
PSX	V07.03.05R000

2.3. Known Issues and Limitations

1. Genesys SIP Server does not support UUI as per AT&T specification.
2. G.726 Codec is not supported by Polycom Sound point IP 550 phones.
3. Ringback feature is not supported by GVP.
4. During interoperability between AT&T, Sonus and Genesys Voice Platform, it was observed that initial part of an announcement/queue music played by GVP may not be heard by the caller when GVP responds with a multiple codec response to a multiple codec offer in the initial INVITE.

   GVP by default is configured to honor the codec ordering as present in the offer and creates a response list that consists of lists of offered codecs that it can support in the response.

   From GVP's perspective, RTP/media is considered negotiated with the highest priority codec in the multiple codec 200 OK response returned for the initial INVITE. GVP starts streaming the RTP (announcement/queue music) with highest ordered priority codec at this point.

However from AT&T perspective, the multiple codec 200 OK response is not considered as the final negotiatiated RTP. AT&T locks down the RTP negotiation by sending a Re-INVITE with SDP containing only single codec in the offer. The single codec in the Re-INVITE offer picked by At&T is the highest priority codec returned in the multiple codec 200 OK response returned by GVP to the initial INVITE.

GVP responds to the re-INVITE with a 200 OK response containing the single codec response. Upon receiving the 200 OK response on re-INVITE, AT&T considers the call setup as complete and opens the media channel for the caller with GVP. Because of this difference in interpretation of call setup procedure, caller may experience initial announcement clipping as the end to end media channel is open only after the call setup with re-INVITE is completed.

This issue is not observed, when GVP is configured to respond to a multiple codec offer with a single codec response. In this case, no Re-INVITE is sent by AT&T and call setup is complete and end to end media channels opened after the first stage of INVITE/200 OK/ACK.

2.3.1. Issues identified with Genesys products

Description	**Component/Product Name**	**Version**
None		

2.3.2. Issues identified with third party products

Description	**Component/Product Name**	**Version**
For SIP REFER(Attended) TP Busy scenario, Sonus Sends 500 internal server error for the Re-invite (to place Redirecting Party on hold) sent by AT&T network. This error is non reproducible.	Sonus GSX	V07.03.05 R000

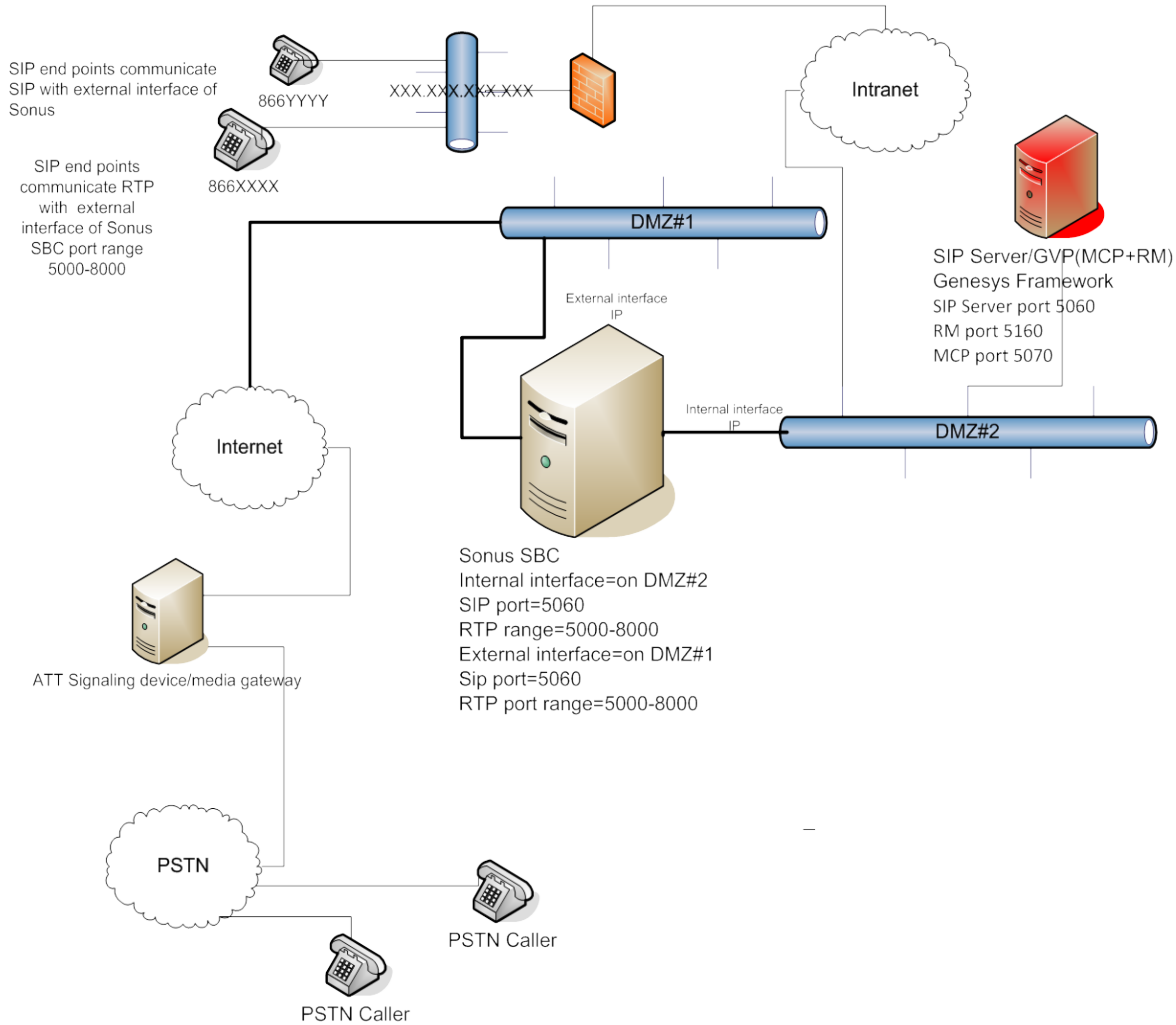
3. Integration and Configuration Section

In this section we will discuss various components involved in interoperability of AT&T certification with GVP & Sonus NBS.

In section 3.1 Integration Points, we will describe at a high level the functionality of each of the components involved in the solution below.

In section 3.2 Integration Logic, we will describe the detailed call flows between the various components.

In section 3.3 we explain the Genesys configuration and in section 3.4, we cover the Sonus configuration.



Note: All the IP addresses have been blanked out intentionally

3.1. Integration Points

SIP Server – SIP Server is a B2B UA. It processes and accepts SIP calls from a valid SIPtrunks/peer like Sonus NBS defined under its SIP Server switch in CME. SIP agents register to Sonus NBS which forwards the sip registration to SIP Server.

Stat Server – tracks the real-time states of interaction management resources, and to calculate basic measurements about the performance of contact center events and activities. Stat Server is a client of T-Servers and receives call events about resources from T-Servers and updates resource status accordingly.

Genesys Administrator (GA) – GA is a multi functionality web interface for maintaining the GVP solution. It connects to Config Server for all the Configuration related details, Solution Control Server (SCS) for all the real time information about the health and state of the processes running and the LogDBServer for centralized logging. It also connects to the Reporting Server (RS) to acquire real time and historical information about the call handling statistics. In short it acts as a one stop solution for Configuration Manager & Solution Control Interface.

Genesys Voice Platorm (GVP)- GVP is a software suite that constitute a robust, carrier-grade voice processing platform. GVP unifies voice and web technologies to provide a complete solution for customer self-service or assisted service. GVP comprises of following components:

- **Media Control Platform(MCP)** – MCP is the VXML Browser & the media server for GVP 8.x and works with SIP protocol. It also has conferencing and announcement capabilities as a media server and handles VXML applications.

- **Resource Manager(RM)** – Entry point of calls for GVP. It performs the call validation based on the DNIS which it receives & matches against the DNIS range to fetch the correct IVR profile. Based on the configuration of the IVR profile RM routes the call to a VXML

- **Media Server (MS) -** Genesys Media Server 8.1 is a unified media server that handles all media interactions, such as network prompts, IVR interactions, conferencing, call-progress detection, and call recording. It generates and processes media streams in Real-time Transport Protocol (RTP) format and is responsible for interacting with SIP User Agents (UA) and passing the results of those interaction to the SIP Server.
  Media Server is a subset of the Genesys Voice Platform (GVP), containing a minimum set of core components—the Media Control Platform and Resource Manager—that provide media services within a telephony environment.

IP Toll-Free (IPTF) – An AT&T service that is a combination of enterprise voice and data networks that helps users strengthen the efficiency and capabilities of toll-free services used in contact centers, conferencing, voice messaging and other critical toll-free applications.

IP Transfer Connect (IPXC) – An AT&T service that provides pre- and post-answer SIP-based redirection features that give your customer the ability to activate network-based transfers, using out-of-band signaling, to any other AT&T IP Toll-Free or nodal Toll-Free site. It is designed to help customer's lower costs and complete more transactions by efficiently moving toll-free callers to the appropriate agents, departments or locations without asking callers to redial. Calls are transferred using Speed Dial Codes that map to IP or Nodal Routing Numbers (RRNs) representing AT&T IP Toll-Free or nodal Toll-Free terminations.

Note: Please refer to the section 3.4 Sonus Configuration Section, for description and configuration of various Sonus components.

3.2. Integration Logic

This section covers the important call flow logic for various IP toll free and IP transfer connect scenarios. The end point registration is common to all call flows and is explained in 2 steps below.

1. All end points will register to Genesys SIP Server through Sonus NBS. The SIP end points will configure SIP REGISTRAR as Sonus external IP interface. Once Sonus receives the SIP REGISTER messages, they will change the contact or Address of record (AOR) from original IP address of phones to Sonus internal interface and send the REGISTER to Genesys SIP Server (IP PBX).

2. Therefore for each extensions registered to SIP Server, the contact associated with that extension is the Sonus internal interface.

Note: All the interactions in the call flow from AT&T to GVP via SIP Server and vice versa will be taking place through Sonus NBS.

Call from AT&T is sent to GSX first which then performs a DIAMETER policy dip with PSX and routes the call to SIP Server through the internal interface. Likewise when a new transfer call leg originates from SIP Server towards Sonus internal interface, Sonus GSX would perform a DIAMETER policy dip with PSX and then routes the call towards AT&T.

This policy dip between GSX and PSX is assumed in all the call flows and is not explicitly shown in the call flow steps or the ladder diagram.

3.2.1. Basic IP Toll Free inbound call Transfer using REFER from GVP to agent

1. Caller dials the toll free number as advertised by the vendor and provided for by AT&T. AT&T on receipt of this call over PSTN, routes a SIP call to Sonus NBS external interface with the provisioned DNIS. Note that the DNIS may be pre-pended with some numbering pattern. In our test environment, AT&T provided a DNIS with 5 leading zeroes (i.e. “00000”).

   Note:- This DNIS value is configured as a TrunkDN (with prefix=”00000”, replace-prefix=”” value and contact as Resource Manager IP address and port) on SIP Server Switch. For e.g., if the original DNIS=000001002 then the provisioned DNIS for VXML application is 1002 on GVP.

2. When Sonus receives a Request URI with 5 leading zeros “00000”, the dial-plan logic should be built in Sonus to route that call to Genesys SIP Server i.e. send SIP INVITE to SIP Server through Sonus internal interface.

3. When the call arrives on SIP Server, SIP Server matches this DNIS to the prefix value of Trunk DN created on the SIP Server Switch and picks the Trunk DN and looks up the contact associated with that Trunk DN. The contact of the Trunk DN should match to Resource Manager IP address.

4. When the call is received by Resource Manager, it fetches the IVR profile associated with the DNIS and chooses the MCP resource to route the call for self service. A VXML application is then executed which plays self service prompts to the caller, collect user input and eventually transfer the call to an agent or disconnect the caller after playing appropriate prompts related to the self service VXML application.

5. If VXML application has logic to transfer the call to a SIP Server agent, then GVP sends a INVITE to SIP Server and SIP Server will send a new SIP INVITE to SIP extension.SIP Server looks up the contact value associated with the TServer section of this extension. The contact is Sonus internal interface IP address (see step 2 above under Section 3.2 Integration Logic).

6. When Sonus internal interface receives a SIP INVITE to extension, Sonus PSX will look up the AOR for this extension and it will send the INVITE to the SIP extension through the Sonus external interface.

7. Finally the agent will answer the call with 200 OK and response will travel back in the reverse direction.

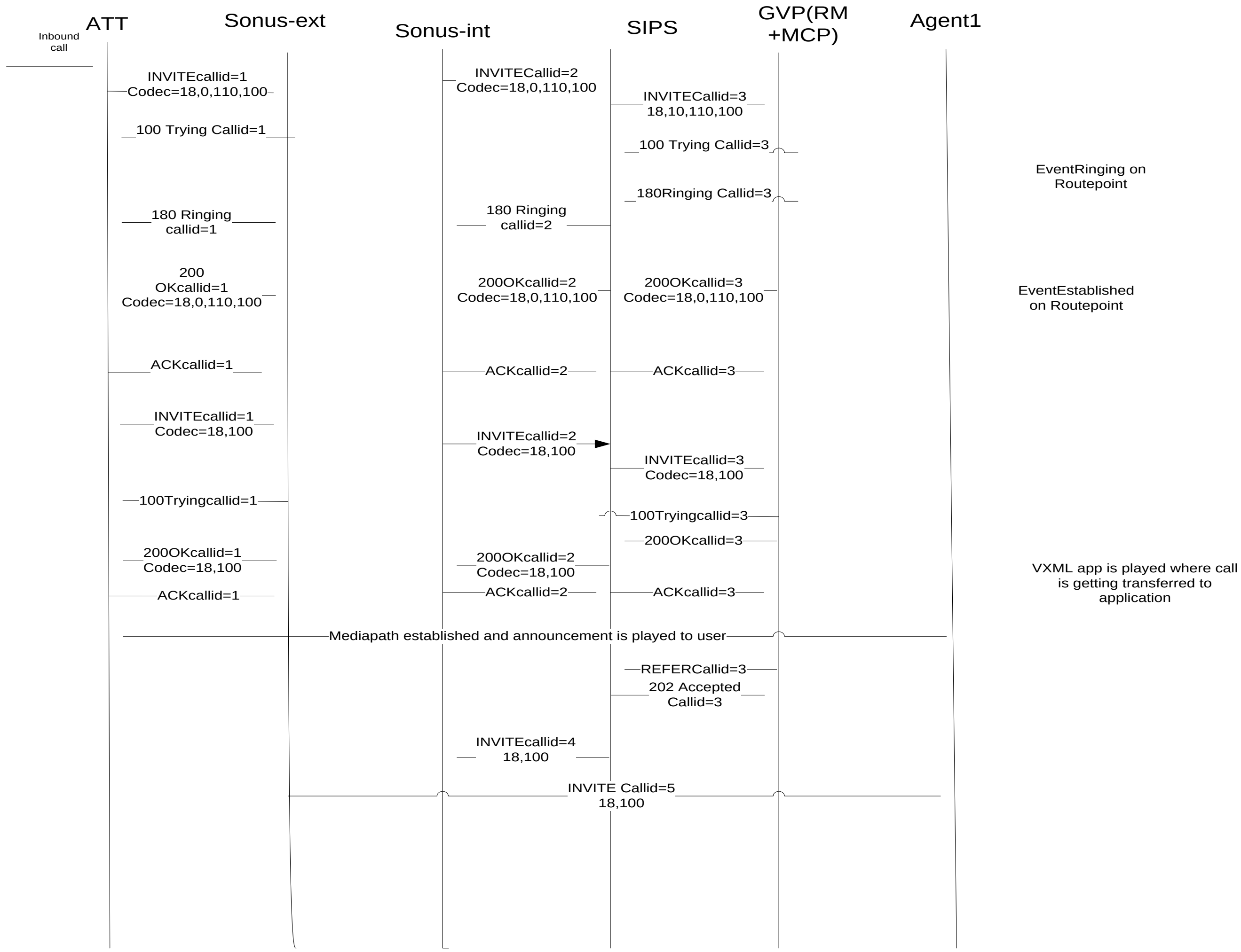
This next sub-section explains codec negotiation as carried out between AT&T IPTF or IP XC services and Sonus/Genesys.

Codec negotiation

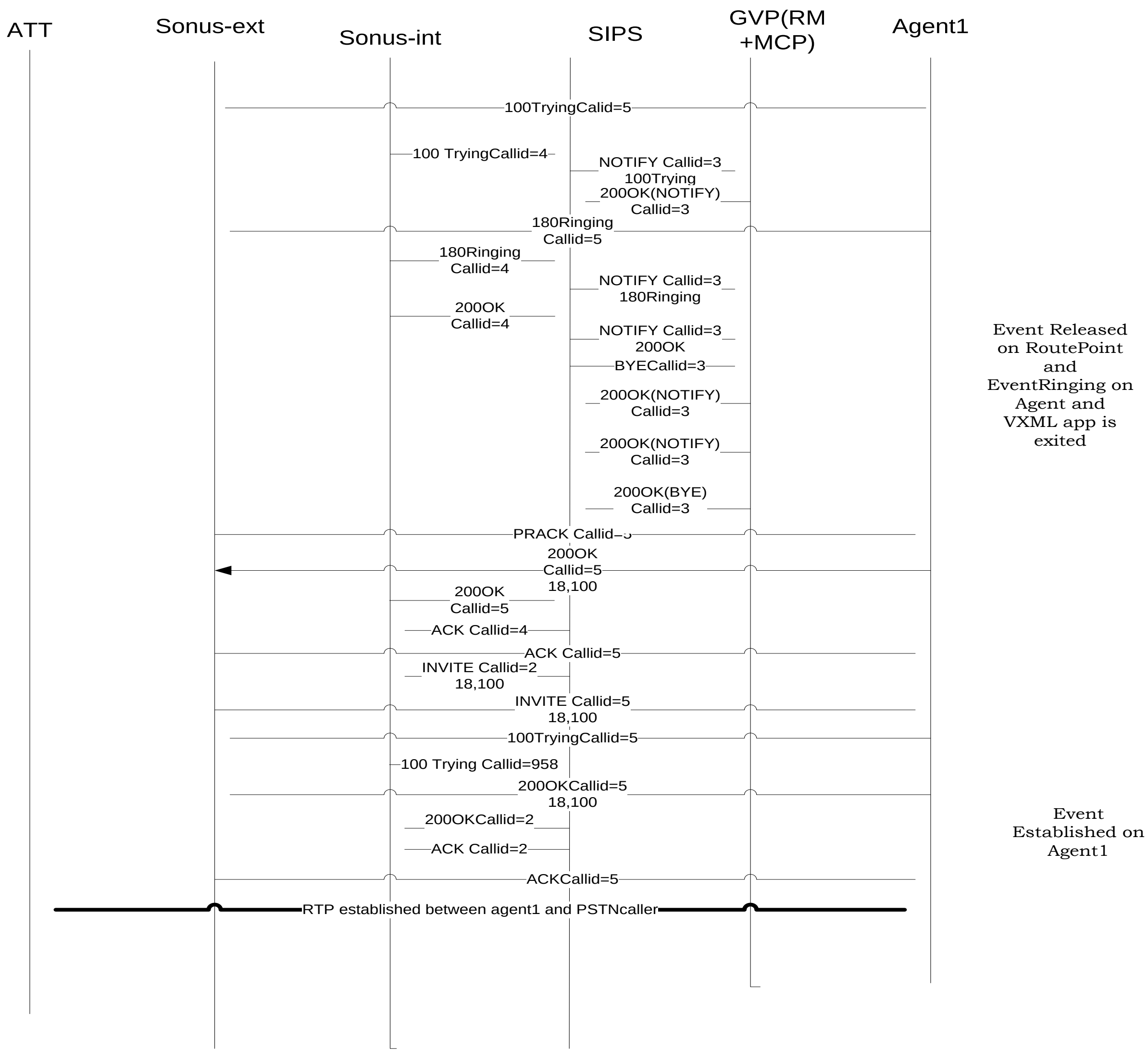
1. Generally, AT&T sends an initial INVITE with an offer that contains multiple codecs (example G729, gG711, G726, dtmf payload).
2. GVP by default is configured to honor the codec ordering as present in the offer and creates a response list that consists of lists of offered codecs that it can support in the response. (example response contains G729, G711, G726, dtmf payload while maintaining the ordering as present in the offer).
3. GVP **always** begins its media (RTP) transmission immediately after it generates a 200 OK on the initial INVITE.
4. If the far end (in this case GVP) responds with multiple codecs, AT&T performs a re-INVITE that contains a new offer containing the highest priority codec (in this case G729) chosen from the response list returned by the GVP.
5. GVP responds with a 200 OK response containing the single codec (example G729) matching the offered codec.
6. Alternately, GVP can be specifically configured to return a single codec response instead of a multiple codec response to the initial INVITE offered from AT&T containing multiple codecs. This is possible by setting the **mpc.answerwithonecodec** option to “1” within MCP component of GVP.This causes GVP to pick the highest priority codec it supports and return that in the response. In case, the first codec in the offer is not supported by GVP, it picks the next supported codec in the list as its response.
7. If GVP returns a single codec in the response to the initial INVITE from AT&T, AT&T will not generate a re-INVITE and will start media (RTP) transmission using the negotiated codec value.

   Refer to the ladder diagram - **GVP responds with a multi codec response to a multi codec offer** below which shows codec negotiation between AT&T and Sonus/Genesys when GVP responds to a multi-codec offer with a multi-codec response.

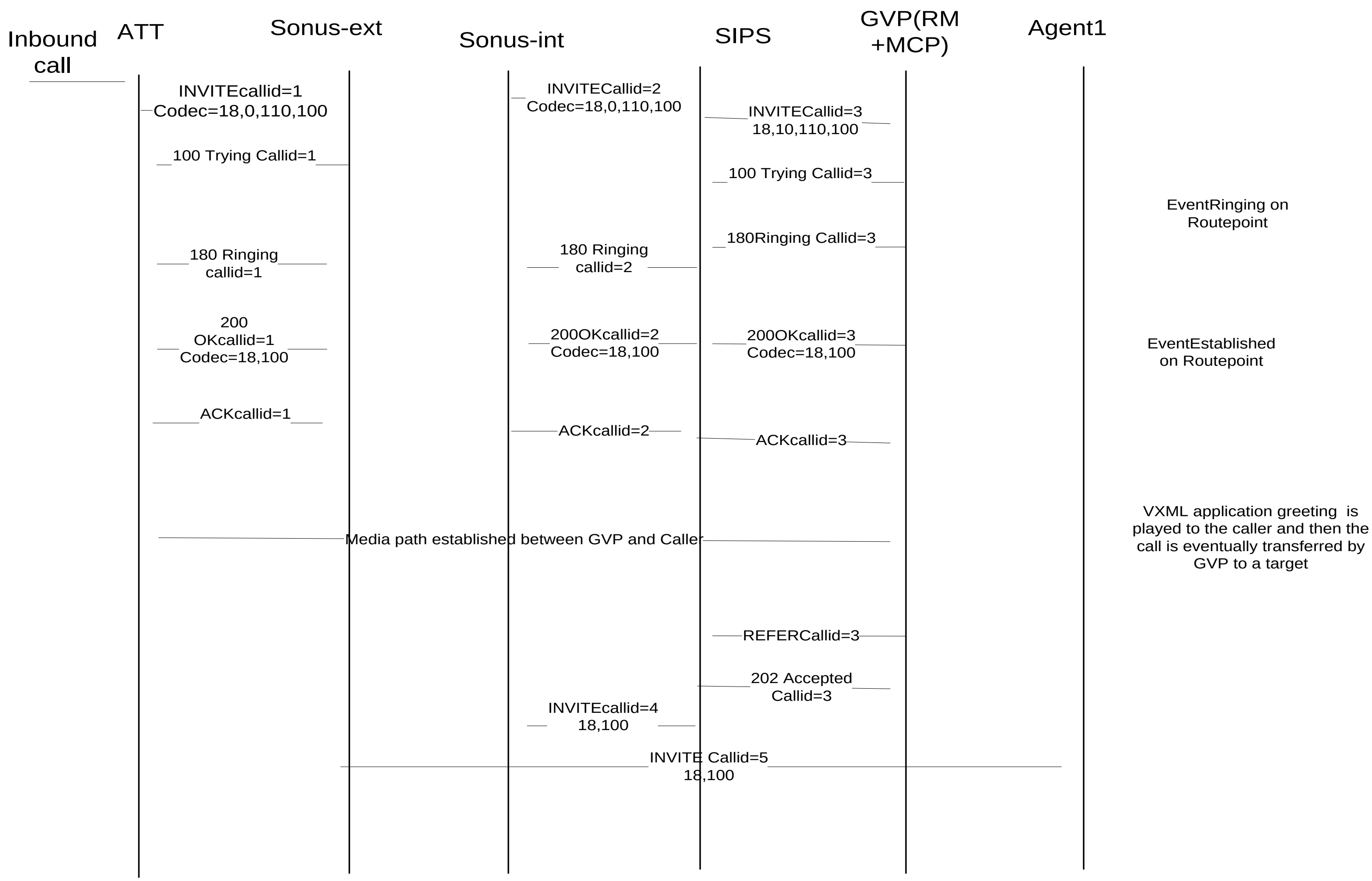
   The subsequent ladder diagram - **GVP responds with a single codec response to a multi codec offer** shows codec negotiation between AT&T and Sonus/Genesys when GVP responds to a multi-codec offer with a single codec response.



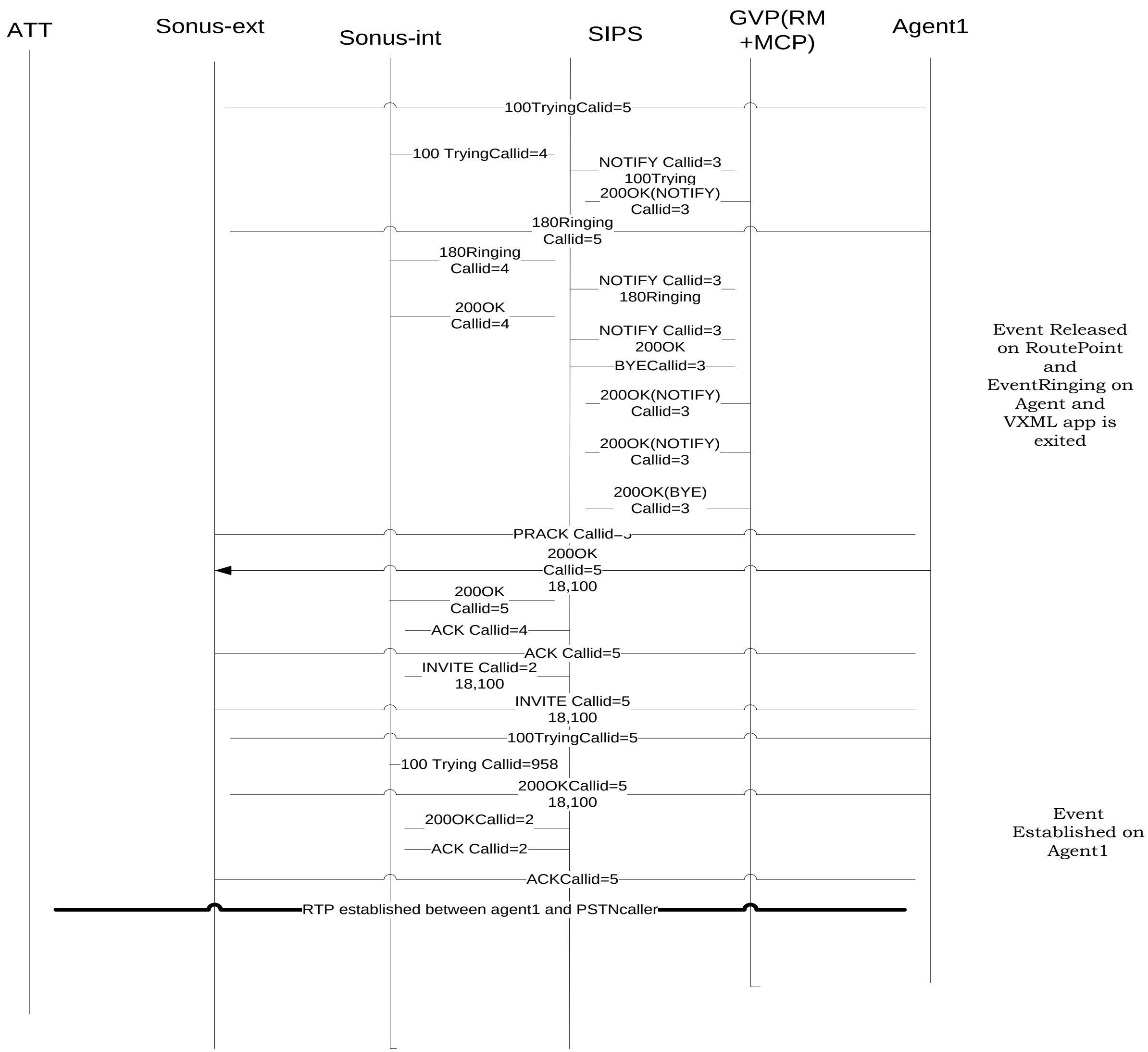
GVP responds with a multi codec response to a multi codec offer –Part 1 / 2



GVP responds with a multi codec response to a multi codec offer –Part 2 / 2



GVP responds with a single codec response to a multi codec offer –Part 1/2



GVP responds with a single codec response to a multi codec offer –Part 2/2

3.2.2 Intra-site unattended transfer call flow for IPTF

1. Inbound call is answered by GVP and transferred to an agent as specified in section 3.2.1 - Basic IP Toll Free inbound call.
   Note: Once the call is transferred to Agent1(as per step1 above) through GVP (using VXML Application), VXML app is exited, and in the rest of the call flow, the communication will be happening between SIP Server, AT&T, Sonus and Media Server(to play Music on Hold).

2. Agent1 (Redirecting Party)/end point1 initiates the 1pcc blind transfer to Agent2 (Target Party)/end point2 on the same SIP Server by pressing the transfer button on the phone (Agent1). This results in phone first placing caller on hold and then sending REFER to Sonus with refer-to=Agent2.

3. Agent1 send re-INVITE to Sonus to place the call on hold with a=sendonly.

4. Upon receipt of re-INVITE from agent, Sonus internal interface sends a re-INVITE to SIP Server w/ a=sendonly attribute.

5. SIP Server sends a re-INVITE w/ c=0.0.0.0 (RFC 2543 based SIP hold) to Sonus external interface.

6. Sonus external interface sends re-INVITE with a=inactive to AT&T Network to place call is on hold.

7. AT&T responds to the re-INVITE by sending a=inactive in 200 OK response to Sonus external interface.

8. Sonus internal interface responds with 200OK c=0.0.0.0 (RFC 2543 based SIP hold) to SIP Server.

9. SIP Server establishes msml based dialog with Genesys Media server to stream music on hold to the caller. Media server responds to the msml INVITE request from SIP Server with a 200 OK.

10. SIP Server then re-INVITEs the caller with the media description of the media server to play music on hold to the caller.

11. SIP Server sends 200 OK c=0.0.0.0 to Agent1 to place it on call Hold.

12. Agent1 sends REFER with Refer-to=Agent2 to Sonus external interface to complete blind transfer.

13. Sonus internal interface forwards the REFER request to SIP Server.

14. SIP Server processes the REFER request and INVITE Agent2.

15. SIP Server sends a BYE to Media Server to end music on hold to be played to the caller. Media server responds to the BYE with a 200 OK response.

16. Agent2 responds back with 100 trying, 180 ringing towards SIP Server through Sonus external and internal interfaces.

17. SIP Server generates NOTIFY (100 trying) message on receipt of 100 trying response and generates NOTIFY (180 ringing) and NOTIFY (200 OK) message on receipt of 180 ringing response to the INVITE. These NOTIFY message provide status of the call transfer to the target party (Agent2) to the referring party (Agent1).

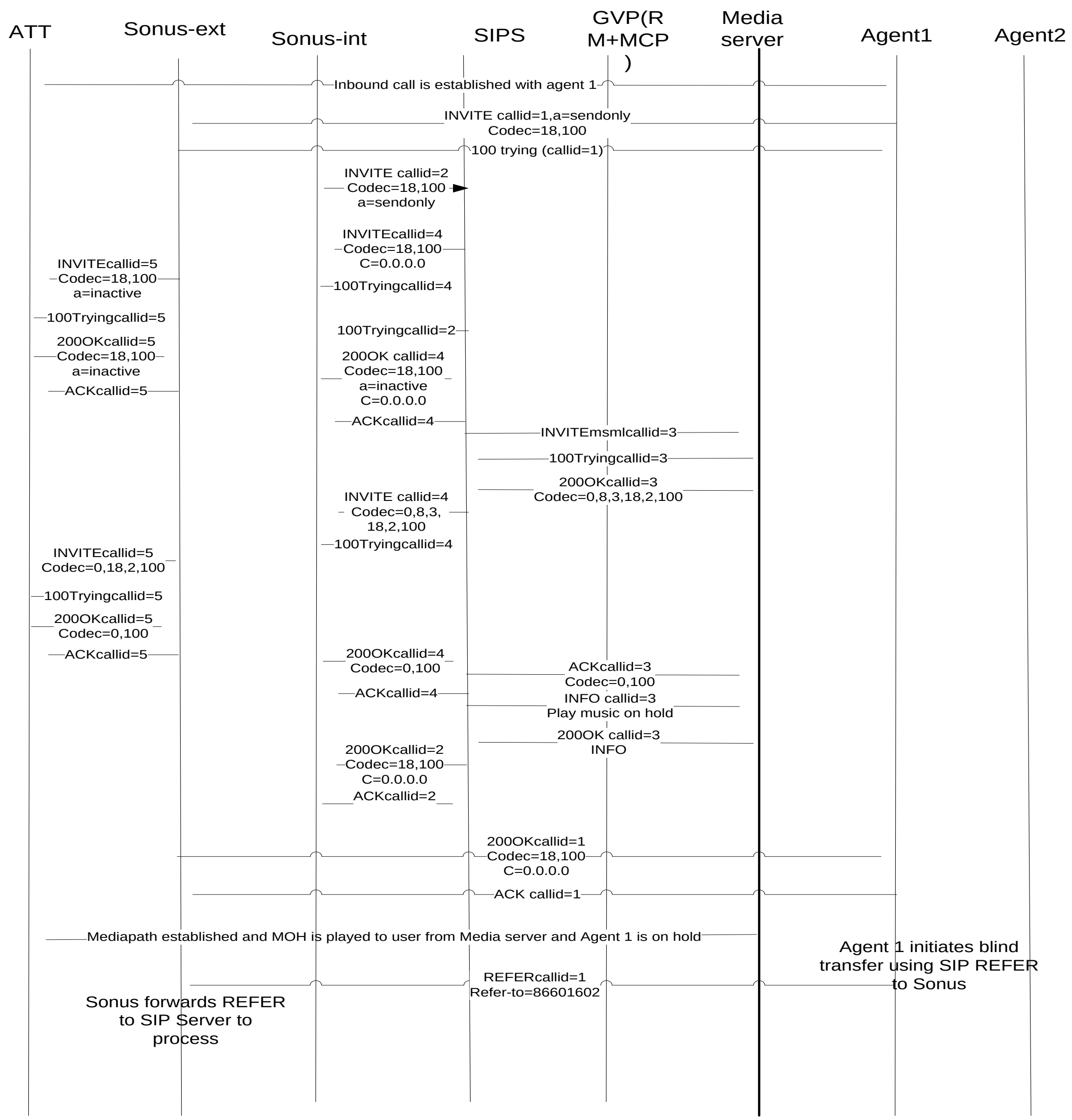
18. Sonus internal interface receives these NOTIFY messages from SIP Server and forwards to the original referring party (Agent1) through the external interface.

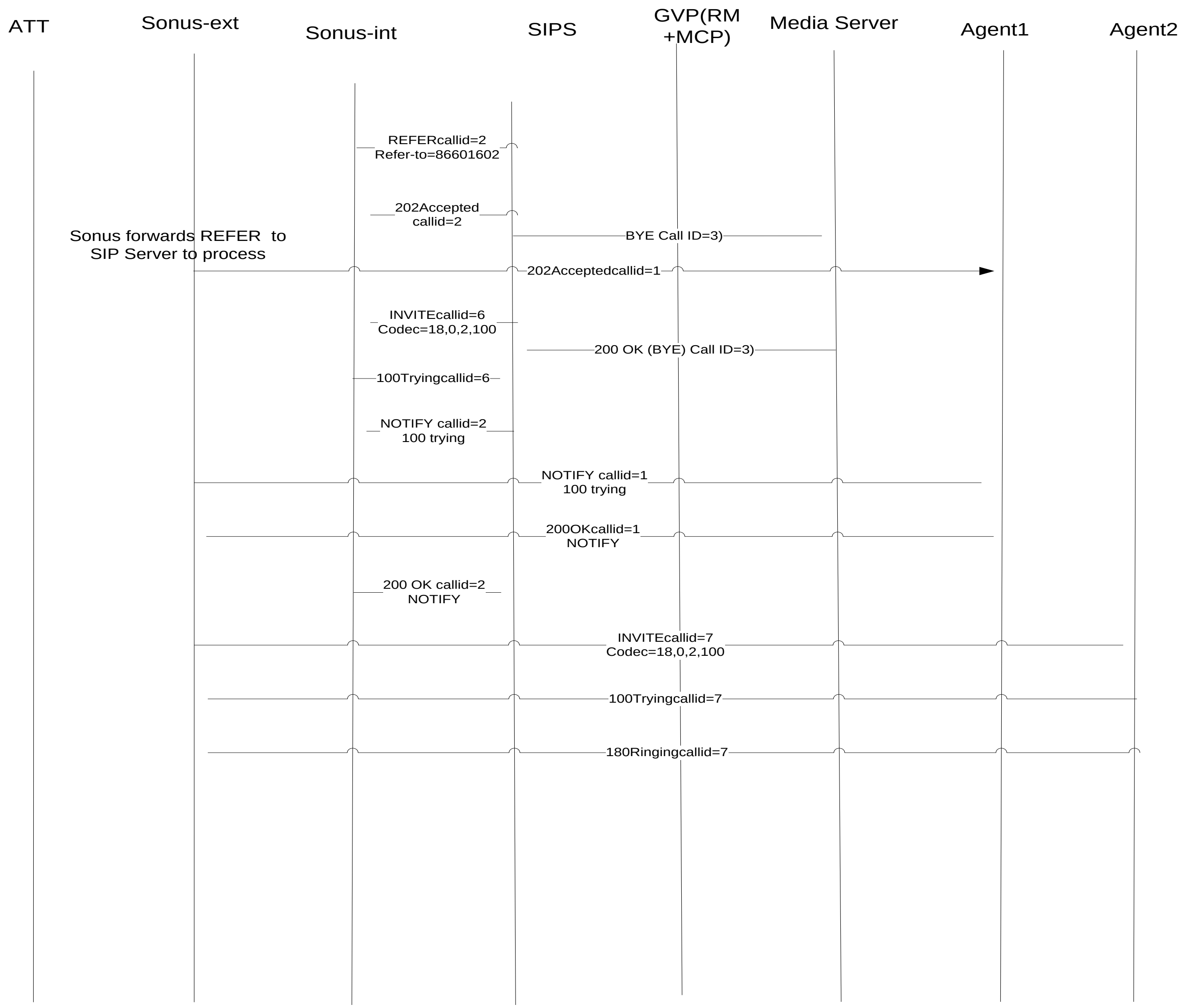
19. Both SIP Server and Agent1 generate BYE to disconnect Agent1 from the call on receipt of NOTIFY (200 OK) responses.

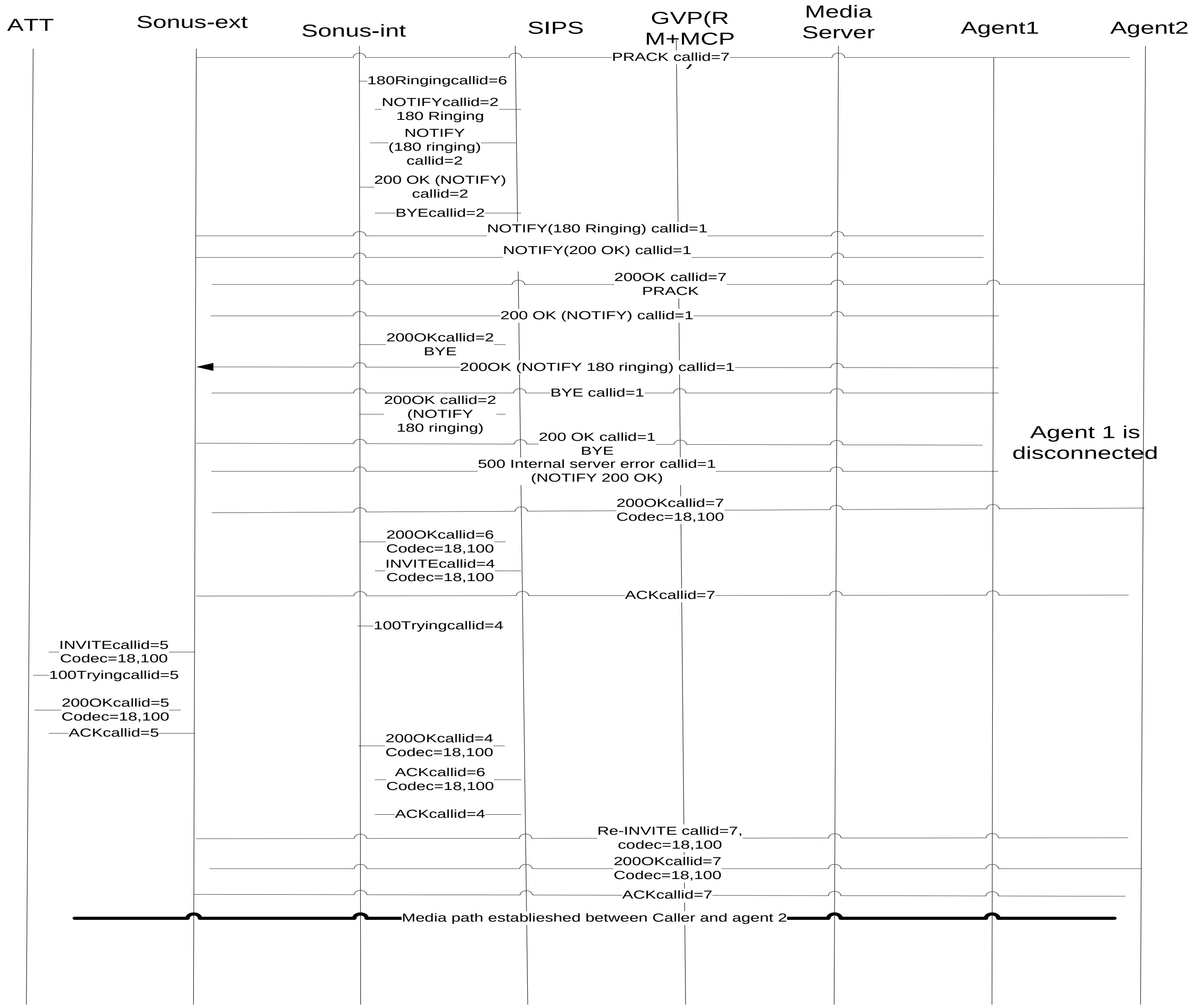
20. Agent2 answers the call with 200 OK response and is received by SIP Server via thr sonus external and internal interface.

21. SIP Server re-INVITE Caller (AT&T) to bridge the media between caller and Agent2.

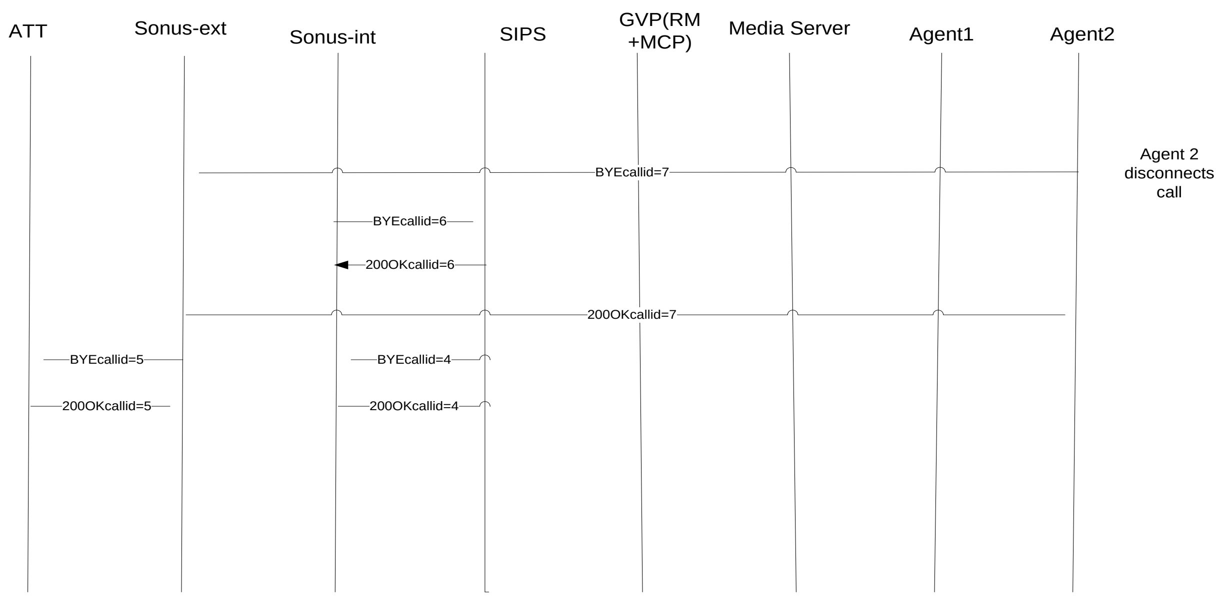
22. Caller and Target Party (Agent2) are in conversation.


ATT
Sonus-ext
Sonus-int
SIPS
GVP(RM+MCP)
Media server
Agent1
Agent2
Inbound call is established with agent 1
INVITE callid=1,a=sendonly Codec=18,100
100 trying (callid=1)
INVITE callid=2 Codec=18,100 a=sendonly
INVITEcallid=4 Codec=18,100 C=0.0.0.0
INVITEcallid=5 Codec=18,100 a=inactive
100Tryingcallid=4
100Tryingcallid=5
100Tryingcallid=2
200OKcallid=5 Codec=18,100 a=inactive
200OK callid=4 Codec=18,100 a=inactive C=0.0.0.0
ACKcallid=5
ACKcallid=4
INVITEmsmlcallid=3
100Tryingcallid=3
200OKcallid=3 Codec=0,8,3,18,2,100
INVITE callid=4 Codec=0,8,3, 18,2,100
100Tryingcallid=4
INVITEcallid=5 Codec=0,18,2,100
100Tryingcallid=5
200OKcallid=5 Codec=0,100
ACKcallid=5
200OKcallid=4 Codec=0,100
ACKcallid=3 Codec=0,100
ACKcallid=4
INFO callid=3 Play music on hold
200OK callid=3 INFO
200OKcallid=2 Codec=18,100 C=0.0.0.0
ACKcallid=2
200OKcallid=1 Codec=18,100 C=0.0.0.0
ACK callid=1
Mediapath established and MOH is played to user from Media server and Agent 1 is on hold
Agent 1 initiates blind transfer using SIP REFER to Sonus
REFERcallid=1 Refer-to=86601602
Sonus forwards REFER to SIP Server to process


ATT
Sonus-ext
Sonus-int
SIPS
GVP(RM +MCP)
Media Server
Agent1
Agent2
REFERcallid=2 Refer-to=86601602
202Accepted callid=2
Sonus forwards REFER to SIP Server to process
BYE Call ID=3)
202Acceptedcallid=1
INVITEcallid=6 Codec=18,0,2,100
200 OK (BYE) Call ID=3)
100Tryingcallid=6
NOTIFY callid=2 100 trying
NOTIFY callid=1 100 trying
200OKcallid=1 NOTIFY
200 OK callid=2 NOTIFY
INVITEcallid=7 Codec=18,0,2,100
100Tryingcallid=7
180Ringingcallid=7



ATT
Sonus-ext
Sonus-int
SIPS
GVP(RM+MCP
Media Server
Agent1
Agent2
PRACK callid=7
180Ringingcallid=6
NOTIFYcallid=2
180 Ringing
NOTIFY
(180 ringing)
callid=2
200 OK (NOTIFY)
callid=2
BYEcallid=2
NOTIFY(180 Ringing) callid=1
NOTIFY(200 OK) callid=1
200OK callid=7
PRACK
200 OK (NOTIFY) callid=1
200OKcallid=2
BYE
200OK (NOTIFY 180 ringing) callid=1
BYE callid=1
200OK callid=2
(NOTIFY
180 ringing)
200 OK callid=1
BYE
500 Internal server error callid=1
(NOTIFY 200 OK)
Agent 1 is disconnected
200OKcallid=7
Codec=18,100
200OKcallid=6
Codec=18,100
INVITEcallid=4
Codec=18,100
ACKcallid=7
100Tryingcallid=4
INVITEcallid=5
Codec=18,100
100Tryingcallid=5
200OKcallid=5
Codec=18,100
ACKcallid=5
200OKcallid=4
Codec=18,100
ACKcallid=6
Codec=18,100
ACKcallid=4
Re-INVITE callid=7,
codec=18,100
200OKcallid=7
Codec=18,100
ACKcallid=7
Media path establieshed between Caller and agent 2

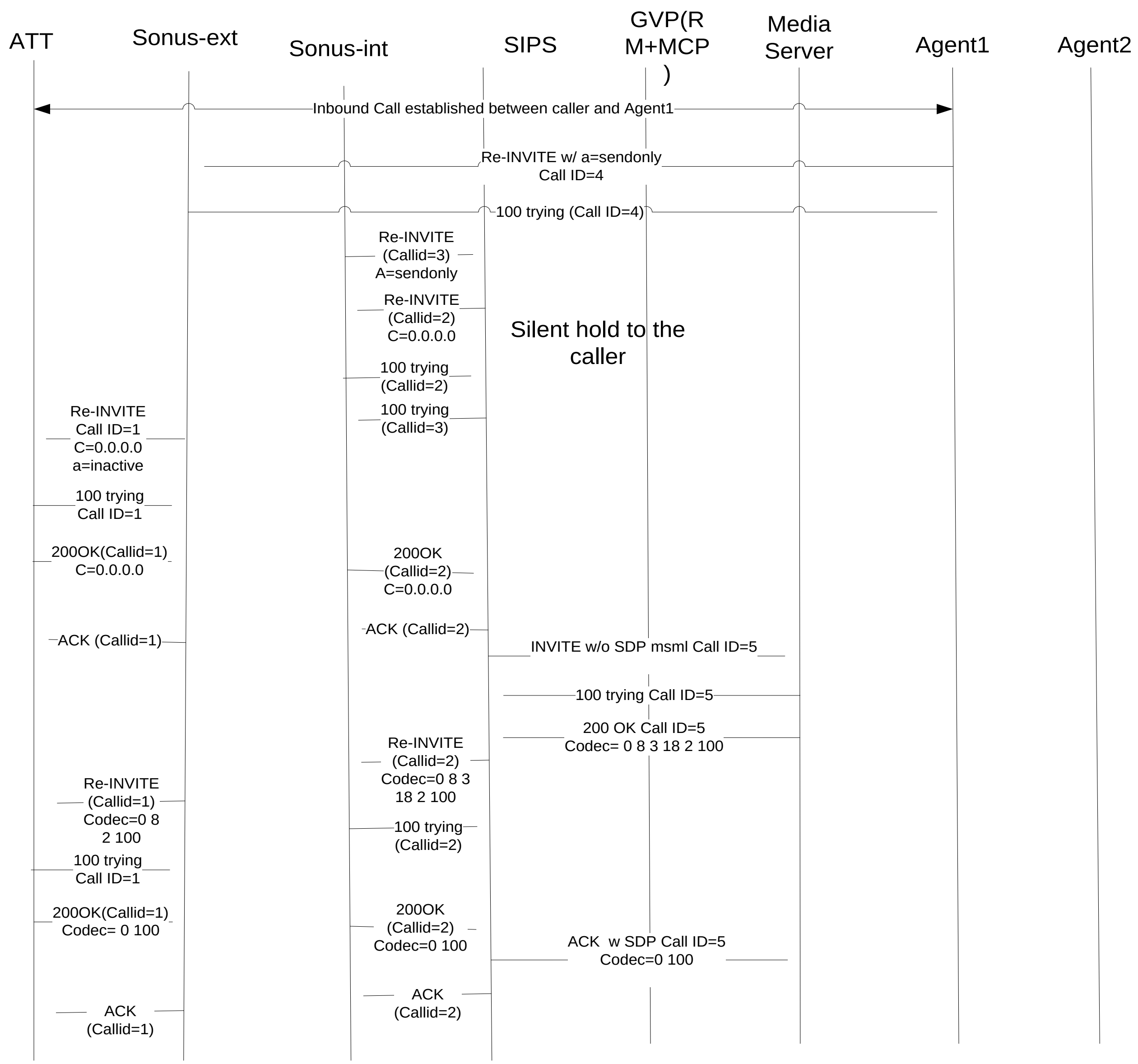

ATT
Sonus-ext
Sonus-int
SIPS
GVP(RM +MCP)
Media Server
Agent1
Agent2
Agent 2 disconnects call
BYEcallid=7
BYEcallid=6
200OKcallid=6
200OKcallid=7
BYEcallid=5
BYEcallid=4
200OKcallid=5
200OKcallid=4

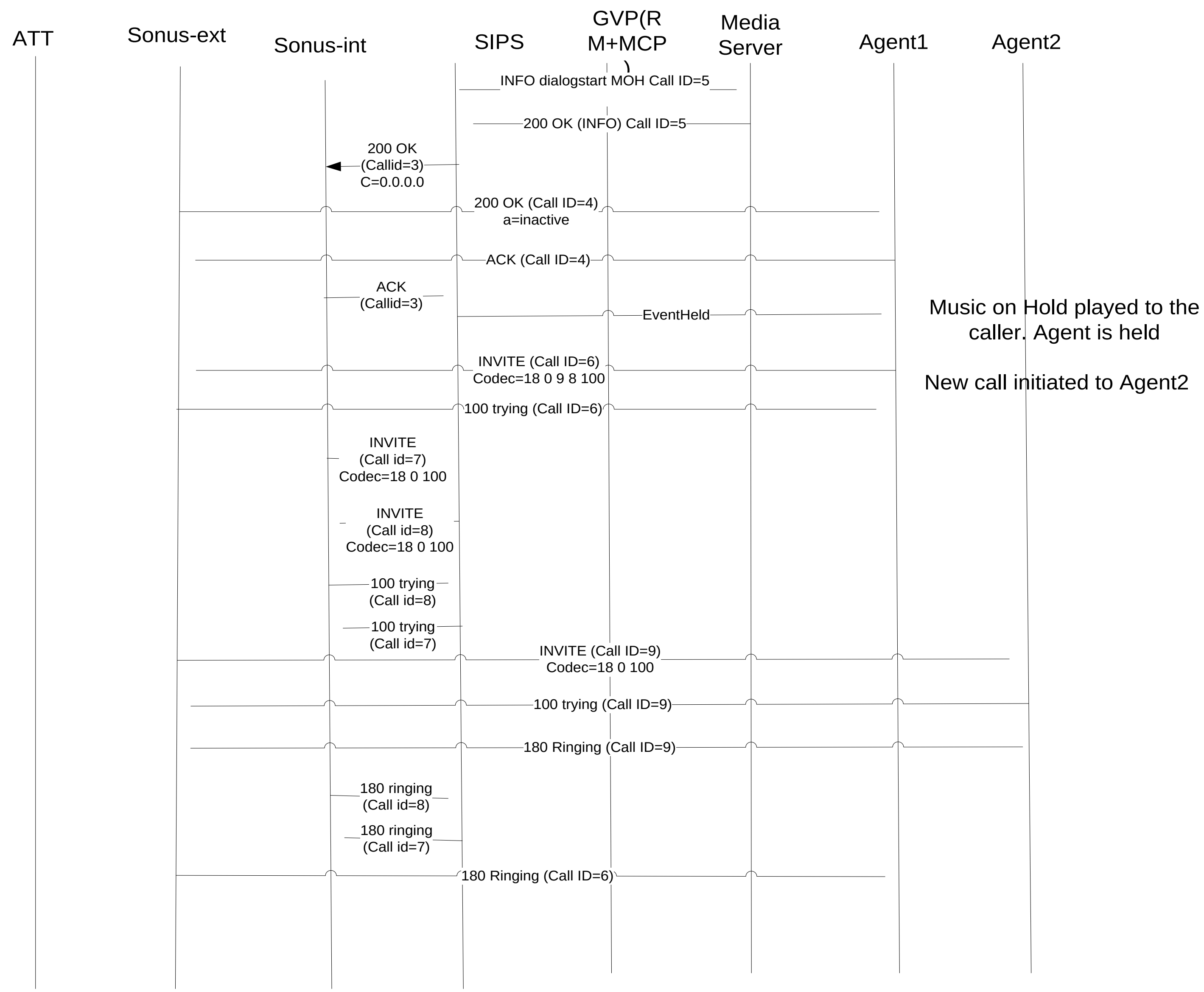
3.2.3 Intra-site attended transfer call flow for IPTF

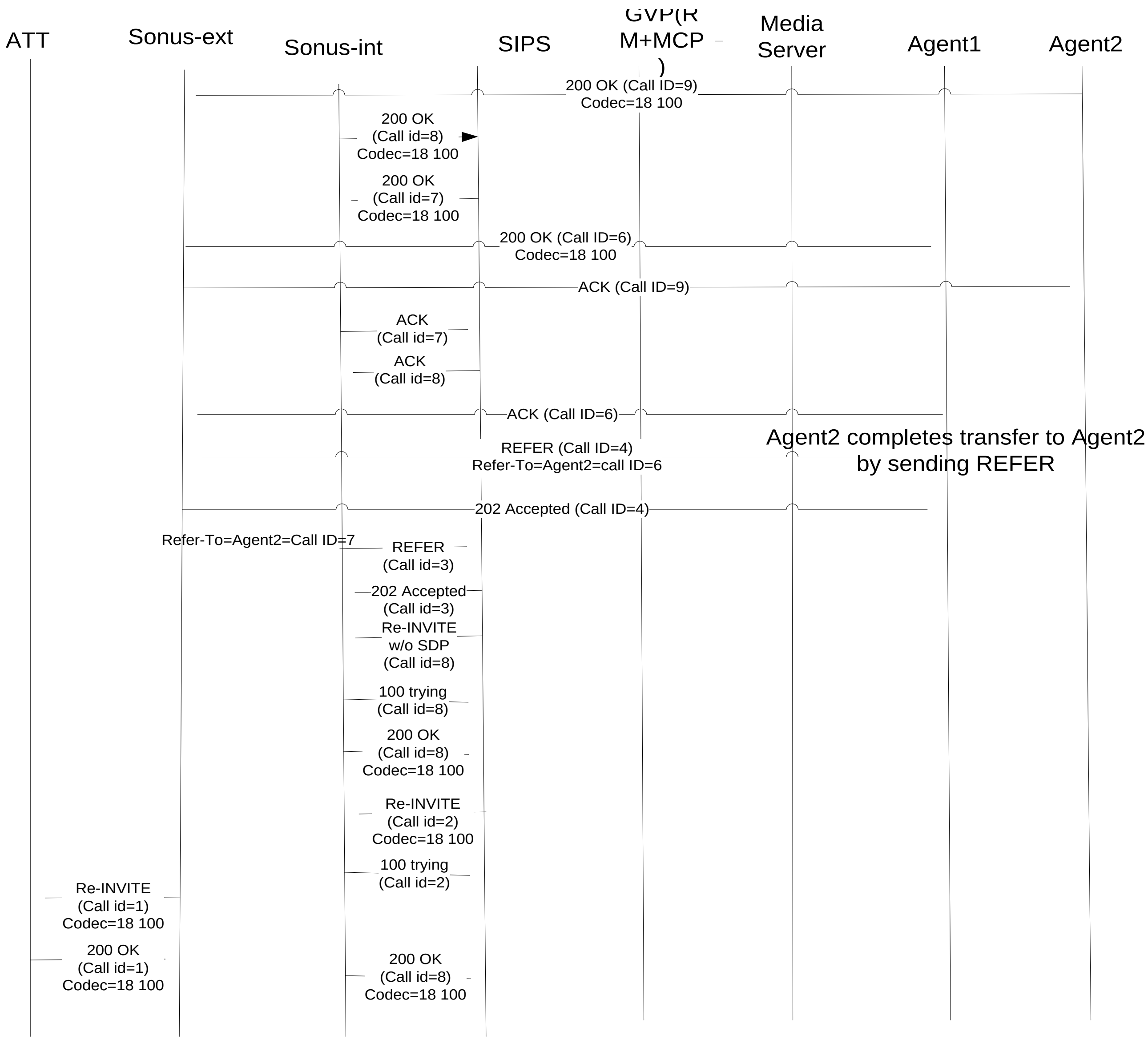
1. Inbound call is answered by GVP and transferred to an agent as specified in section 3.2.1 Basic IP Toll Free inbound call.

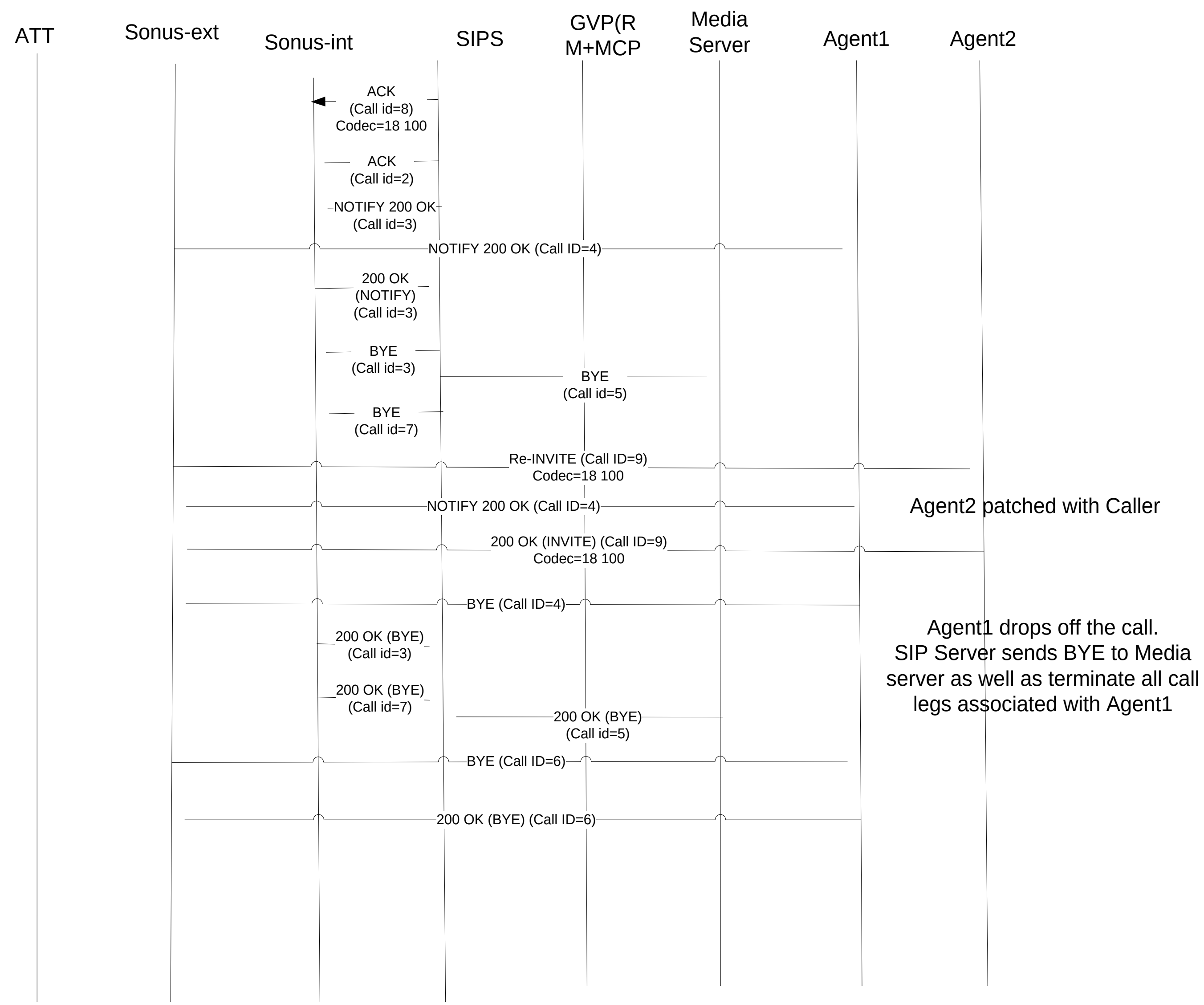
   Note: Once the call is transferred to Agent1 through GVP (using VXML Application), VXML app is exited, and in the rest of the call flow, the communication will be happening between SIP Server, AT&T, Sonus and Media Server(to play Music on Hold).

2. To initiate two step attended transfer, Agent1 first places the caller on silent hold.
3. SIP Server then initiates an MSML dialog with media server and re-invites the caller to play music on hold to the caller.
4. At this stage caller hears music on hold while the Agent1 has placed the call on hold.
5. Agent1 sends a SIP INVITE to Agent2 through Sonus and SIP Server.
6. Agent2 answers the call.
7. Agent1 presses transfer complete on the phone set to complete transfer by sending REFER w/ replaces message to Sonus which in turn forwards the request to SIP Server. The REFER w/ replaces request is generated by the end point to communicate to the caller to replace the callee (Agent1) with the target party (Agent2) in the conversation.
8. SIP Server processes this REFER w/ replaces header by sending re-INVITE to the caller to bridge the call path between caller and Agent2.
9. When the caller (AT&T) answers the call (200 OK response to the re-INVITE request), SIP Server generates NOTIFY (200 OK) response to the originator of REFER w/ replaces (callee/Agent1).
10. Upon receipt of NOTIFY (200 OK) status response, Agent1 drops out of the conversation.
11. Caller and Agent2 are in conversation.


ATT
Sonus-ext
Sonus-int
SIPS
GVP(RM+MCP)
Media Server
Agent1
Agent2
Inbound Call established between caller and Agent1
Re-INVITE w/ a=sendonly Call ID=4
100 trying (Call ID=4)
Re-INVITE (Callid=3) A=sendonly
Re-INVITE (Callid=2) C=0.0.0.0
Silent hold to the caller
100 trying (Callid=2)
100 trying (Callid=3)
Re-INVITE Call ID=1 C=0.0.0.0 a=inactive
100 trying Call ID=1
200OK(Callid=1) C=0.0.0.0
200OK (Callid=2) C=0.0.0.0
ACK (Callid=1)
ACK (Callid=2)
INVITE w/o SDP msml Call ID=5
100 trying Call ID=5
200 OK Call ID=5 Codec= 0 8 3 18 2 100
Re-INVITE (Callid=2) Codec=0 8 3 18 2 100
Re-INVITE (Callid=1) Codec=0 8 2 100
100 trying (Callid=2)
100 trying Call ID=1
200OK(Callid=1) Codec= 0 100
200OK (Callid=2) Codec=0 100
ACK w SDP Call ID=5 Codec=0 100
ACK (Callid=2)
ACK (Callid=1)


ATT
Sonus-ext
Sonus-int
SIPS
GVP(RM+MCP)
Media Server
Agent1
Agent2
INFO dialogstart MOH Call ID=5
200 OK (INFO) Call ID=5
200 OK (Callid=3) C=0.0.0.0
200 OK (Call ID=4) a=inactive
ACK (Call ID=4)
ACK (Callid=3)
EventHeld
INVITE (Call ID=6) Codec=18 0 9 8 100
100 trying (Call ID=6)
INVITE (Call id=7) Codec=18 0 100
INVITE (Call id=8) Codec=18 0 100
100 trying (Call id=8)
100 trying (Call id=7)
INVITE (Call ID=9) Codec=18 0 100
100 trying (Call ID=9)
180 Ringing (Call ID=9)
180 ringing (Call id=8)
180 ringing (Call id=7)
180 Ringing (Call ID=6)
Music on Hold played to the caller. Agent is held
New call initiated to Agent2


ATT
Sonus-ext
Sonus-int
SIPS
GVP(RM+MCP)
Media Server
Agent1
Agent2
200 OK (Call ID=9)
Codec=18 100
200 OK
(Call id=8)
Codec=18 100
200 OK
(Call id=7)
Codec=18 100
200 OK (Call ID=6)
Codec=18 100
ACK (Call ID=9)
ACK
(Call id=7)
ACK
(Call id=8)
ACK (Call ID=6)
REFER (Call ID=4)
Refer-To=Agent2=call ID=6
Agent2 completes transfer to Agent2
by sending REFER
202 Accepted (Call ID=4)
Refer-To=Agent2=Call ID=7
REFER
(Call id=3)
202 Accepted
(Call id=3)
Re-INVITE
w/o SDP
(Call id=8)
100 trying
(Call id=8)
200 OK
(Call id=8)
Codec=18 100
Re-INVITE
(Call id=2)
Codec=18 100
100 trying
(Call id=2)
Re-INVITE
(Call id=1)
Codec=18 100
200 OK
(Call id=1)
Codec=18 100
200 OK
(Call id=8)
Codec=18 100



ATT
Sonus-ext
Sonus-int
SIPS
GVP(RM+MCP
Media Server
Agent1
Agent2
ACK (Call id=8) Codec=18 100
ACK (Call id=2)
NOTIFY 200 OK (Call id=3)
NOTIFY 200 OK (Call ID=4)
200 OK (NOTIFY) (Call id=3)
BYE (Call id=3)
BYE (Call id=5)
BYE (Call id=7)
Re-INVITE (Call ID=9) Codec=18 100
NOTIFY 200 OK (Call ID=4)
Agent2 patched with Caller
200 OK (INVITE) (Call ID=9) Codec=18 100
BYE (Call ID=4)
Agent1 drops off the call.
SIP Server sends BYE to Media server as well as terminate all call legs associated with Agent1
200 OK (BYE) (Call id=3)
200 OK (BYE) (Call id=7)
200 OK (BYE) (Call id=5)
BYE (Call ID=6)
200 OK (BYE) (Call ID=6)

3.2.4 IPXC REFER inbound call flow

1. Point 1 to 3 listed above for a basic IPTF call also applies to IPXC REFER call scenario.

2. When this call is received on SIP Server and SIP Server matches this DNIS to the prefix value of Trunk DN created on the SIP Server Switch and picks the Trunk DN and looks up the contact associated with that Trunk DN. The contact of the Trunk DN should match to Resource Manager IP address.

3. GVP answers the incoming INVITE by playing an announcement via the MCP and then REFERing the caller with an AT&T speed dial code represented in the Refer-to header of the REFER message.

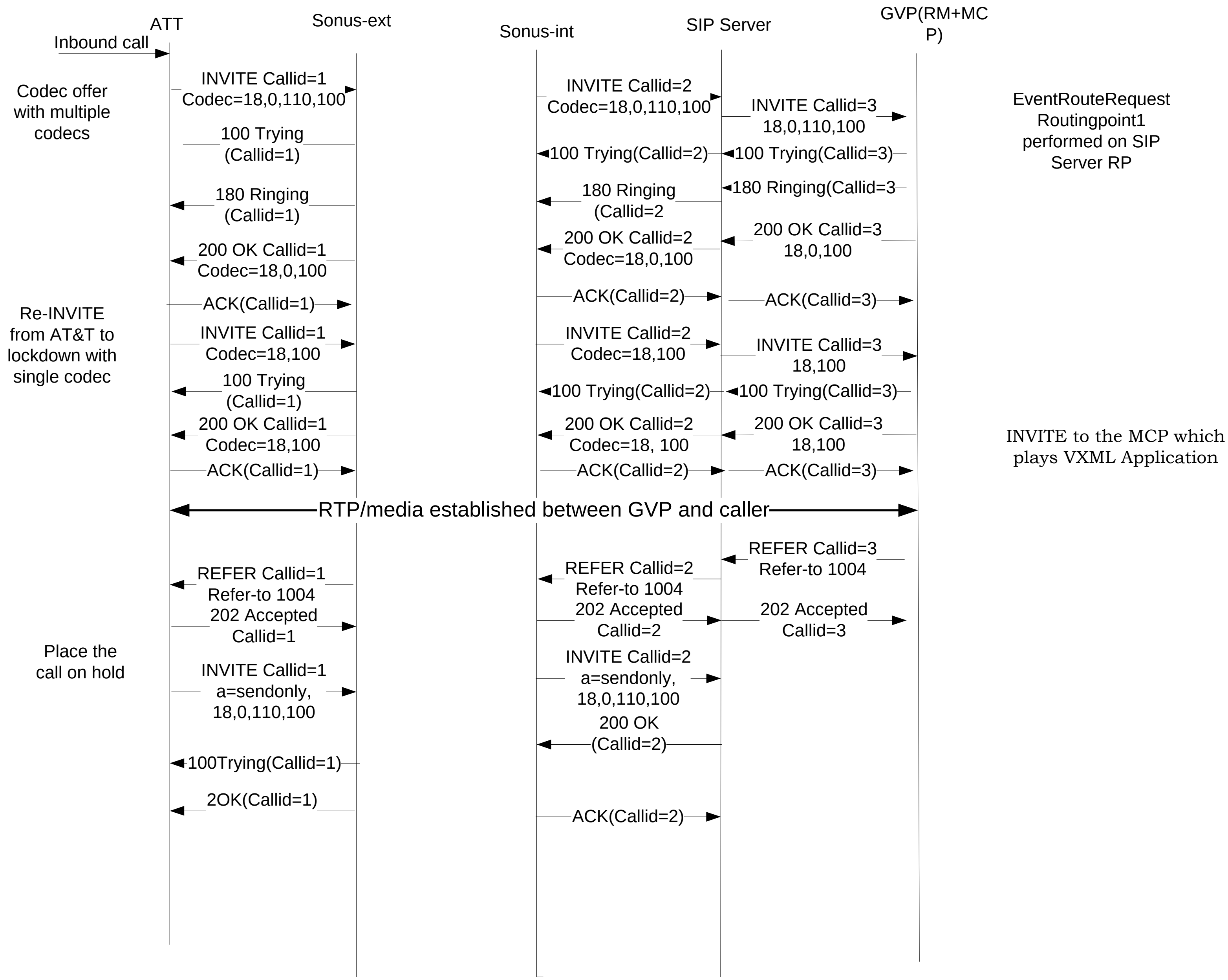
4. When Sonus receives this REFER request from GVP via SIP Server, it will respond with 202 Accepted and forward the REFER request to AT&T.

5. AT&T will process the REFER request containing the AT&T speed dial code within the Refer-to header and based on the provisioning of the routing point on AT&T, AT&T can transfer connect this call to either an IP Toll free or IPXC or another AT&T advanced service end point.

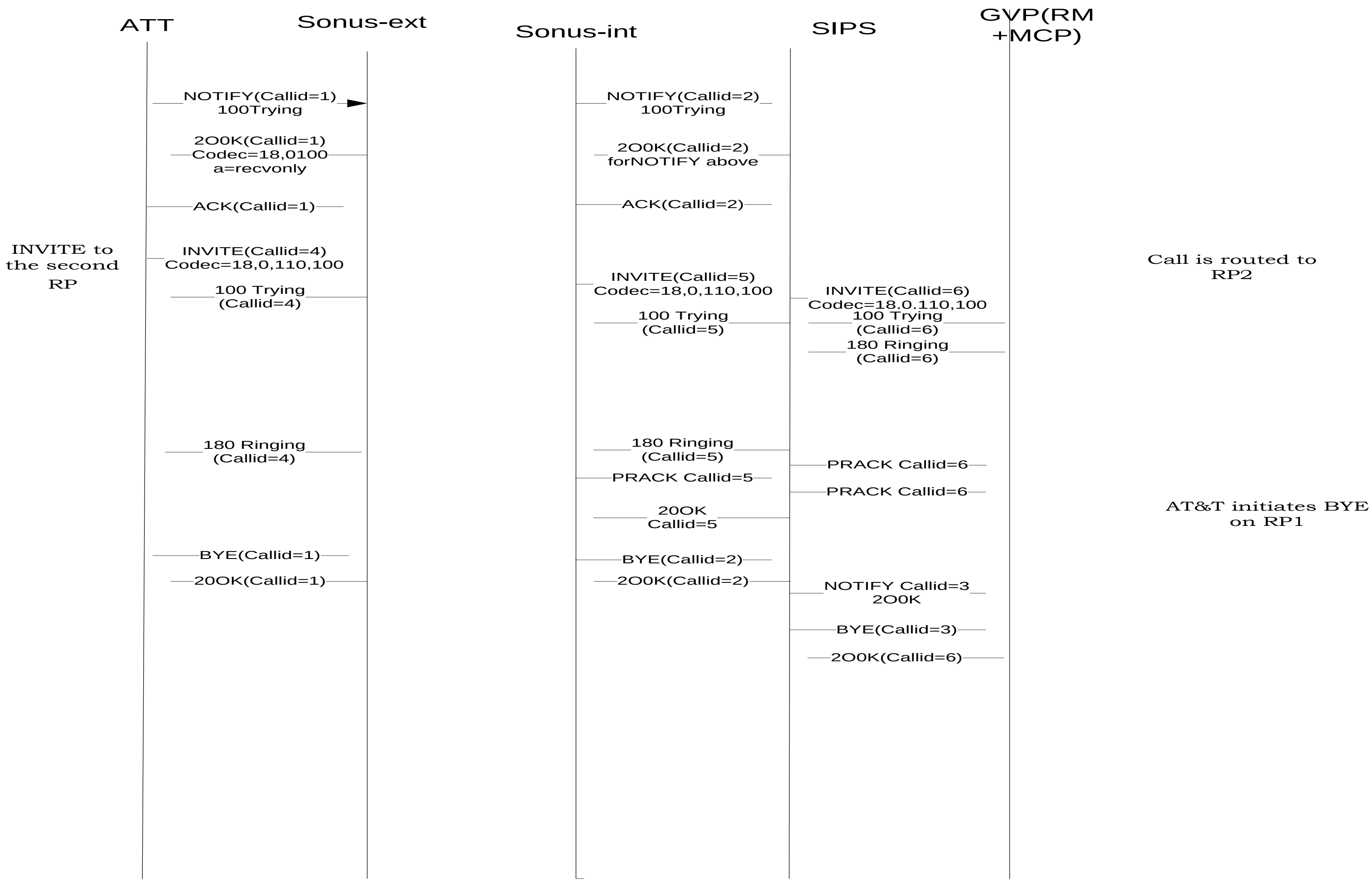
6. AT&T on receipt of REFER can perform either an unattended or an attended transfer as provisioned on AT&T as per customer's request. In an unattended call transfer scenario, when AT&T receives a NOTIFY (100 trying) after sending a new INVITE, it will send a BYE on the original call leg to disconnect the caller with the redirecting party. In an attended call scenario, ATT waits for NOTIFY (200 OK) before sending a BYE on original call leg to disconnect the caller with the redirecting party.

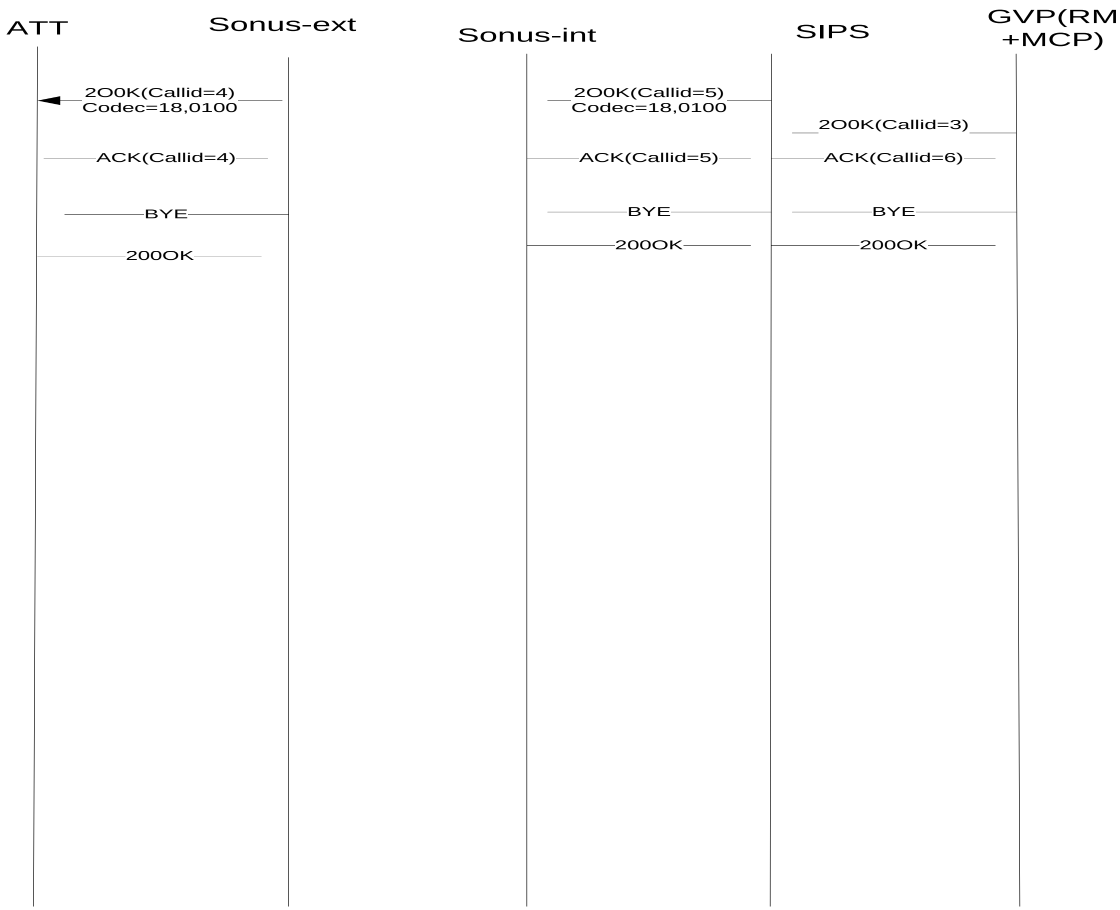
7. In this certification test, AT&T has provisioned to send the new INVITE to Sonus. This INVITE is processed by Sonus and sent to SIP Server and then to GVP for processing. A second VXML application is loaded on the GVP DNIS #2 matching the new SIP URI's user-part. This VXML application will target a SIP agent. Essentially, the new call received on DNIS #2 is similar to the IP toll free call described above as far as SIP messaging is concerned from GVP point of view.

   Note: Please keep in mind, that an IP toll free and IPXC call traverses across different signaling elements on AT&T before it is routed to Sonus NBS and GVP via Genesys SIP Server.



ATT
Sonus-ext
Sonus-int
SIP Server
GVP(RM+MC P)
Inbound call
Codec offer with multiple codecs
INVITE Callid=1 Codec=18,0,110,100
100 Trying (Callid=1)
180 Ringing (Callid=1)
200 OK Callid=1 Codec=18,0,100
ACK(Callid=1)
INVITE Callid=2 Codec=18,0,110,100
100 Trying(Callid=2)
180 Ringing (Callid=2
200 OK Callid=2 Codec=18,0,100
ACK(Callid=2)
INVITE Callid=3 18,0,110,100
100 Trying(Callid=3)
180 Ringing(Callid=3
200 OK Callid=3 18,0,100
ACK(Callid=3)
EventRouteRequest Routingpoint1 performed on SIP Server RP
Re-INVITE from AT&T to lockdown with single codec
INVITE Callid=1 Codec=18,100
100 Trying (Callid=1)
200 OK Callid=1 Codec=18,100
ACK(Callid=1)
INVITE Callid=2 Codec=18,100
100 Trying(Callid=2)
200 OK Callid=2 Codec=18, 100
ACK(Callid=2)
INVITE Callid=3 18,100
100 Trying(Callid=3)
200 OK Callid=3 18,100
ACK(Callid=3)
INVITE to the MCP which plays VXML Application
RTP/media established between GVP and caller
REFER Callid=3 Refer-to 1004
202 Accepted Callid=3
REFER Callid=2 Refer-to 1004
202 Accepted Callid=2
INVITE Callid=2 a=sendonly, 18,0,110,100
200 OK (Callid=2)
ACK(Callid=2)
REFER Callid=1 Refer-to 1004
202 Accepted Callid=1
Place the call on hold
INVITE Callid=1 a=sendonly, 18,0,110,100
100Trying(Callid=1)
2OK(Callid=1)

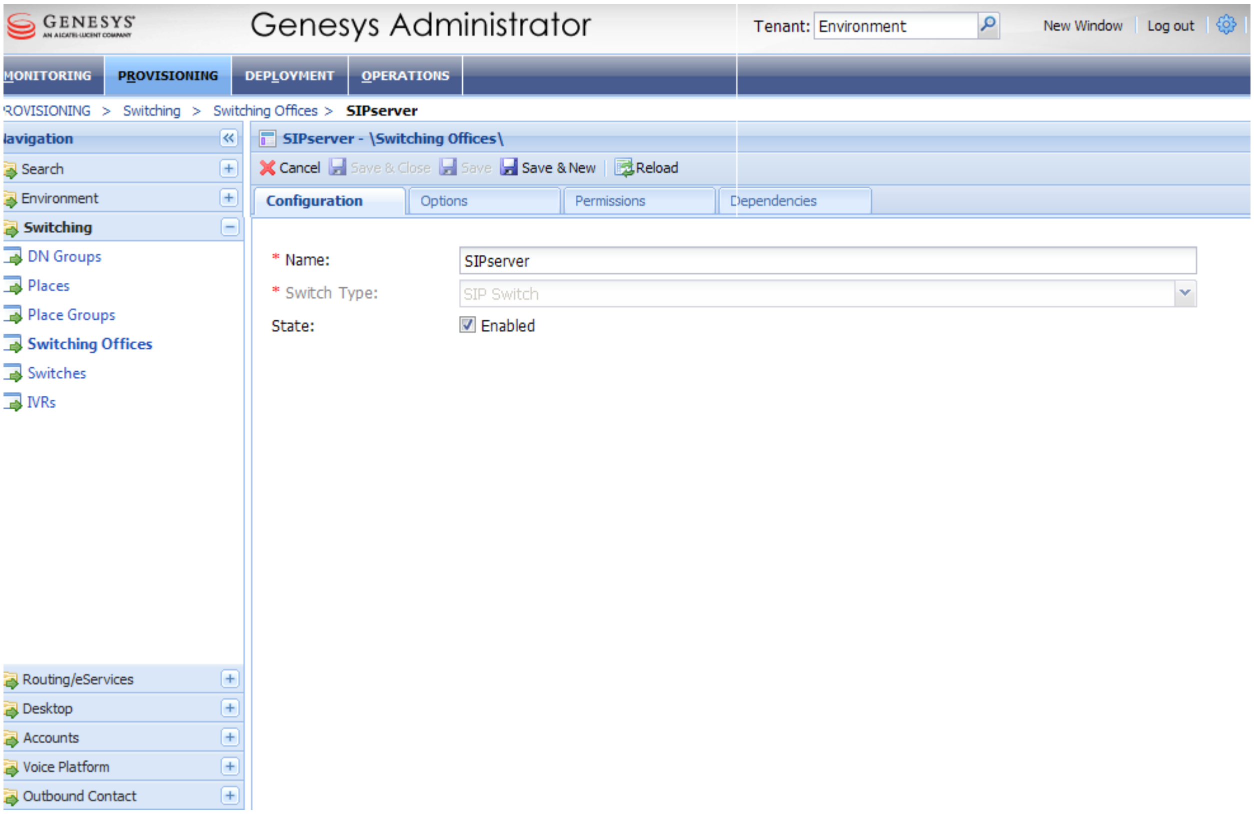

ATT
Sonus-ext
Sonus-int
SIPS
GVP(RM +MCP)
NOTIFY(Callid=1) 100Trying
2OOK(Callid=1) Codec=18,0100 a=recvonly
ACK(Callid=1)
INVITE to the second RP
INVITE(Callid=4) Codec=18,0,110,100
100 Trying (Callid=4)
180 Ringing (Callid=4)
BYE(Callid=1)
20OK(Callid=1)
NOTIFY(Callid=2) 100Trying
2O0K(Callid=2) forNOTIFY above
ACK(Callid=2)
INVITE(Callid=5) Codec=18,0,110,100
100 Trying (Callid=5)
180 Ringing (Callid=5)
PRACK Callid=5
20OK Callid=5
BYE(Callid=2)
2O0K(Callid=2)
INVITE(Callid=6) Codec=18,0,110,100
100 Trying (Callid=6)
180 Ringing (Callid=6)
PRACK Callid=6
PRACK Callid=6
NOTIFY Callid=3 2O0K
BYE(Callid=3)
2O0K(Callid=6)
Call is routed to RP2
AT&T initiates BYE on RP1


ATT
Sonus-ext
Sonus-int
SIPS
GVP(RM +MCP)
2O0K(Callid=4) Codec=18,0100
2O0K(Callid=5) Codec=18,0100
2O0K(Callid=3)
ACK(Callid=4)
ACK(Callid=5)
ACK(Callid=6)
BYE
BYE
BYE
200OK
200OK
200OK

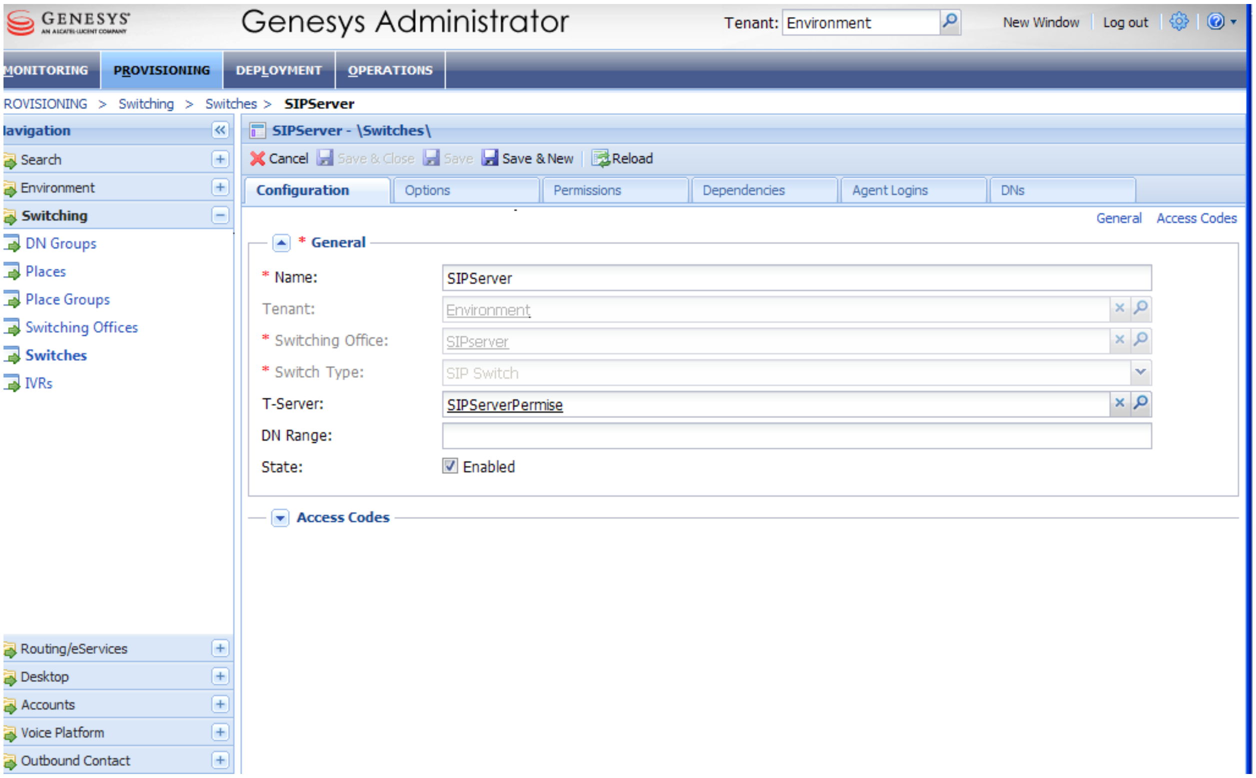
3.3. Genesys Configuration Section

This section explains the configuration of the Genesys components used for the interoperability testing.

3.3.1 Creating SIP Switch in Genesys Administrator (GA)

Step	Description
1.	Within GA, Create Switching Office – SIPS-Switching Office 

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>2.</td><td>Within GA, create a SIP Server Switch and associate the Switching Office created in previous step with this switch.



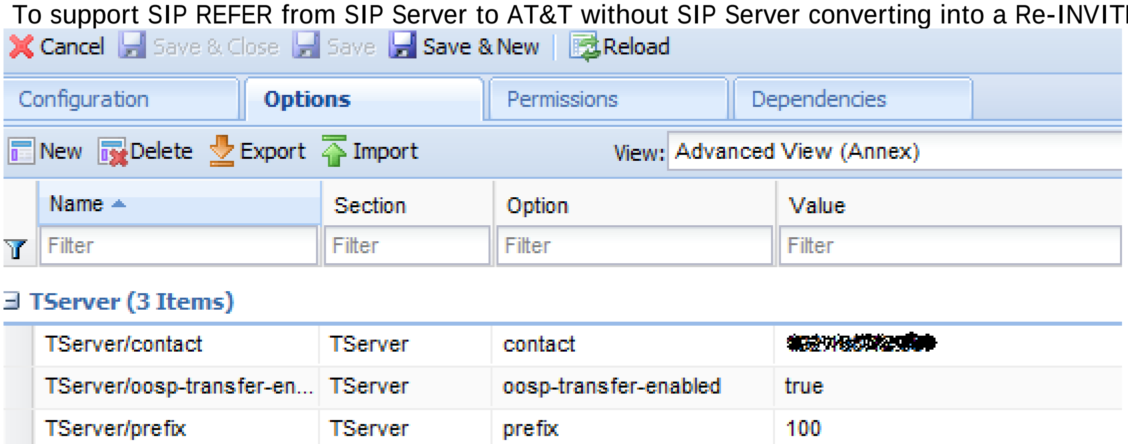
</td></tr>
<tr><td>3.</td><td>Under the SIPSwitch created in the above step, SIP trunks representing connection of SIP Server to Sonus, conneciton of SIP Server to RM and a “msml” voice over ip service DN required to integrate SIP Server with Media Server to support call hold, conferencing functionalities.

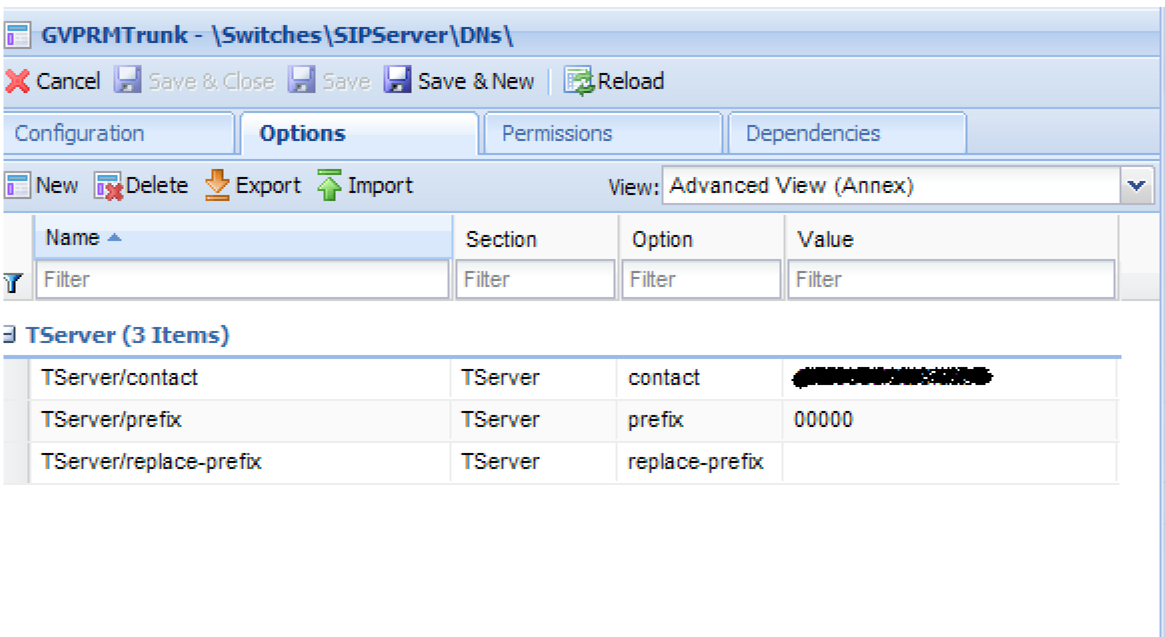
In subsequent steps of this section we will provide additional details required to configure these DNs.</td></tr>
</table>

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>4.</td><td>Define extensions under SIPSwitch with the following TServer Options for various SIP end points that will register to SIP Server.

use-register-for-service-state=true. This option sets DN in/Out of service based on SIP registration message received from SIP end point
contact – Specifies the contact address of the extension DN to which SIP Server should send the SIP call. Here the Contact option value is the IP address of the internal interface of Sonus through which the SIP REGISTER message was received by SIP Server.
sip-preserve-contact = true. This option allows SIP Server to format the Request-URI in the INVITE request that it sends to an endpoint by using an exact match to the value of the URI that is obtained from the Contact header of the SIP REGISTER request.
refer-enabled=false - To complete transfer or a blind transfer in 3pcc mode, SIP Server will send a re-INVITE to the end point with this setting.
sip-cti-control=talk,hold - The SIP method NOTIFY (event talk) or NOTIFY (event hold) is used request the end point to answer or place a call on hold respectively.

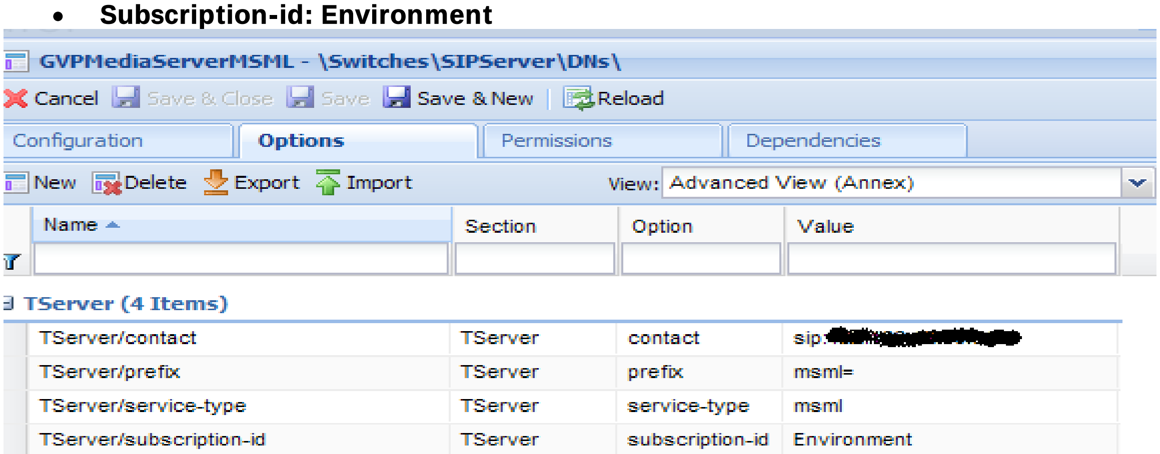
TServer (5 Items)
<table>
<tr><td>TServer/contact</td><td>TServer</td><td>contact</td><td>[redacted]</td></tr>
<tr><td>TServer/refer-enabled</td><td>TServer</td><td>refer-enabled</td><td>false</td></tr>
<tr><td>TServer/sip-cti-control</td><td>TServer</td><td>sip-cti-control</td><td>talk,hold</td></tr>
<tr><td>TServer/sip-preserve-contact</td><td>TServer</td><td>sip-preserve-...</td><td>true</td></tr>
<tr><td>TServer/use-register-for-service-state</td><td>TServer</td><td>use-register-f...</td><td>true</td></tr>
</table>
</td></tr>
</table>

Step	Description
5.	Define a Sonus switch trunk DN to represent all SIP calls arriving from Sonus NBS internal interface to SIP Server. Configure the following options under TServer section of the Trunk DNs. **sonusswitch Trunk DN below:** • **contact:** SIP signaling address of Sonus Internal Interface. Sonus internal interface IP address is used by SIP Server to route or receive calls from AT&T through this interface. • **prefix=100** Use this trunk with IP transfer(network based tranfser) for functionality using variable lenght (4 digit is used for this testing) of AT&T speed dial code. Speed dial code begin with prefix “100”. • **oosp-transfer-enabled=true** To support SIP REFER from SIP Server to AT&T without SIP Server converting into a Re-INVITE. 

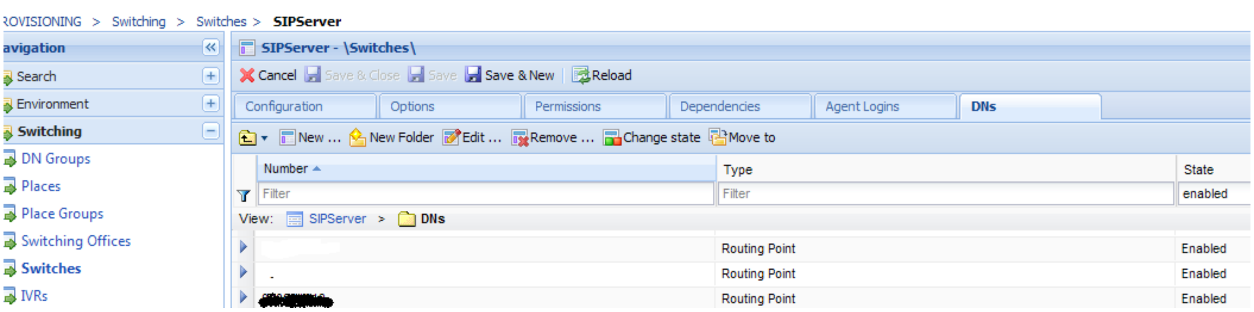
Step	Description
6.	Define a GVP trunk DN to represent calls from SIP Server to RM. Configure the following options under TServer section of the Trunk DNs. **sonusswitch Trunk DN below:** • **contact:** IP address of RM and Port. Sonus internal interface IP address is used by SIP Server to route or receive calls from AT&T through this interface. • **prefix=00000** Use this trunk with IP transfer and connect REFER functionality using 4 digit AT&T speed dial code is used. Speed dial code begin with prefix "00000". • **replace-prefix=** Specifies the characters that are inserted in the DN instead of the prefix for the gateway. If this annex is empty or absent, the initial characters that match the prefix option will be removed from the DN. This DN is matched to the DID configured in the IVR Profile. 

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>7.</td><td>
Defining a MSML voice over IP service DN with the following TServer options

- **Contact:** SIP address of RM: SIP listening port for Resource Manager
- **Prefix: msml=**
- **Service-type: msml=**
- **Subscription-id: Environment**



</td></tr>
<tr><td>8.</td><td>
For ADR scenarios URS strategy is used since SIP Server can map sip errors but not GVP. For URS to execute the strategy, a Routing Point is created on SIP Switch.



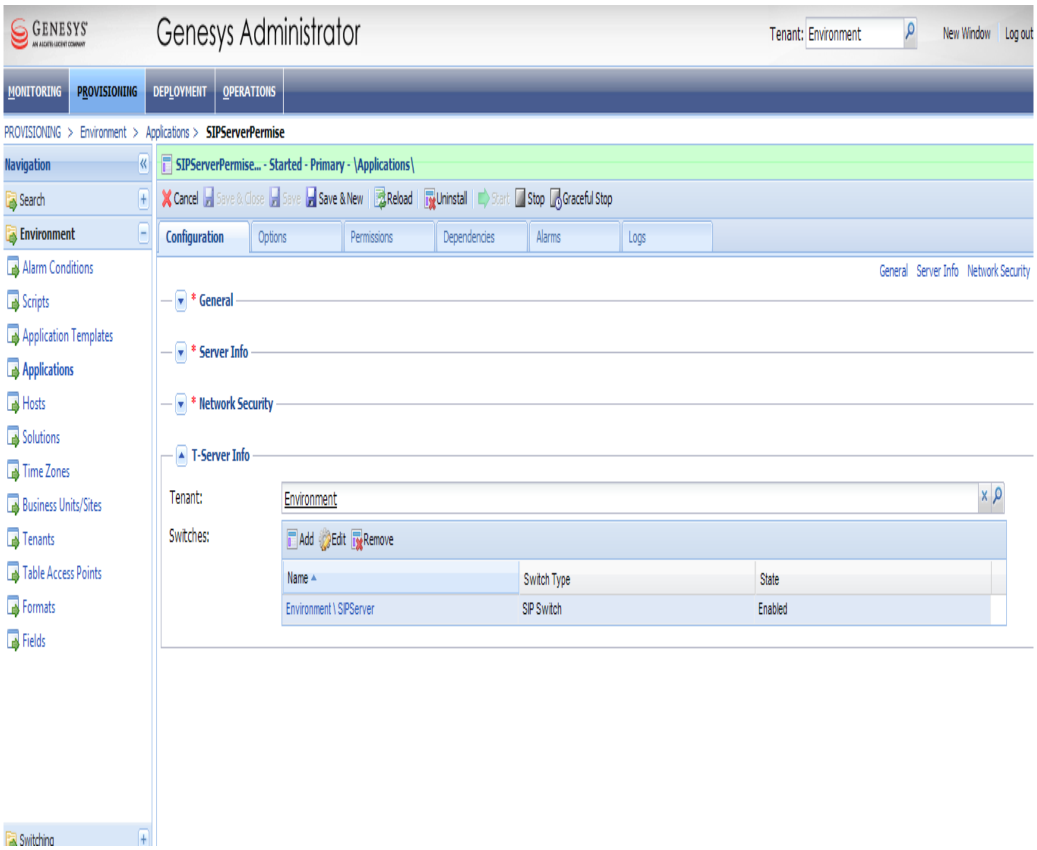
</td></tr>
<tr><td>9.</td><td>
SIP Server must have Full Control permission for the DN objects under SIP Server Switch in order to update various configuration objects under it like the Extension DNs.

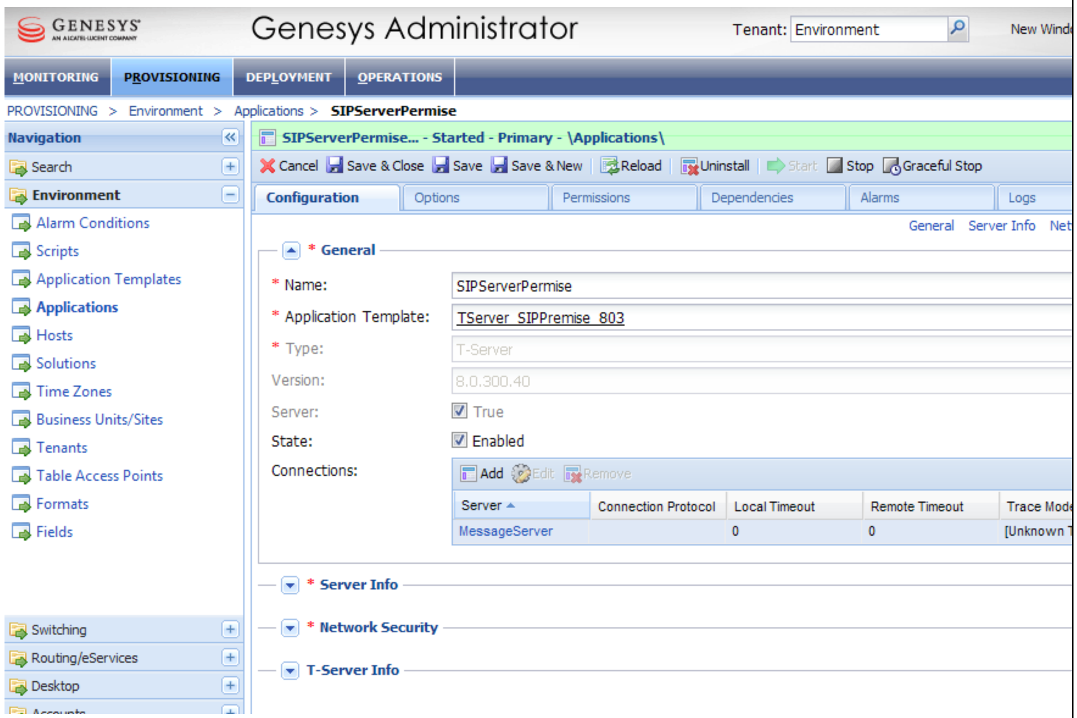
By default, it does not have this permission. You need to grant “Full Control” permission for the System account for the all DNs on the corresponding switch. It is done for all DNs at once by changing the permissions for the system account on the DN folder in the switch object. Or, you can start SIP Server under another account that has change permission on the necessary DNs.

With this full control access, SIP Server Switch grants DNs like Extension to update their option like “contact” when a new SIP register message is received from end points moving to a new IP location.
</td></tr>
</table>

3.3.2 SIP Server Configuration in GA

In this section we will configure the SIP Server to monitor the SIP Server Switch resources like SIP extensions/sip end points registered to SIP Server.

Step	Description
1.	Install and configure SIP Server as per Genesys Deployment Guide.
2.	Add connection to SIP Server Switch created above to monitor all the resources under this switch.(GA-> Provisioning->Environment->Applications->SIP Server Application Also, SIP Server should add a connection to the tenant 

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>3.</td><td>Add a connection to Message Server.

</td></tr>
<tr><td>4.</td><td>Configure the following TServer options on the SIP Server application object through Genesys Administrator or CME.
sip-address: A valid IP address
This option is specifically set if the SIP Server application is installed on a host having more than two Network Interfaces. The sip-address option should be set to the IP address of the network interface used for SIP signaling.
sip-enable-moh : true, Enables music on hold feature.
sip-treatments-continuous: true, Enables the strategy to play a treatment continuously until an agent answers the call.
enable-msml-media-services: true
msml-support: true
These two options allow SIP Server to integrated with Genesys Media server to provide msml/moml based media services.

Following SIP Server configuration options are needed to trigger Alternate Destination Routing (ADR) functionality within AT&T network:
map-sip-errors=false
Genesys Universal Router makes the routing decision for the SIP Server-based solution. If for some reason a call fails to route, then SIP Server generates an appropriate T-Library error message to inform the router. With this new parameter, SIP Server can now propagate SIP error messages to the router. Setting map-sip-errors=false triggers this functionality in SIP Server.
ringing-on-route-point=false
This option must be set to false to trigger ADR feature. Essentially setting this option to false suppresses 180 Ringing response to be sent to the calling side.</td></tr>
</table>

Step	Description
5.	**sip-dtmf-send-rtp=true (default=false)** In order to support DTMF tone generation on behalf of 3pcc based SIP end point application like Interaction Workspace SIP end point. When this option is set to true, SIP Server requests Media server to generate RFC 2833 DTMF tones on behalf of the end point. Some additional SIP Server settings are recommended to generate quicker error timeout when routing call to agent. **after-routing-timeout** set to 5 seconds. If SIP Server does not get a response on routing a call to SIP agent/Extension DN, it will AT&Tempt to try another DN (or default-dn) on expiration of this timer. Make sure to set this timer less than parameter - **rq-expire-tmout** value of 32000 (32 seconds). **rq-expire-tmout** – Request timeout value specified in milliseconds. This value is set to 32 seconds by default but can be explicitly set to a lower value in the application TServer section. **sip-invite-timeout=2** - specifies the number of seconds that SIP Server waits for a response to the INVITE message for a treatment (such as an announcement or music-on-hold). The call times out if no response is received. If the value is 0, or if a value is not specified, then the default SIP call timeout of 32 seconds is used.
6.	SIP Server is able to start properly with proper FlexLM license installed.

3.3.3 GVP Configuration in GA

Step	Description
1.	Install and configure MCP(Media control Platform) using Genesys Voice Portal deployment guide
2.	Within the MCP application's Connections tab, Add connections to SNMP Master Agent, Message Server and Reporting Server (optional). The connections to applications are added for the following reason - Message Server :- To ensure that component log information reaches the Log database and can be viewed in the Solution Control Interface (SCI) Reporting Server :- To ensure that these components detect the Reporting Server to which they are sending reporting data. (Optional) SNMP Master Agent :- To ensure that alarm and trap information is captured.
3.	Install and Configure Resource Manager as per Genesys Voice Platform Deployment guide
4.	**Note:** If SIP Server and Resource Manager are on the same machine, within Resource Manager application, all the default SIP listening port number should be increased by 100 so Resource Manager listening port is set to 5160 and SIP Server application listens in on port 5060. Make the necessary port changes within Resource Manager's sip, proxy, register, subscription and monitor sections.
5.	Within the Resource Manager application's Connections tab, Add connections to SNMP Master Agent, Message Server and Reporting Server (optional). The connections to applications are added for the following reason - Message Server :- To ensure that component log information reaches the Log database and can be viewed in the Solution Control Interface (SCI) Reporting Server :- To ensure that these components detect the Reporting Server to which they are sending reporting data. (Optional) SNMP Master Agent :- To ensure that alarm and trap information is captured.
6.	**Integrating Media Control Platform with the Resource Manager:-** Click the Media Control Platform Application object. The Configuration tab appears. Click the Options tab, and use the View drop-down list to select Show options in groups... Select sip to find the routeset option. In the Value field, type the following: • **<sip:IP_RM:SIPPort_RM;lr>** Where IP_RM is the IP address of the Resource Manager, and SIPPort_RM is the SIP port of the Resource Manager—typically, 5060. Note: You must include the angle brackets in the Value field in the sip.routeset and sip.secureroutеset parameters. In the Value field of the secureroutеset option, type the following: • **<sip:IP_RM:SIPSecurePort_RM;lr>**

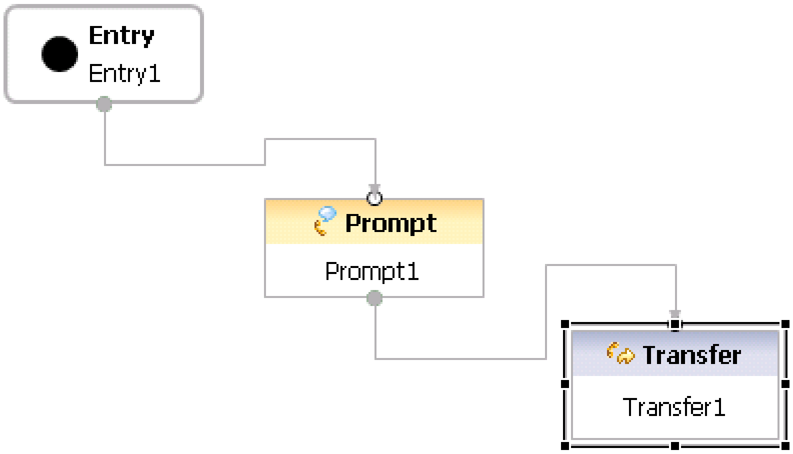
Step	Description
7.	Make sure VoIP service DN of type=msml as specified in section 3.3.1, Creating SIP Switch in Genesys Administrator step 6 above exists to support SIP Server-Media server MSML interactions to support treatments and conferencing capabilities.
8.	To play MOH and music treatments make sure the following options are set in MCP and SIP Server MCP->msml-> **play.basepath** = file://$InstallationRoot$ (this is installation folder of media server. After this is it will automatically look for the announcement sub folder) SIP Server->TServer->**msml-support** = true **NOTE:** • Sonus SBC is configured in such a way that if MCP sends 180 Ringing with SDP message, then Sonus will forward the request to AT&T as 183 Session in Progress. • If MCP sends 180 Ringing with no SDP message then Sonus forwards the same(180 Ringing) to AT&T. To achieve this scenario following paramters needed to be configured in MCP: **[sip]sdpansinprov** = 0, it means SDP message is suppressed in 180 Ringing message.
9.	G.729 media codec is not configured by default as a supported codec or as a codec that can be transcoded. This support can be enabled by adding "g729" as one of the values to the mpc.codec and mpc.transcoders space seperated list. Example: **mpc.transcoders**=PCM GSM G726 G729 **mpc.codec**=g729 pcmu pcma g726 gsm h263 h263-1998 h264 telephone-event In addition GVP (specifically MCP component) can be configured to respond a multiple codec offer request with a single codec response. This feature support is available starting with MCP 8.1.4 release. This setting can be enabled by setting **mpc.answerwithonecodec=1** (Default=0 – MCP responds to multiple codec offer with a multiple codec response list).

Step	Description
10.	**Creating a Resource Group:** 1. This resource Group associate all calls coming to the GVP (Resource Manager or MCP) from a gateway resource. This gateway resource will typically be SIP Server, since SIP Server serves as the initial ingress point to Genesys Voice Platform. This gateway resource will also serve as the egress point when GVP initiates outbound calls or transfers. • On the Provisioning tab, click Voice Platform > Resource Groups. • On the Details pane tool bar, click New. • The Resource Group Wizard opens to the Welcome page. • On the Resource Manager Selection page, add the Resource Manager Application object for which you want to create the group. On the Group Name and Type page: Enter SIPServerGWgroup or any custom name without spaces. Select type as Gateway. • On the Tenant Assignments page, add the child tenant to which the Resource Group will be assigned. **Note:** -The above bullet item is required only if you are creating the Resource Group in a multi-tenant environment. • On the Group Properties page, enter the information as specified below for the Resource Group that you are configuring. **Monitoring Method** - retain the default value: SIP OPTIONS. **Load Balance Scheme** - select round-robin. **CTI usage** - off • On the Resource Assignment page, Allocate the SIP Server resources that will be part of the resource group. Once the call reaches Resource Manager from the gateway resource, Resource Manager should direct the call to either MCP resource based on the IVR profile lookup performed by Resource Manager. For that purspose MCP Resource Group is created below as specified in step 11.

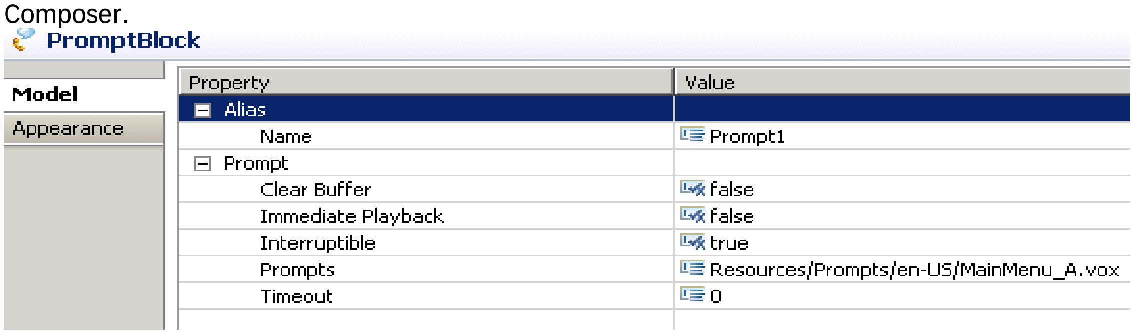
Step	Description
11.	Log in to Genesys Administrator. • On the Provisioning tab, click Voice Platform > Resource Groups. • On the Details pane tool bar, click New. • The Resource Group Wizard opens to the Welcome page. • On the Resource Manager Selection page, add the Resource Manager Application object for which you want to create the group. On the Group Name and Type page: Enter MCPGroup or any custom name without spaces. Select type as Media Control Platform. • On the Tenant Assignments page, add the child tenant to which the Resource Group will be assigned. **Note:** -The above bullet item is required only if you are creating the Resource Group in a multi-tenant environment. • On the Group Properties page, enter the information as specified below for the Resource Group that you are configuring. **Monitoring Method** - retain the default value: SIP OPTIONS. **Load Balance Scheme** - select round-robin. **Port Usage Type** - select in-and-out. **Maximum Conference Size** Enter -1. **Maximum Conference Count – leave blank** Note: For the Media Control Platform group, the Max.Conference Size and Max.Conference Count, and Geo-location options are optional; • On the Resource Assignment page, Allocate the MCP resources that will be part of the resource group. The SIP port and SIPS port specified here are the SIP listening port used by the MCP for non-secure and secure SIP signaling. For a complete list of resource-group options and their descriptions, see the Genesys Voice Platform User’s Guide.

Step	Description
12.	**Defining IVR profiles** • On the Provisioning tab, click Voice Platform > IVR Profiles in the tasks menu, provisioning option on the right side of the browser window, choose Define New IVR Profile • On the Service Type page specify the **IVR Profile Name** **Service Type** - voicexml • On the Service Properties page specify the following **Intial Page URL** - <location of the voicexml page> **Voice XML Interpretor** - Next Generation • On the Usage Limits page specify the following **Maximum Concurrent Sessions** - <maximum number of calls that can be allowed for this particular IVR profile>. • On the IVR Capabilities page, enable the following **Allow Outbound calls** (SIP INVITE) **Allow Transfers**(SIP REFER) **Gateway Selection-** < choose appropriate gateway to decide which Gateway will Resource Manager choose to route outbound calls or call transfers initiated by GVP>. • On the CTI Paramters page uncheck the following **Require CTI interaction** • On the Dailing Rules page **Regular Expression** - <enter the expression in the form of a URL>. • On the Policies page **SQ notification threshold** - <Provide value between 1-100>
13.	**Creating DID Groups:** On the Provisioning tab, click Voice Platform > DID Groups.Select New. In the **Name** field - <enter the name of the DID Group>. In the **IVR Profile** field - <click the browse icon to find the IVR Profile or tenant that you want to associate with this DID Group>.

3.3.4 VXML Applications using Composer

Step	Description
1.	This section shows examples of various VXML applications used during the test. **<u>VXML Application#1 - Routing call to Agent</u>** The application shows that when the caller dials the toll free number, the call is received on this SIP Server and then reaches GVP which executes a VXML application that plays the prompt and routes the call to SIP extension/end point registered to SIP Server 

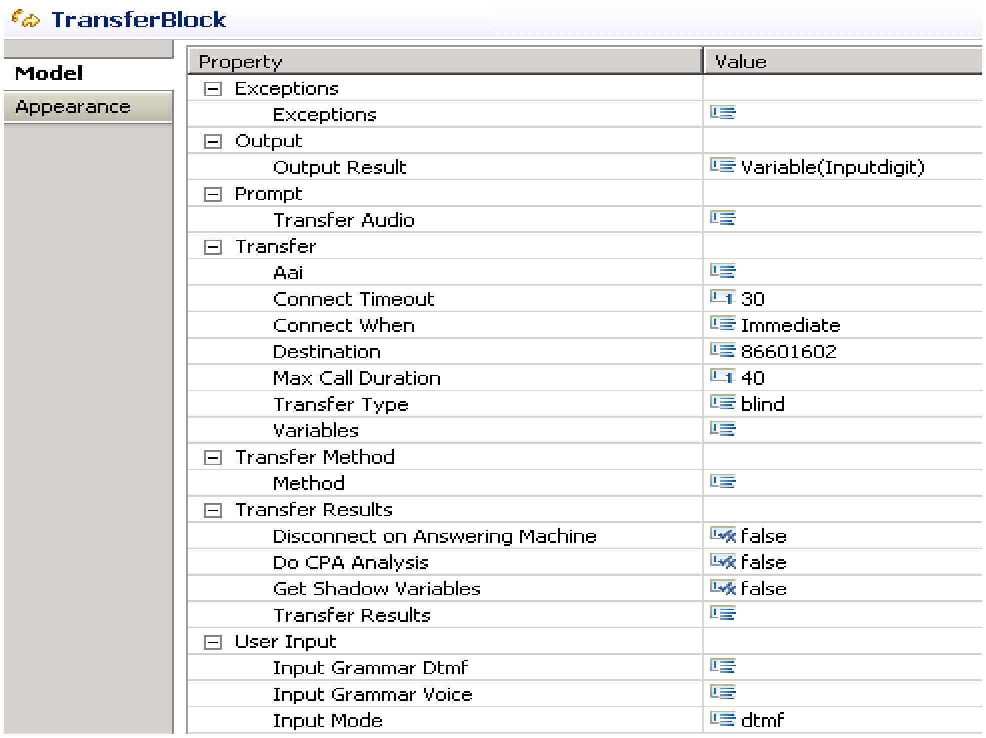
Step	Description
2.	Set the following options for Prompt and Transfer block in the Voice XML application created in Composer.



PromptBlock

Model | Appearance

Property	Value
⊟ Alias	
Name	Prompt1
⊟ Prompt	
Clear Buffer	false
Immediate Playback	false
Interruptible	true
Prompts	Resources/Prompts/en-US/MainMenu_A.vox
Timeout	0

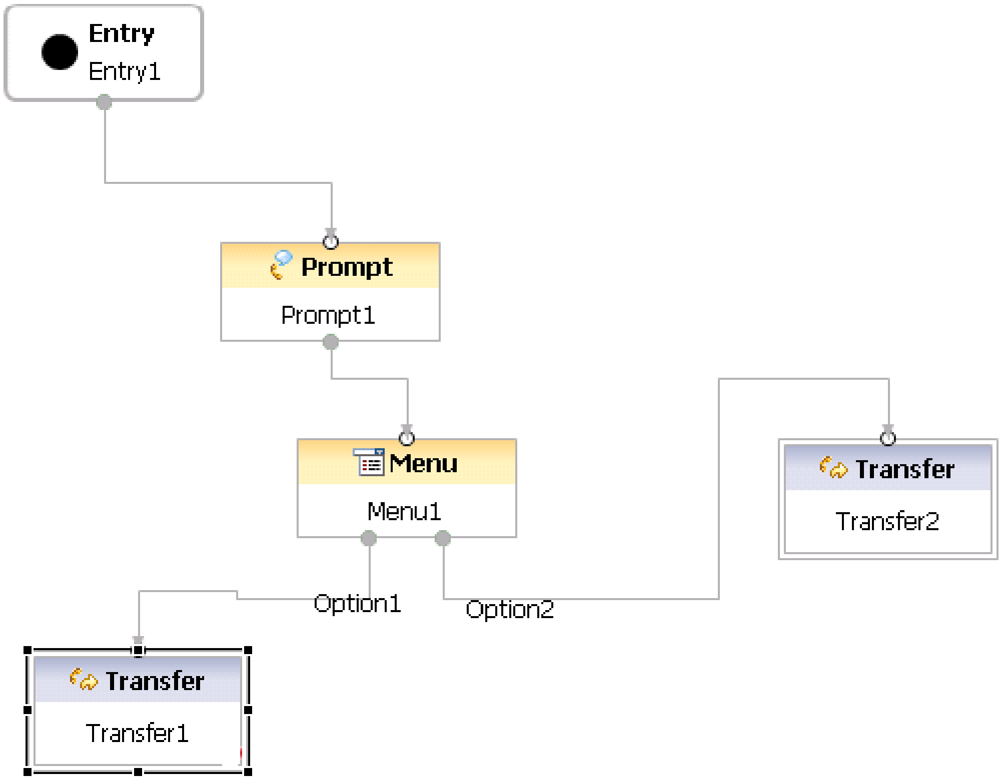


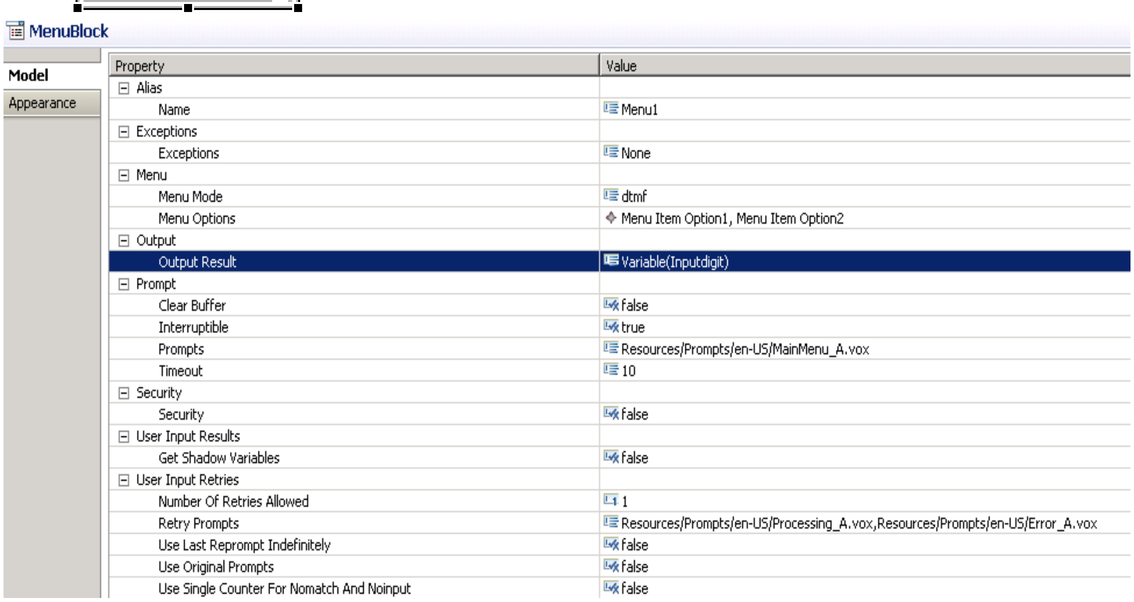
TransferBlock

Model | Appearance

Property	Value
⊟ Exceptions	
Exceptions	
⊟ Output	
Output Result	Variable(Inputdigit)
⊟ Prompt	
Transfer Audio	
⊟ Transfer	
Aai	
Connect Timeout	30
Connect When	Immediate
Destination	86601602
Max Call Duration	40
Transfer Type	blind
Variables	
⊟ Transfer Method	
Method	
⊟ Transfer Results	
Disconnect on Answering Machine	false
Do CPA Analysis	false
Get Shadow Variables	false
Transfer Results	
⊟ User Input	
Input Grammar Dtmf	
Input Grammar Voice	
Input Mode	dtmf

Step	Description
3.	**<u>VXML Application #2 - Play Announcement and Collect User Input from Caller and route call to appropriate agent based on User Input</u>**. VXML application that plays announcement and depending on the user enter digits the call is routed to the agent.



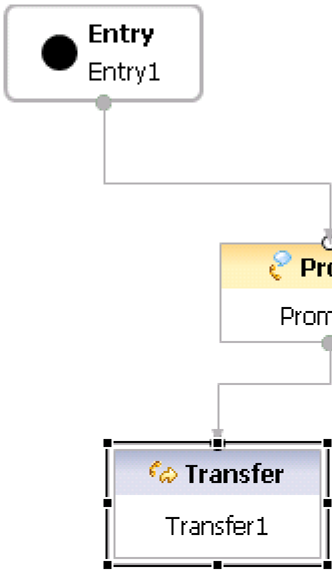


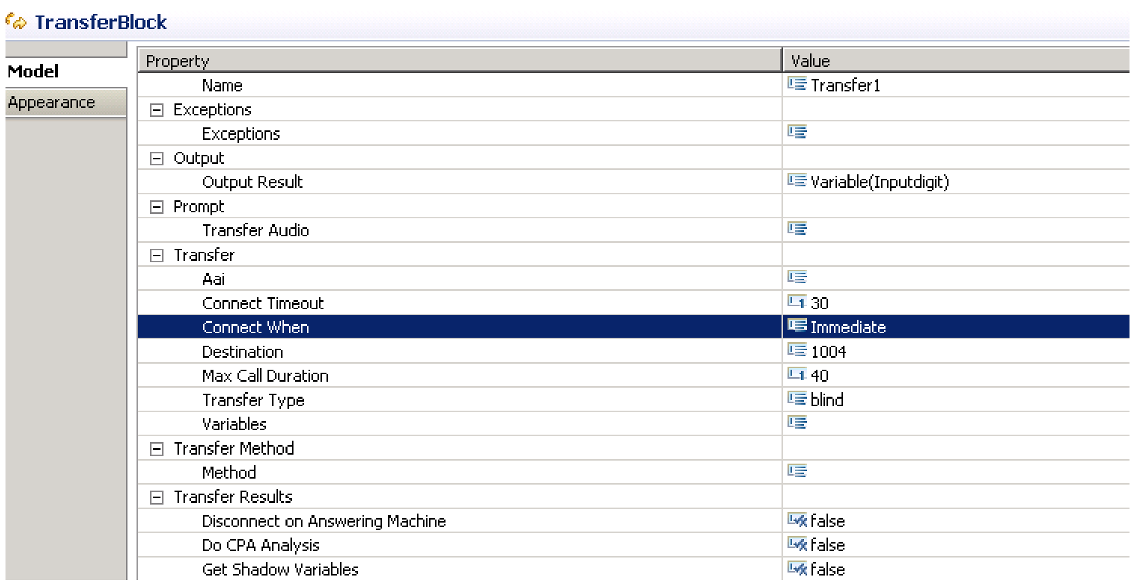
MenuBlock

Model | Appearance

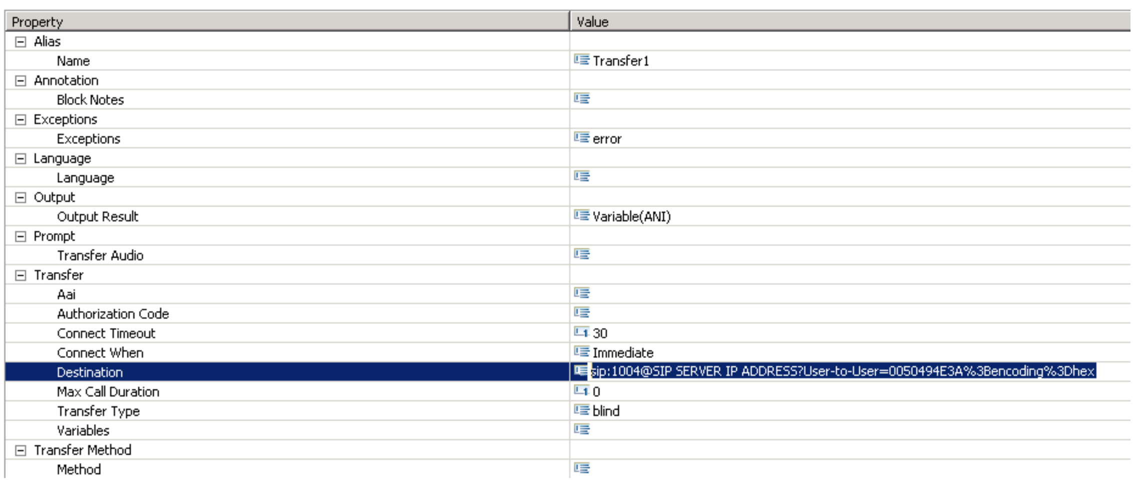
Property	Value
⊟ Alias	
Name	Menu1
⊟ Exceptions	
Exceptions	None
⊟ Menu	
Menu Mode	dtmf
Menu Options	Menu Item Option1, Menu Item Option2
⊟ Output	
Output Result	Variable(Inputdigit)
⊟ Prompt	
Clear Buffer	false
Interruptible	true
Prompts	Resources/Prompts/en-US/MainMenu_A.vox
Timeout	10
⊟ Security	
Security	false
⊟ User Input Results	
Get Shadow Variables	false
⊟ User Input Retries	
Number Of Retries Allowed	1
Retry Prompts	Resources/Prompts/en-US/Processing_A.vox,Resources/Prompts/en-US/Error_A.vox
Use Last Reprompt Indefinitely	false
Use Original Prompts	false
Use Single Counter For Nomatch And Noinput	false

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>4.</td><td>
For AT&T IP Transfer and Connect SIP REFER based transfer to a 4 digit AT&T speed dial code, two VXML applications are required.

First VXML application (<u>Application #3 – Post answer call and then REFER the caller to a 4 digit AT&T speed dial code</u>)




<table>
<tr><th>Property</th><th>Value</th></tr>
<tr><td>Name</td><td>Transfer1</td></tr>
<tr><td>⊟ Exceptions</td><td></td></tr>
<tr><td>Exceptions</td><td></td></tr>
<tr><td>⊟ Output</td><td></td></tr>
<tr><td>Output Result</td><td>Variable(Inputdigit)</td></tr>
<tr><td>⊟ Prompt</td><td></td></tr>
<tr><td>Transfer Audio</td><td></td></tr>
<tr><td>⊟ Transfer</td><td></td></tr>
<tr><td>Aai</td><td></td></tr>
<tr><td>Connect Timeout</td><td>30</td></tr>
<tr><td>Connect When</td><td>Immediate</td></tr>
<tr><td>Destination</td><td>1004</td></tr>
<tr><td>Max Call Duration</td><td>40</td></tr>
<tr><td>Transfer Type</td><td>blind</td></tr>
<tr><td>Variables</td><td></td></tr>
<tr><td>⊟ Transfer Method</td><td></td></tr>
<tr><td>Method</td><td></td></tr>
<tr><td>⊟ Transfer Results</td><td></td></tr>
<tr><td>Disconnect on Answering Machine</td><td>false</td></tr>
<tr><td>Do CPA Analysis</td><td>false</td></tr>
<tr><td>Get Shadow Variables</td><td>false</td></tr>
</table>
</td></tr>
</table>

Step	Description
	A second VXML Application - [Application #1](#) (specified in step 1) that can re-utilized is loaded on the second Route point to route the call to a SIP end point. For detailed SIP IP transfer and connect REFER call flow, please refer the [3.2.4 Basic IPXC REFER](#) section and refer the sub-section "b. IPXC REFER Inbound
5	To get User to User data from an INVITE, create a variable in the composer start block  Note the name UU_Data can be any value the user chooses To populate User to User data in a transfer request (REFER) enter the data in the destination field  Where sip 1004@SIP SERVER Address is the AT&T speed dial code followed by the SIP Server address. ?User-to-User=0050494E3A%3Bencoding%3Dhex is what will be added to the REFER TO header in the transfer request

3.4. Sonus Configuration

3.4.1. High level description of Sonus components

GSX

GSX9000. The Sonus GSX is a high-density, high performance Media Gateway. This is multi-slotted chassis that contains management, TDM, and IP cards as part of a distributed architecture. The GSX9000 is one of the most widely deployed Media Gateways in the world and is at the heart of many of the world's largest carrier and enterprise networks.

GSX4000

GSX4000. The Sonus GSX4000 Open Services Switch allows service providers to realize the benefits of the GSX9000 on a platform that is right-sized for their immediate needs. By delivering the proven reliability of the GSX9000 in a smaller form factor, service providers are able to cost-effectively support a highly distributed subscriber base or create an initial presence in a new market. The Sonus GSX portfolio offers an expansion path from several hundred ports to several million ports to meet the demands of the worldwide market.

NBS

NBS. The Sonus NBS is a security and session control solution based on the Sonus GSX9000. The NBS is deployed at the packet peering point where two VoIP networks interconnect in order to provide the necessary media inter-working, administrative controls, and security. The NBS is a software enhancement to the GSX9000 Gateway. This is a purpose-built voice switch that has been in production for over 11 years.

PSX

PSX. Sonus' advanced PSX policy and routing server brings all of the network's call routing into the IP core, enabling Wells Fargo to efficiently route calls over an IP network rather than through a series of tandem switches. This centralized routing scheme allows network operators to enjoy the lower operating costs of IP technology, enhanced features and more efficient use of their existing network resources.

EMS. The Insight EMS platform is used as the Element Management Server. It implements operations, administration, maintenance, and configuration functions for Sonus system
elements. The EMS has robust reporting capabilities and goes beyond typical "box-level" statistics to provide information that is critical to ensuring quality of service.

DSI

DSI. The Sonus DataStream Integrator (DSI) creates an open channel of communication between Sonus network elements and a wide range of third-party and proprietary back-office systems. The DSI solution provides rich insight into your network by delivering call and network data to a host of back-office applications—including traffic management—via customized DSI adaptors.

NBS5200. Based on Sonus' new IP to IP session management platform – ConnexIP™ , announced in May 2010, the NBS5200 provides a reliable, scalable solution for session border control and delivers the longstanding strengths of the Sonus NBS9000---security, session control, bandwidth management, advanced media services and integrated billing/reporting tools, in a smaller, 2U form-factor. The NBS5200 represents the next generation of session border control and is the first NBS to include media transcoding, robust security and advanced call routing in a high-performance, small form-factor device. Encryption is available in the form of TLS and IPSec for signaling and Secure RTP for media. Other security features include traffic policing, DoS, D-DOS and Rogue RTP protection. With on-board DSP processing modules, the NBS5200 provides integrated media services including media transcoding, DTMF relay and interworking and support for data (modem) and T.38 fax relay or interworking. The NBS5200 includes a local Sonus PSX server for advanced routing in standalone mode or can be configured to access a centralized PSX or 3rd party Softswitch.

Step	Description
1.	Sonus GSX has 3 physical interfaces, "external/un-trusted interface", "internal/trusted interface" and "management interface". The Sonus configuration has been split into 4 parts which applies configuration changes to GSX and PSX. 1. Basic configuration to setup GSX and PSX. 2. GSX and PSX Configuration essential for Sonus interaction with Genesys SIP Server. 3. GSX and PSX Configuration essential for Sonus interaction with carrier/network. 4. GSX and PSX Configuration essential to setup sip end points that register to Genesys SIP Server through the Sonus NBS. **Note: In the configuration and screenshots shown below, the IP addresses of the host have been hidden.**

3.4.2. GSX CLI commands to create and configure the base config of the GSX prior to adding Genesys, Carrier or Agent IPTGs.

At a high level, following configuration is performed as part of base configuration.

1. CREATE GSX NODE labgsx01 (name must match GSX Gateway in PSX)
2. CREATE CARRIER genesys (this will be assigned to Trunkgroups)
3. CREATE NTP SERVER dsi1 (system timing used for billing records)
4. CONFIG NIF for RTP and SIP signaling (Public and Private)
5. CREATE NIFGROUP FOR PUBLIC INTERFACE
6. CREATE NIFGROUP FOR PRIVATE(Internal) INTERFACE
7. CREATE SIGNALING ZONE FOR EXTERNAL and INTERNAL SIGNALING
8. CREATE SIP SIGNALING PORTS INTERNAL and EXTERNAL (Assign Zone and NifGroup to each)
9. CREATE SOFTSWITCH (PSX) FOR ROUTING
10. CREATE STATIC ROUTES ON NIFS (nexthop for signaling and media)

```
#############################################################################
#         CREATE GSX NODE labgsx01 (name must match GSX Gateway in PSX)
#
#############################################################################

CREATE NODE labgsx01
CONFIG NODE labgsx01 TELNET ENABLED LOCATION SF01 MODE INSERVICE

SHOW NODE ADMIN

Node: labgsx01                                 Date: 2010/12/03 03:39:51  GMT
                                               Zone: GMTMINUS05-EASTERN-US

Name:          labgsx01
Contact:       None
Location:      SF01
Mode:          INSERVICE
Telnet Access: ENABLED

#############################################################################
#        CREATE CARRIER genesys (this will be assigned to Trunkgroups)
#
#############################################################################

CREATE CARRIER genesys
CONFIG CARRIER CODE 9999 sTATE ENABLED

SHOW CARRIER genesys ADMIN

Node: labgsx01                                 Date: 2010/12/03 03:42:46  GMT
                                               Zone: GMTMINUS05-EASTERN-US

     Carrier Name         Code   Type      Network  Plan           State
----------------------   ---- -------- ------------------------- ---------
genesys                  9999 NATIONAL N-4_DIGIT_CARRIERCODE      ENABLED

#############################################################################
```

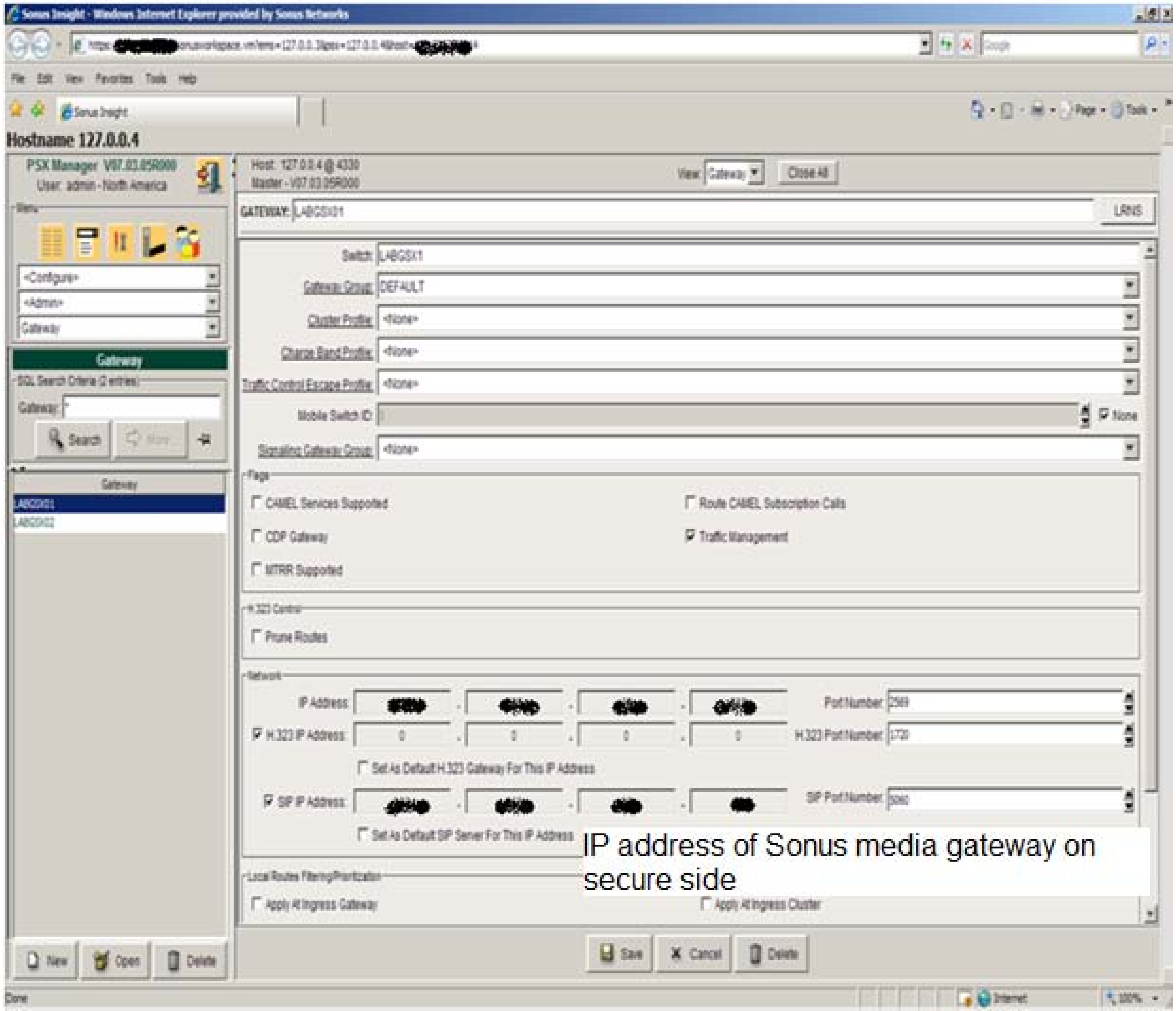
```
#       CREATE NTP SERVER dsi1 (system timing used for billing records)
#
#############################################################################

CREATE NTP SERVER dsi1
CONFIG NTP SERVER dsi1 IPADDRESS 1XX.2XX.6X.2XX STATE ENABLED

SHOW NTP SERVER dsi1 ADMIN

Node: labgsx01                              Date: 2010/12/03 03:43:37  GMT
                                            Zone: GMTMINUS05-EASTERN-US

Server                  IpAddress      Client    Vers MinPoll MaxPoll State
                                                      (2^x)S  (2^x)S
----------------------- -------------- --------- ---- ------- ------- ------
dsi1                    1XX.2XX.6X.2XX POLL       3     3       10    ENABLED


#############################################################################
#         CONFIG NIF for RTP and SIP signaling  (Public and Private)        #
#############################################################################

CONFIG NIF ENET-1-3 IPADDRESS 19X.1X8.1X.X9 MASK 255.255.255.240
CONFIG NIF ENET-1-3 STATE ENABLED
CONFIG NIF ENET-1-3 MODE INSERVICE

CONFIG NIF ENET-1-4 IPADDRESS 1X9.XX.2XX.XX MASK 255.255.255.224
CONFIG NIF ENET-1-4 STATE ENABLED
CONFIG NIF ENET-1-4 MODE INSERVICE

SHOW NIF ALL ADMIN

Node: labgsx01                              Date: 2010/12/03 03:45:57  GMT
                                            Zone: GMTMINUS05-EASTERN-US

loc  port name                    index      mode     IP address      deviatn
          type                    class      action   mask            conting
                                  state      timeout  nexthop
                                  DGPstate            DGP bucket      DGP rate
                                  PVPstate            PVP bucket      PVP rate
                                  LGPstate   xnq
                                  vstate     tag      IPsec Policy
---- ---- ----------------------- ---------- -------- --------------- ------
-
1-1  3    ENET-1-3                31         IS       19X.1X8.1X.X9   0
          ETHERNETCSMACD          GENERAL    DRYUP    255.255.255.240 0
                                  ENABLED    60       0.0.0.0
                                  DISABLED            N/A              N/A
                                  DISABLED            24992           1048576
                                  DISABLED   DISABLED
                                  DISABLED   0
1-1  4    ENET-1-4                32         IS       1X9.XX.2XX.XX   0
          ETHERNETCSMACD          GENERAL    FORCE    255.255.255.224 0
                                  ENABLED    60       0.0.0.0
                                  DISABLED            N/A              N/A
                                  DISABLED            24992           1048576
                                  DISABLED   DISABLED
                                  DISABLED   0

#############################################################################
#                   CREATE NIFGROUP FOR PUBLIC INTERFACE
#############################################################################
```

```
CREATE NIFGROUP NG_OUT
CONFIG NIFGROUP NG_OUT INTERFACE ENET-1-4
CONFIG NIFGROUP NG_OUT STATE ENABLED

SHOW NIFGROUP NG_OUT ADMIN

Node: labgsx01                              Date: 2010/12/03 03:44:27  GMT
                                            Zone: GMTMINUS05-EASTERN-US

NIF Group:    NG_OUT
Admin State:  ENABLED

Interface Name          Interface State
---------------------------------------
ENET-1-4                ENABLED

#############################################################################
#                   CREATE NIFGROUP FOR PRIVATE(Internal) INTERFACE
#############################################################################

CREATE NIFGROUP NG_IN
CONFIG NIFGROUP NG_IN INTERFACE ENET-1-3
CONFIG NIFGROUP NG_IN STATE ENABLED


SHOW NIFGROUP NG_IN ADMIN

Node: labgsx01                              Date: 2010/12/03 03:44:51  GMT
                                            Zone: GMTMINUS05-EASTERN-US

NIF Group:    NG_IN
Admin State:  ENABLED

Interface Name          Interface State
---------------------------------------
ENET-1-3                ENABLED

#############################################################################
#           CREATE SIGNALING ZONE FOR EXTERNAL and INTERNAL SIGNALING
#############################################################################

CREATE ZONE SZ_INSIDE
CREATE ZONE SZ_OUTSIDE

SHOW ZONE ALL ADMIN

Node: labgsx01                              Date: 2010/12/03 03:48:52  GMT
                                            Zone: GMTMINUS05-EASTERN-US

-----------------------------------------------------
 Zone Name:                     INTERNAL
-----------------------------------------------------
 Zone ID:                       0
 TCP Connect Timeout:           5
 Transport Protocols:           SIP-UDP
 TLS Profile Name:              defaultTlsProfile


-----------------------------------------------------
 Zone Name:                     SZ_INSIDE
-----------------------------------------------------
 Zone ID:                       1
 TCP Connect Timeout:           5
 Transport Protocols:           SIP-UDP
```

```
 TLS Profile Name:             defaultTlsProfile


-----------------------------------------------------
 Zone Name:                    SZ_OUTSIDE
-----------------------------------------------------
 Zone ID:                      2
 TCP Connect Timeout:          5
 Transport Protocols:          SIP-UDP
 TLS Profile Name:             defaultTlsProfile

############################################################################
#          CREATE SIP SIGNALING PORTS INTERNAL and EXTERNAL                #
#          (Assign Zone and NifGroup to each)                              #
############################################################################

CREATE SIP SIGNALING PORT IPADDRESS 1XX.1XX.12.XX
CONFIG SIP SIGNALING PORT 1 INTERFACE NIF ZONE INTERNAL NIFGROUP NG_IN
CONFIG SIP SIGNALING PORT 1 STATE ENABLED
CONFIG SIP SIGNALING PORT 1 MODE INSERVICE

CREATE SIP SIGNALING PORT IPADDRESS 1XX.XX.2XX.XX
CONFIG SIP SIGNALING PORT 2 INTERFACE NIF ZONE SZ_OUTSIDE NIFGROUP NG_OUT
CONFIG SIP SIGNALING PORT 2 STATE ENABLED
CONFIG SIP SIGNALING PORT 2 MODE INSERVICE

SHOW SIP SIGNALING PORT ALL ADMIN

Node: labgsx01                               Date: 2010/12/03 03:47:18  GMT
                                             Zone: GMTMINUS05-EASTERN-US

                                                        Sig. Zone/
Port Pri IP Addr/                   Mode/      State/     NIF Group/
Num  Sec IP Addr     Port   Intf.   Recorder   Slot       Sctp Profile
UDP Checksum State
---- --------------- -----  ------ ---------- -------- ----------------
   1 1XX.1XX.12.XX    5060  NIF    INSERVICE  ENABLED  INTERNAL
     0.0.0.0                       DISABLED          1 NG_IN
                                                       defaultSctpProfile
DISABLED
   2 1XX.XX.2XX.XX    5060  NIF    INSERVICE  ENABLED  SZ_OUTSIDE
     0.0.0.0                       DISABLED          1 NG_OUT
                                                       defaultSctpProfile
DISABLED




############################################################################
#            CREATE SOFTSWITCH FOR ROUTING                                 #
############################################################################

CREATE SONUS SOFTSWITCH psx1
CONFIG SONUS SOFTSWITCH psx1 IPADDRESS 1XX.2XX.6X.2XX
CONFIG SONUS SOFTSWITCH psx1 STATE ENABLED
CONFIG SONUS SOFTSWITCH psx1 MODE ACTIVE

SHOW SONUS SOFTSWITCH ALL ADMIN

Node: labgsx01                               Date: 2010/12/03 13:44:48  GMT
                                             Zone: GMTMINUS05-EASTERN-US

Index SoftSwitchName          IpAddress       Port  SubPort  Mode      State
```

```
---------------------------------------------------------------------------
   1  psx1                    1XX.2XX.6X.2XX  3055  3053     ACTIVE   ENABLED


#############################################################################
#       CREATE STATIC ROUTES ON NIFS (nexthop for signaling and media)      #
#############################################################################

CONFIGURE IP ROUTE ADD IFINDEX 31 IPADDRESS 0.0.0.0 MASK 0.0.0.0 NEXTHOP
1XX.1XX.1X.33
CONFIGURE IP ROUTE ADD IFINDEX 32 IPADDRESS 0.0.0.0 MASK 0.0.0.0 NEXTHOP
1XX.XX.2XX.65

SHOW IP NETSTAT ALL ROUTES ADMIN
Node: labgsx01                               Date: 2010/12/06 15:32:32  GMT
                                             Zone: GMTMINUS05-EASTERN-US

Destination     Mask            Nexthop         Index  Pref
--------------- --------------- --------------- -----  ----
0.0.0.0         0.0.0.0         1XX.1XX.1X.33   31     10
0               0.0.0.0         199.79.227.65   32     10
```

3.4.3. PSX GUI tables that need to be populated prior to building specific IPTGs and routing.



Screenshot 1/1

3.4.4 GSX CLI commands for creating Genesys SIP Server IPTG

At a high level, following configuration changes are required on GSX to connect Sonus to SIP Server through an IP Trunk group.

1. Create Network Selector Table for ingress TG determination
2. Create TG GF_INT_TG (same name in PSX TG) - Config with Network selector Table GF_INT
3. Create SIP Service under TG GF_INT_TG - Config Sig Zone, Nif Group, Session Timer and Out Adaptor
4. Fix in SONUS00106060 sip frag body support in Notify - Delete the X-Jmac header and create VAR1
5. Modify the Content-Type header with value of VAR1

```
##############################################################
# Create Network Selector Table for ingress TG determination #
##############################################################


CREATE IP NETWORK SELECTOR TABLE GF_INT
CONFIGURE IP NETWORK SELECTOR TABLE GF_INT ADD NUMBER 1XX.1XX.12.XX MASK
255.255.255.0
SHOW IP NETWORK SELECTOR TABLE GF_INT ADMIN

Node: labgsx01                               Date: 2010/12/03 18:33:51  GMT
                                             Zone: GMTMINUS05-EASTERN-US

Table Name                      Network Number    Network Mask
-----------------------------------------------------------------------
GF_INT                        1XX.1XX.12.XX   255.255.255.255

###############################################################
#         Create TG GF_INT_TG (same name in PSX TG)           #
#         Config with Network selector Table GF_INT           #
###############################################################

CREATE TRUNK GROUP GF_INT_TG
CONFIGURE TRUNK GROUP GF_INT_TG NETWORK SELECTOR TABLE GF_INT


SHOW TRUNK GROUP GF_INT_TG ad
Node: labgsx01                               Date: 2010/12/03 18:31:15  GMT
                                             Zone: GMTMINUS05-EASTERN-US


Local Trunk Name: GF_INT_TG
----------------------------------------------------------------------------
 State                                         ENABLED
 Inbound Reserve (percent)                     0
 Mode                                          INSERVICE
 Action                                        DRYUP
 Timeout (min)                                 5
 Circuit Reservation State                     DISABLED
 Reserved Priority Calls (circuits)            1
 Reserved Incoming Calls (circuits)            1
 Reserved Outgoing Calls (percent)             10
 Alternate Trunk Group Name
 Trunk Group Rename Timer (sec)                10
 SILC State                                    DISABLED
 SILC Congestion Level 1 Calls Allowed (percent) 075
 SILC Congestion Level 2 Calls Allowed (percent) 025
 Trunk Group Type                              IPSELECTED
```

```
 IP Trunk Group Direction                                BOTHWAYS
 Parent IP Trunk Group
 IP Network Selection Table                              GF_INT
 IP Call Limit                                           UNLMT
 IP Bandwidth Limit                                      UNLMT
 Packet Outage Detection Minimum Duration                6000
 Packet Outage Detection Minimum Calls                   1000
 Packet Outage Detection Bandwidth Limit Reduct          50
 Packet Outage Detection State                           DISABLED
 Packet Outage Detection Interval (minutes)              15
 Master Trunk Group Name
 Calls Requested Per MTRG Request                        100
 Bandwidth Requested Per MTRG Request (1K bps)           12400
 Maximum Ingress Sustained Call Rate                     0
 Maximum Ingress Call Burst Size                         0
 Maximum Ingress Sustained SIP nonInvite Rate            0
 Maximum Ingress SIP nonInvite Burst Size                0
 Maximum Egress Sustained Call Rate                      0
 Maximum Egress Call Burst Size                          0
 Maximum Egress Sustained SIP nonInvite Rate             0
 Maximum Egress SIP nonInvite Burst Size                 0
 Ingress NonPriority Call Threshold                      0
 Egress NonPriority Call Threshold                       0
 HPC Profile Name                                        defaultintipqueuing
 HPC Early ACM or SIP-18X                                USEDEFAULT
 HPC IP Oversubscription 0verride                        DISABLED
 HPC IP Oversubscription Factor                          10
 Emergency IP Oversubscription Factor                    10
 Local Policy Trunk Profile
 IP Registration Limit                                   UNLMT
 IP Estimated Child Registrations                        1

########################################################################
#         Create SIP Service under TG GF_INT_TG                        #
#         Config Sig Zone, Nif Group, Session Timer and Out Adaptor    #
########################################################################

CREATE SIP SERVICE GF_INT_SG
CONFIGURE SIP SERVICE GF_INT_SG SIGNALING ZONE INTERNAL
CONFIGURE SIP SERVICE GF_INT_SG MEDIA NIFGROUP NG_IN
CONFIGURE SIP SERVICE GF_INT_SG TIMER SESSIONKEEPALIVE 0
CONFIGURE SIP SERVICE GF_INT_SG OUT_ADAPTOR PROFILE NOTIFY_SIPFRAG_OUT

% SHOW SIP SERVICE GF_INT_SG ADMIN
Node: labgsx01                                         Date: 2010/12/03 18:34:31  GMT
                                                       Zone: GMTMINUS05-EASTERN-US

-------------------------------------------------------------------------
 SIP Service                          : GF_INT_SG
-------------------------------------------------------------------------
 Admin State                          : ENABLED
 Mode                                 : INSERVICE
 Action                               : DRYUP
 Dryup Timeout (min)                  : 5
 Trunk Group                          : GF_INT_TG
 Disc Treatment                       : sipDefault
 Tone Package                         : default
 Source Address Filtering             : DISABLED
 Ans Supervision Timeout              : 300
 Ans Supervision Timeout Action       : RELEASE
 Signaling Zone                       : INTERNAL
 Media Zone                           : INTERNAL
 Media NIF Group                      : NG_IN
 NAPT for Signaling                   : DISABLED
```

```
NAPT for Media                          : DISABLED
NAPT QualificationTable name            :
Parse Embedded BGID                     : DISABLED
Congestion Reject Method                : RELEASE
Congestion Retry Timer Min (sec)        : 10
Congestion Retry Timer Max (sec)        : 30
Congestion Release Timeout (sec)        : 0
SIP Timer T1 (msec)                     : 500
SIP Timer T2 (msec)                     : 4000
SIP Session Keepalive Timer (sec)       : 0
SIP Session Term Delta Time (sec)       : 0
SIP Minimum Session Timer (sec)         : 90
Retry Count for SIP Messages            : 7
Retry Count for INVITE Message          : 1
Retry Count for RE-INVITE Message       : 0
Retry Count for BYE Message             : 3
Retry Count for CANCEL Message          : 3
Session keepalive retry  on 422         : 5
Session keepalive retry  on 491         : 5
Use Route Set                           : DISABLED
OPTIONS                                 : ALLOW
REFER                                   : ALLOW
SUBSCRIBE                               : ALLOW
NOTIFY                                  : ALLOW
INFO                                    : ALLOW
REGISTER                                : ALLOW
MESSAGE                                 : ALLOW
PUBLISH                                 : ALLOW
Address Reachability Service Profile :
REGISTER redirection method             : NONE
Registration                            : NONE
Registrant CAC Profile                  :
Use CallingParty from PAI(priority1) : ENABLED
Use CallingParty from PPI(priority2) : ENABLED
Use CallingParty from RPI(priority3) : ENABLED
Use CallingParty from FROM(priority4): ENABLED
Registrar Minimum Expires (sec)         : 3600
Use CPC Param Received in               : DEFAULT
Relay ISUP MIME Body                    : DISABLED
Privacy Param Restricted                : DEFAULT
Long Duration Call Timeout (mins)       : 0
Long Duration Call Action               : NOACTION
Long Duration Call Release Cause        : 41
Long Duration Call Emergency Calls      : EXCLUDE
Resource Priority Header Profile        : defaultSipResPriorProf
Variant Type                            : SONUS
Trusted Source flag                     : ENABLED
COMEDIA connection role                 : NONE
Crank Back Profile                      : 1
Skip Crank Back Profile                 : DISABLED
DNS Support                             : A-ONLY
Receive Side Filter Profile             :
Direct Media Allowed                    : DISABLED
TCP Retransmit Interval in Seconds      : 6
SCTP Retransmit Interval in Seconds     : 6
Registration Max-Expires NON-NAT        : 3600
Registration Max-Expires NAT-TCP        : 240
Registration Max-Expires NAT-UDP        : 60
Call Redirection                        : ENABLED
Transport Protocol Preference #1        : NONE
Transport Protocol Preference #2        : NONE
Transport Protocol Preference #3        : NONE
Transport Protocol Preference #4        : NONE
Factor Value for Hop Counter            : 1
```

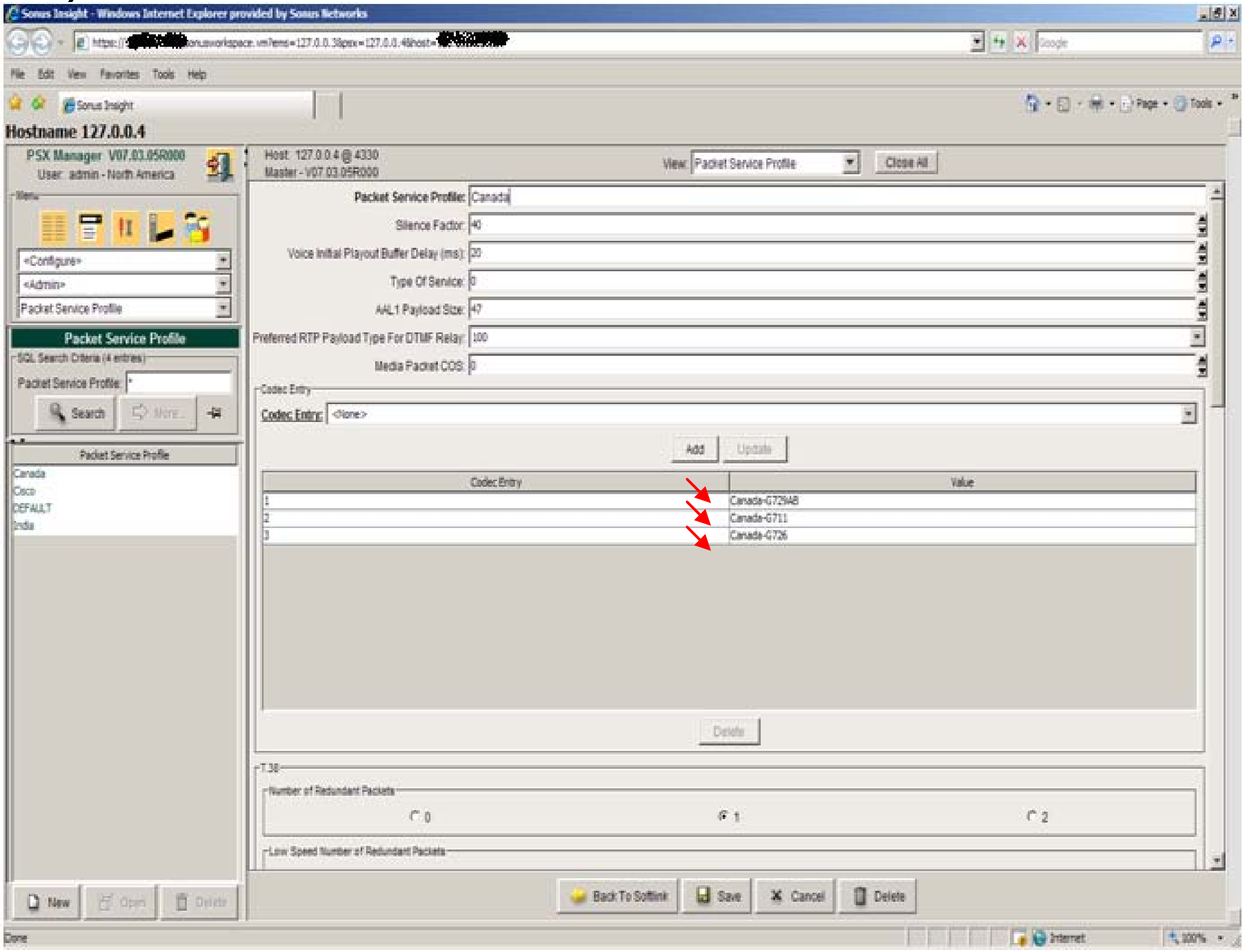
```
 Max Fwds Hdr Default                    : 70
 Route Msg Validation                    : NOVALIDATION
 Overlap Addressing Support              : DISABLED
 Overlap Min Digits For Query            : 0
 Overlap Timer Digit Collection          : 10
 Overlap Timer IOW3                      : 4
 Timer IOW2                              : 0
 Inter Operator ID                       :
 URI PRESENTATION PREFERENCE             : NONE
 Additional Headers Transmit Profile     :
 Strict Parse                            : DISABLED
 TMR Unrestricted 64kbit/s               : DISABLED
 Include Application Headers             : DISABLED
 Transmit Preconditions                  : NONE
 Receive Preconditions                   : NONE
 DataPathMode Passthru                   : DISABLED
 CPC to SIP Cause Map Profile Index      : 0
 SIP to CPC Cause Map Profile Index      : 0
 Set NOA to International                : DISABLED
 Relay Non-Invite Requests               : DISABLED
 Default MaxPtime                        : 150
 The 100Rel support                      : ENABLED
 Late Media support                      : CONVERT
 Emergency Profile                       :
 Estimated Child Registrations           : 1
 Input Adaptor Profile                   :
 Output Adaptor Profile                  : NOTIFY_SIPFRAG_OUT    #This should go
away with SONUS00106060 (see detail for SMM below)
 Bckwd Info Msg After Confirmed Dialog: DISABLED
 Use Ingress Originating CA              : DISABLED
 Add Egress Originating CA               : DISABLED
 ISDN SubAddress Preference              : RFC2806
 Peer Overload Throttling                : DISABLED
 Dynamic Blacklist Profile               :
 Send Originating CIC                    : DISABLED
 Use Ingress Charge Info                 : DISABLED
 Send Charge Info                        : DISABLED
 Media Recording                         : DISABLED
 Refer Reject Response Code              : 403
 Redirect Disconnect Code                : 503


#######################################################
# Fix in SONUS00106060 sip frag body support in Notify#
#                                                     #
#######################################################
# Delete the X-Jmac header and create VAR1            #
#######################################################
CREATE SIPADAPTOR PROFILE NOTIFY_SIPFRAG_OUT
CONFIGURE SIPADAPTOR PROFILE NOTIFY_SIPFRAG_OUT ADD RULE 1
CONFIGURE SIPADAPTOR PROFILE NOTIFY_SIPFRAG_OUT RULE 1 ADD CRITERION MESSAGE
CRITERION MESSAGE MESSAGE_TYPES REQUEST METHOD_TYPE NOTIFY CONDITION EXIST

CONFIGURE SIPADAPTOR PROFILE NOTIFY_SIPFRAG_OUT RULE 1 ADD CRITERION VARIABLE
CRITERION VARIABLE VARIABLE_ID VAR1 CONDITION ABSENT
CONFIGURE SIPADAPTOR PROFILE NOTIFY_SIPFRAG_OUT RULE 1 ADD CRITERION HEADER
CRITERION HEADER NAME "X-Jmac" CONDITION EXIST
CONFIGURE SIPADAPTOR PROFILE NOTIFY_SIPFRAG_OUT RULE 1 ADD ACTION 1 TYPE
VARIABLE
CONFIGURE SIPADAPTOR PROFILE NOTIFY_SIPFRAG_OUT RULE 1 ACTION 1 TO VARIABLE
VAR1
CONFIGURE SIPADAPTOR PROFILE NOTIFY_SIPFRAG_OUT RULE 1 ACTION 1 FROM VALUE
"message/sipfrag;version=2.0"
```

```
CONFIGURE SIPADAPTOR PROFILE NOTIFY_SIPFRAG_OUT RULE 1 ACTION 1 OPERATION
STORE
CONFIGURE SIPADAPTOR PROFILE NOTIFY_SIPFRAG_OUT RULE 1 ADD ACTION 2 TYPE
HEADER
CONFIGURE SIPADAPTOR PROFILE NOTIFY_SIPFRAG_OUT RULE 1 ACTION 2 OPERATION
DELETE
CONFIGURE SIPADAPTOR PROFILE NOTIFY_SIPFRAG_OUT RULE 1 ACTION 2 TO
HEADER_NAME "X-Jmac"
#######################################################
# Modify the Content-Type header with value of VAR1   #
#######################################################
CONFIGURE SIPADAPTOR PROFILE NOTIFY_SIPFRAG_OUT ADD RULE 2
CONFIGURE SIPADAPTOR PROFILE NOTIFY_SIPFRAG_OUT RULE 2 ADD CRITERION MESSAGE
CRITERION MESSAGE MESSAGE_TYPES REQUEST METHOD_TYPE NOTIFY CONDITION EXIST

CONFIGURE SIPADAPTOR PROFILE NOTIFY_SIPFRAG_OUT RULE 2 ADD CRITERION VARIABLE
CRITERION VARIABLE VARIABLE_ID VAR1 CONDITION EXIST
CONFIGURE SIPADAPTOR PROFILE NOTIFY_SIPFRAG_OUT RULE 2 ADD CRITERION HEADER
CRITERION HEADER NAME "Content-Type" CONDITION EXIST
CONFIGURE SIPADAPTOR PROFILE NOTIFY_SIPFRAG_OUT RULE 2 ADD ACTION 1 TYPE
HEADER
CONFIGURE SIPADAPTOR PROFILE NOTIFY_SIPFRAG_OUT RULE 2 ACTION 1 HEADER_INFO
HEADERVALUE
CONFIGURE SIPADAPTOR PROFILE NOTIFY_SIPFRAG_OUT RULE 2 ACTION 1 OPERATION
MODIFY
CONFIGURE SIPADAPTOR PROFILE NOTIFY_SIPFRAG_OUT RULE 2 ACTION 1 FROM VAR VAR1
CONFIGURE SIPADAPTOR PROFILE NOTIFY_SIPFRAG_OUT RULE 2 ACTION 1 TO
HEADER_NAME "Content-Type"
CONFIGURE SIPADAPTOR PROFILE NOTIFY_SIPFRAG_OUT STATE ENABLE
CONFIGURE SIP SERVICE GF_INT_SG OUT_ADAPTOR PROFILE NOTIFY_SIPFRAG_OUT


% SHOW SIPADAPTOR PROFILE NOTIFY_SIPFRAG_OUT ADMIN
Node: labgsx01                                 Date: 2010/12/03 18:37:19  GMT
                                               Zone: GMTMINUS05-EASTERN-US

SIP Adaptor Profile Name:            NOTIFY_SIPFRAG_OUT
SIP Adaptor Profile Index:           4
SIP Adaptor Profile State:           ENABLED

------------------------------------------------------
SIP Manipulation Rules
------------------------------------------------------
Rule Index:                          1
    Apply Match Header:              ONE
    Apply Match Header Range:
Rule Index:                          2
    Apply Match Header:              ONE
    Apply Match Header Range:

------------------------------------------------------
SIP Manipulation Criterions
------------------------------------------------------
Rule 1 MESSAGE Criterion
    Match Condition:               EXIST
    Message Types:                 REQUEST
    Method Types:                  NOTIFY
Rule 1 HEADER Criterion
    Match Condition:               EXIST
    Header Name:                   X-Jmac
    Header Value:
    Header Instance:               ALL
    Header Range:
Rule 1 VARIABLE Criterion
```

```
    Match Condition:               ABSENT
    Variable:                      VAR1
    Match Value:
Rule 2 MESSAGE Criterion
    Match Condition:               EXIST
    Message Types:                 REQUEST
    Method Types:                  NOTIFY
Rule 2 HEADER Criterion
    Match Condition:               EXIST
    Header Name:                   Content-Type
    Header Value:
    Header Instance:               ALL
    Header Range:
Rule 2 VARIABLE Criterion
    Match Condition:               EXIST
    Variable:                      VAR1
    Match Value:

------------------------------------------------------
SIP Manipulation Actions
------------------------------------------------------
Rule 1 Action 1 Type VARIABLE
    Operation:             STORE
    From Operand:          message/sipfrag;version=2.0
    To Operand:            VAR1
Rule 1 Action 2 Type HEADER
    Operation:             DELETE
    Header Position:
    Header Info:
    To Operand:            X-Jmac
Rule 2 Action 1 Type HEADER
    Operation:             MODIFY
    Header Position:
    Header Info:           HEADERVALUE
    From Operand:          VAR1
    To Operand:            Content-Type
```

3.4.5 PSX GUI tables needed for Genesys SIP Server IPTG

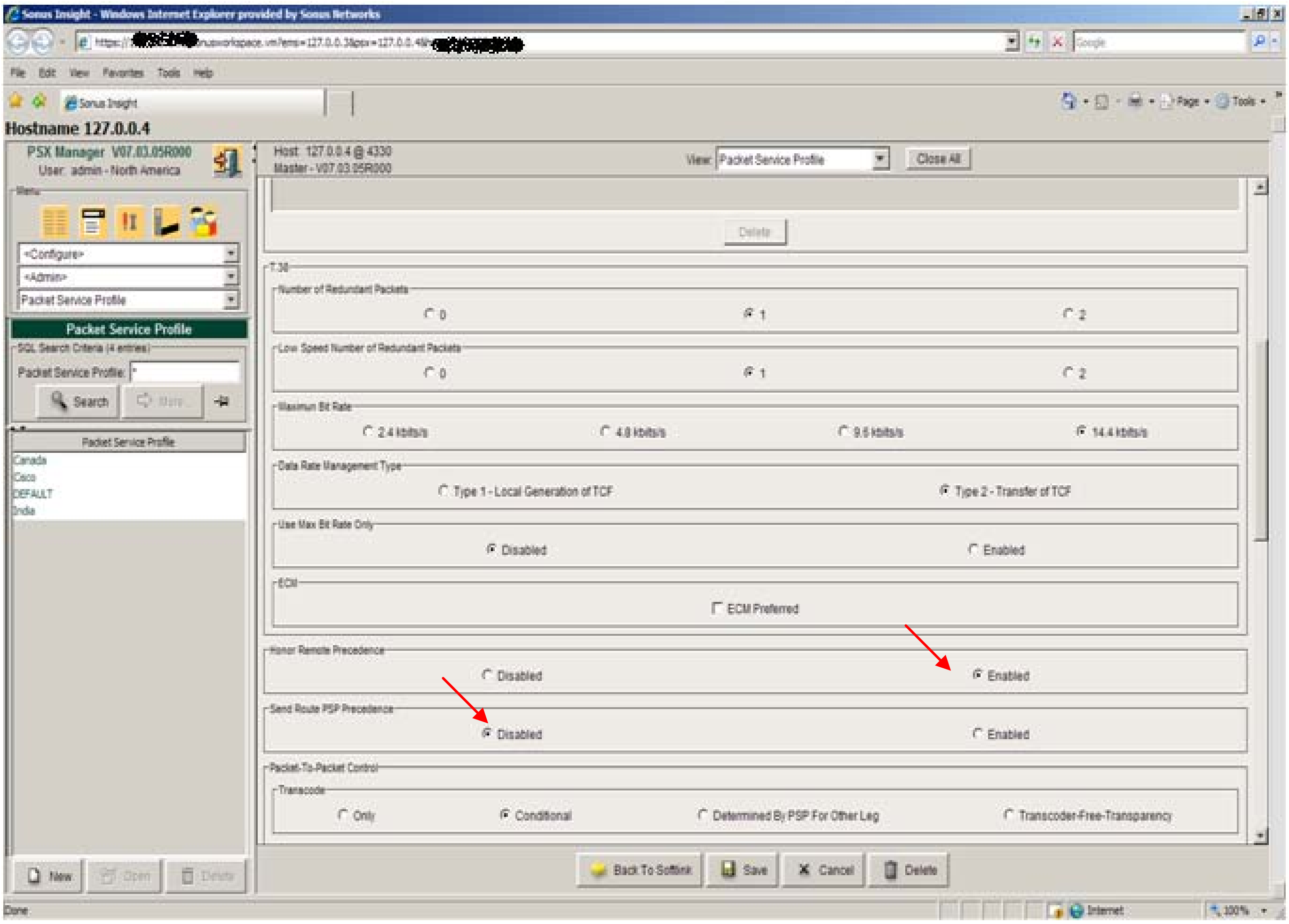
STEP 1: Create Packet Service Profile Packet Service Profile (codecs G729AB, G711 and G726)



Screenshot 1/3

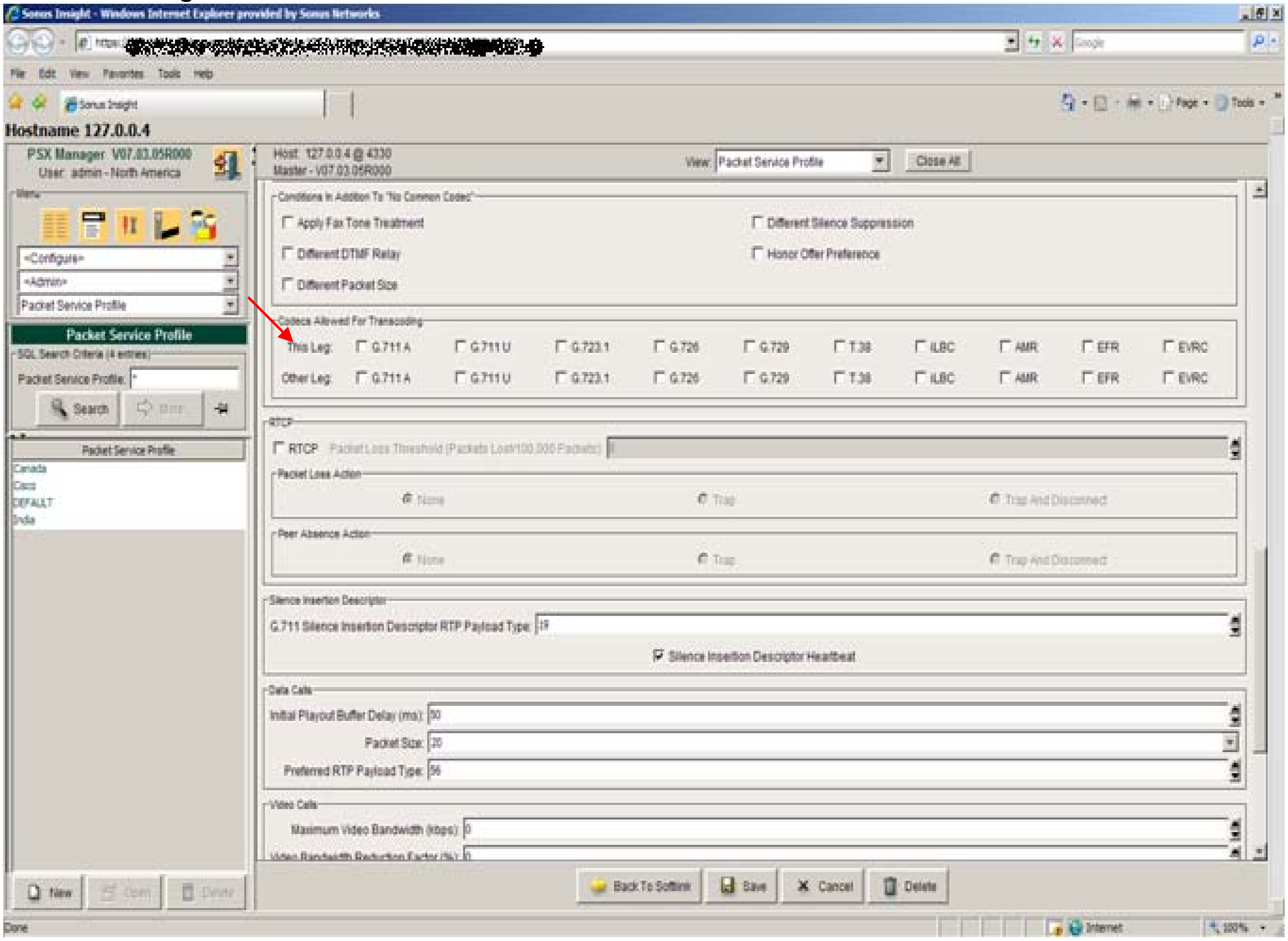
Honor Remote Precedence set to Enabled
Send Route PSP Precedence set to Disabled

- The PSP had "Send Route PSP" enabled which meant the order assigned in the PSP on the SIP Server TG "GF_INT_TG" is used when sending an INVITE out to SIP Server. By disabling this Sonus is using the order of codecs sent from AT&T.



Screenshot 2 / 3

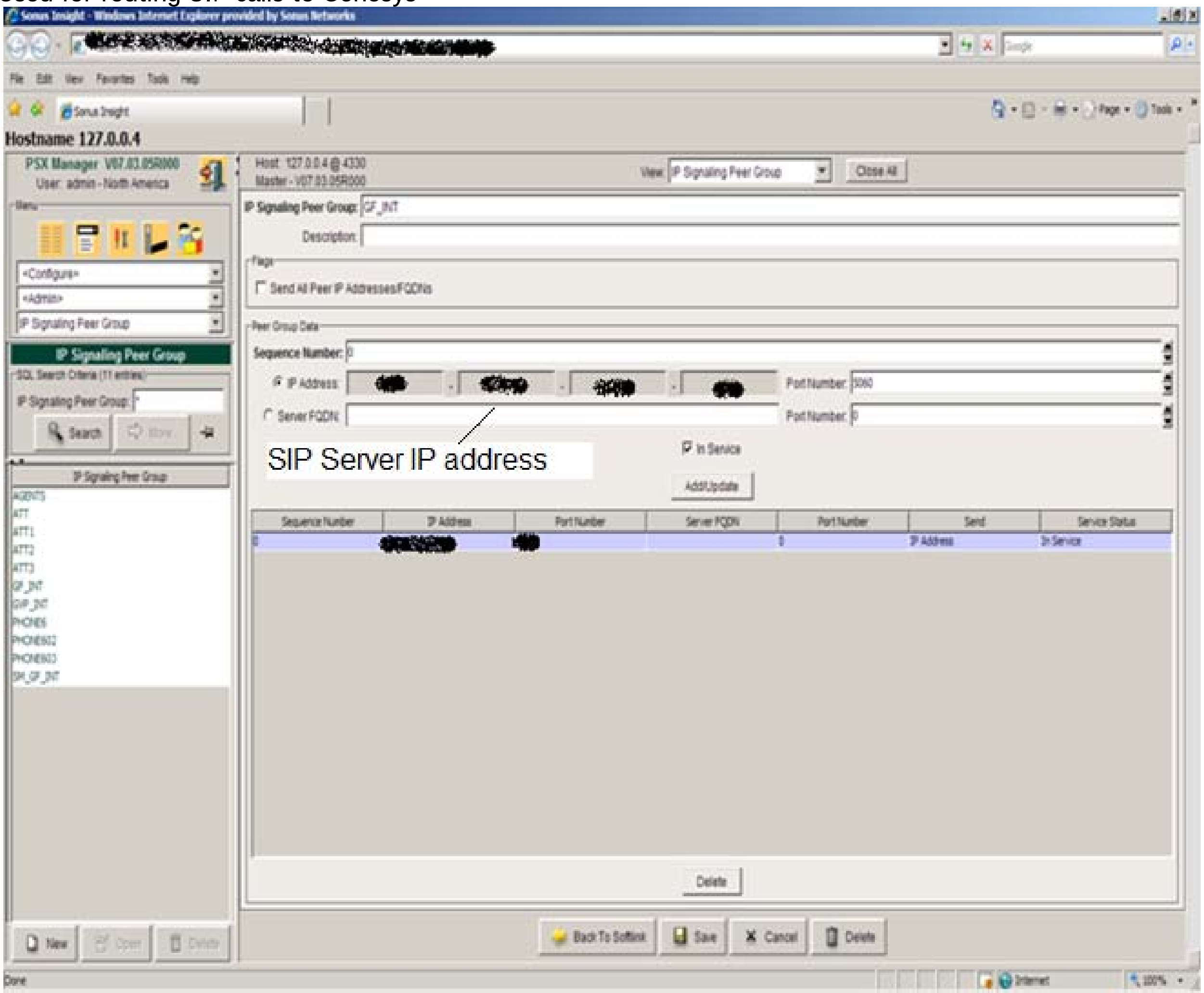
No Transcoding combos are selected



Screenshot 3 / 3

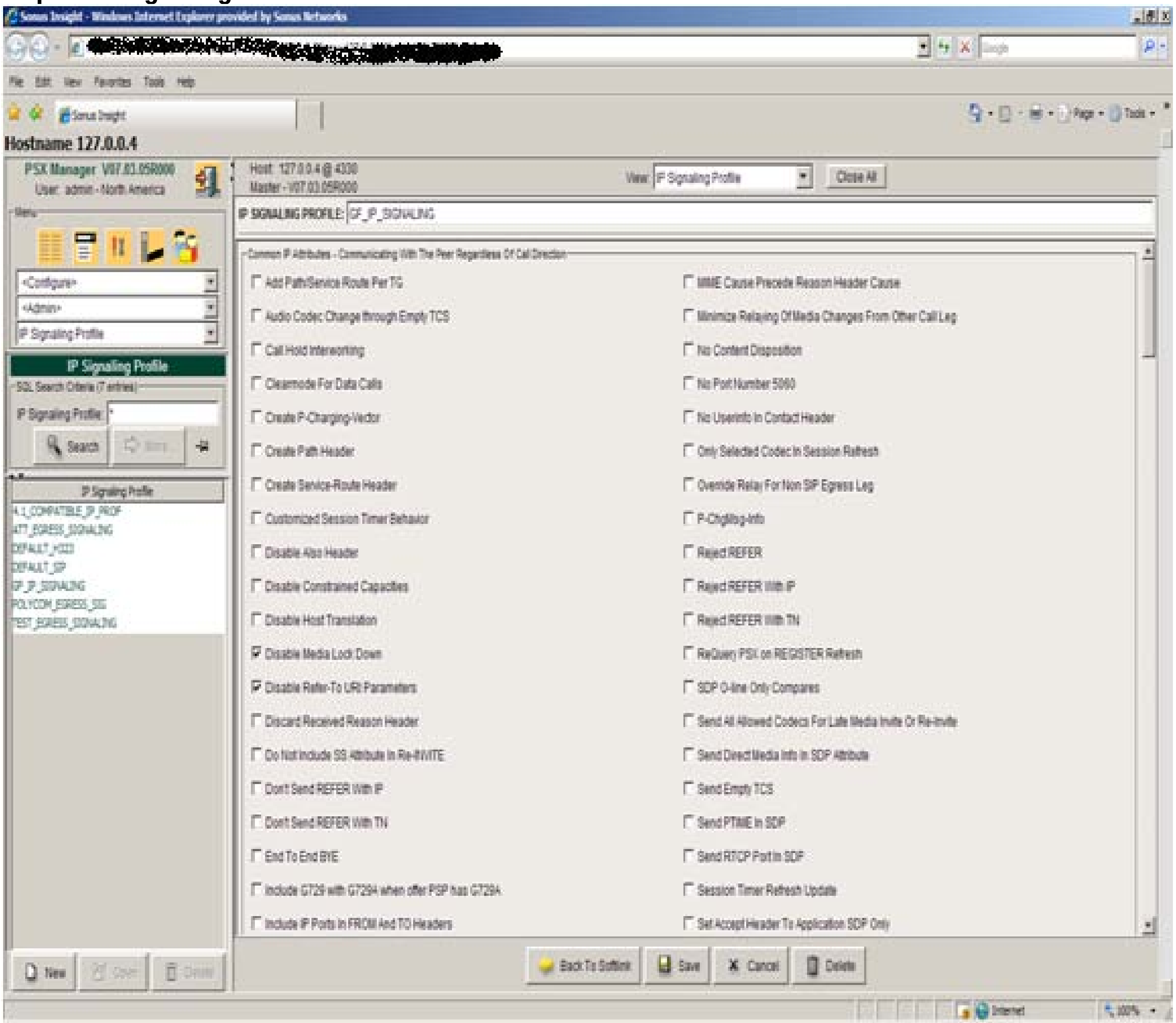
Step 2: IP Signaling Peer Group

Used for routing SIP calls to Genesys



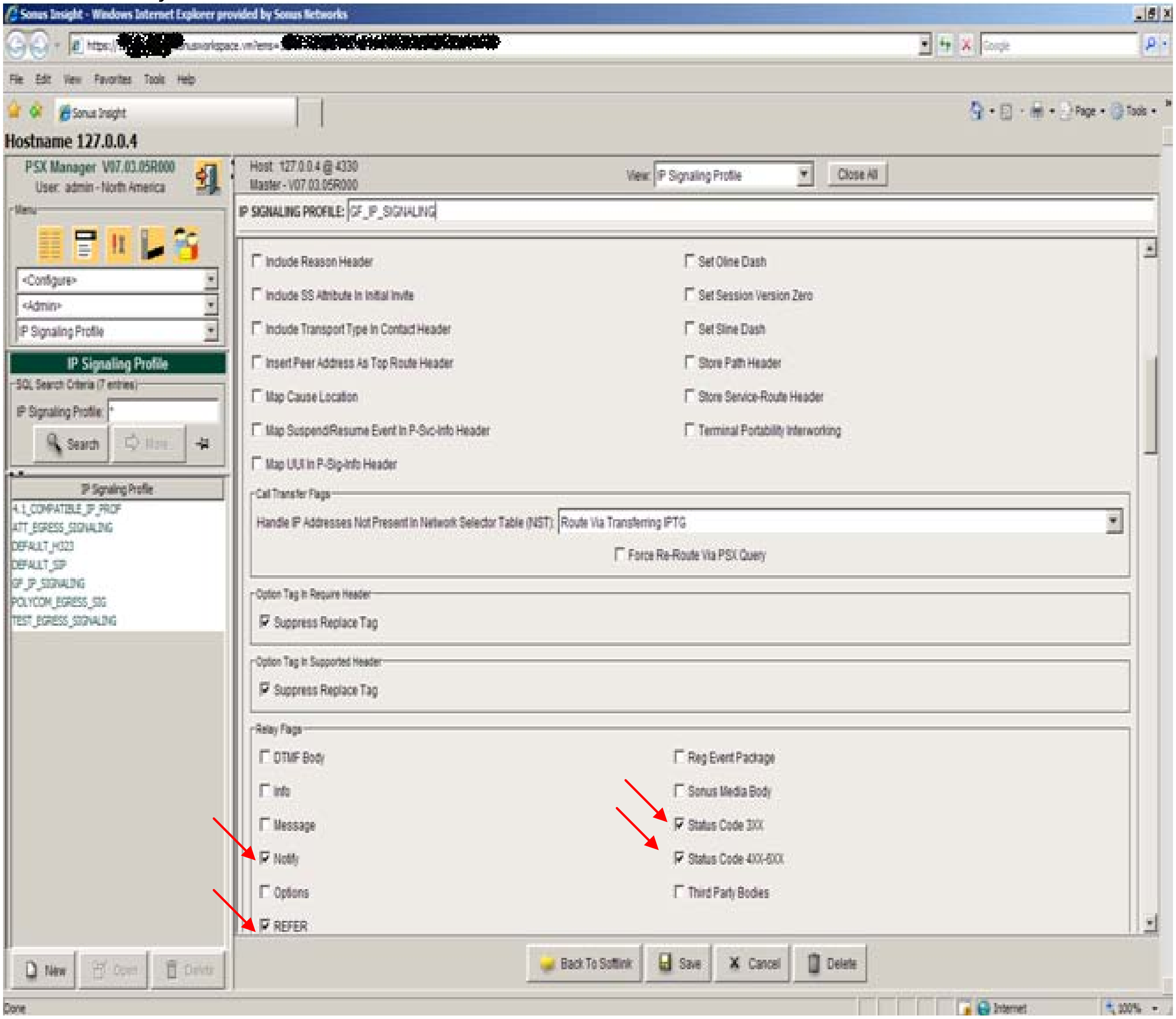
Screenshot 1 / 1

Step 3: IP Signaling Profile



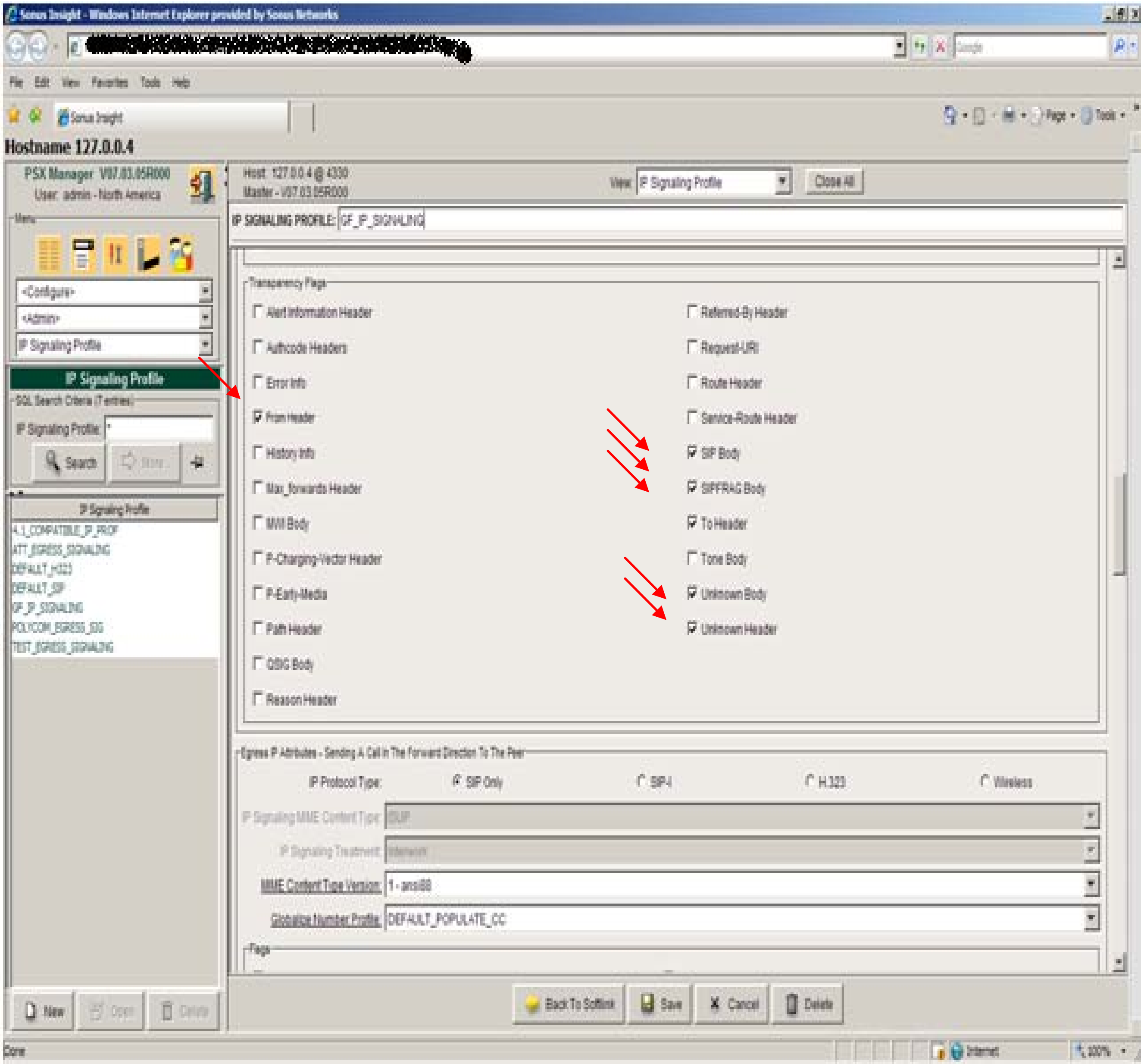
Screenshot 1 / 7

Relay flags set: Notify, REFER, 3XX and 4XX-6XX
By default NBS would process any 3xx message and route according to the Contact. This has been changed to the needed Transparent method where we Relay the message and AT&T processes the 3xx from the Genesys SIP Server now.

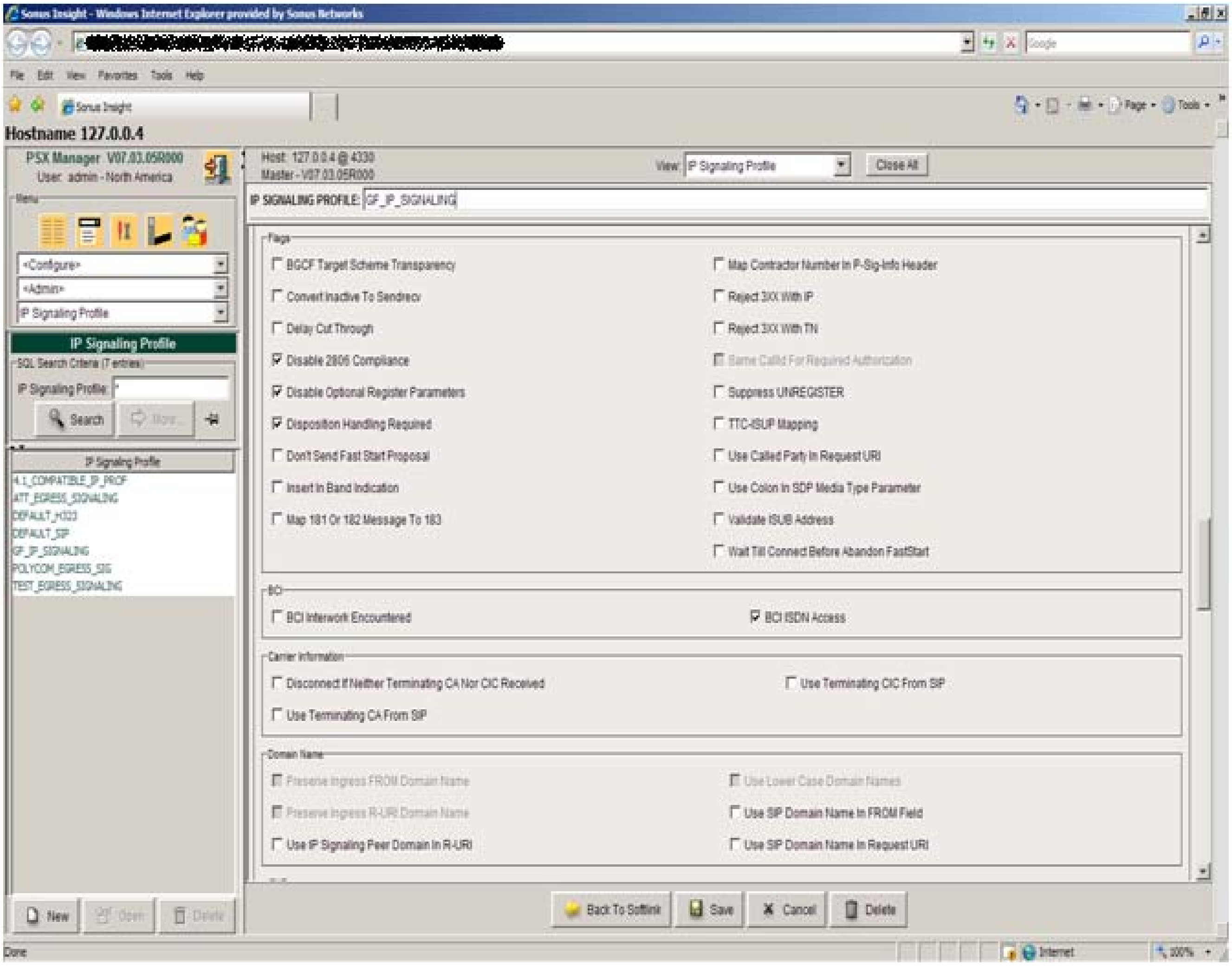


Screenshot 2 / 7

Transparancy Flags set: From Header, SIP Body, SIPFRAG Body, To Header, Unknown Body and Unknown Header.

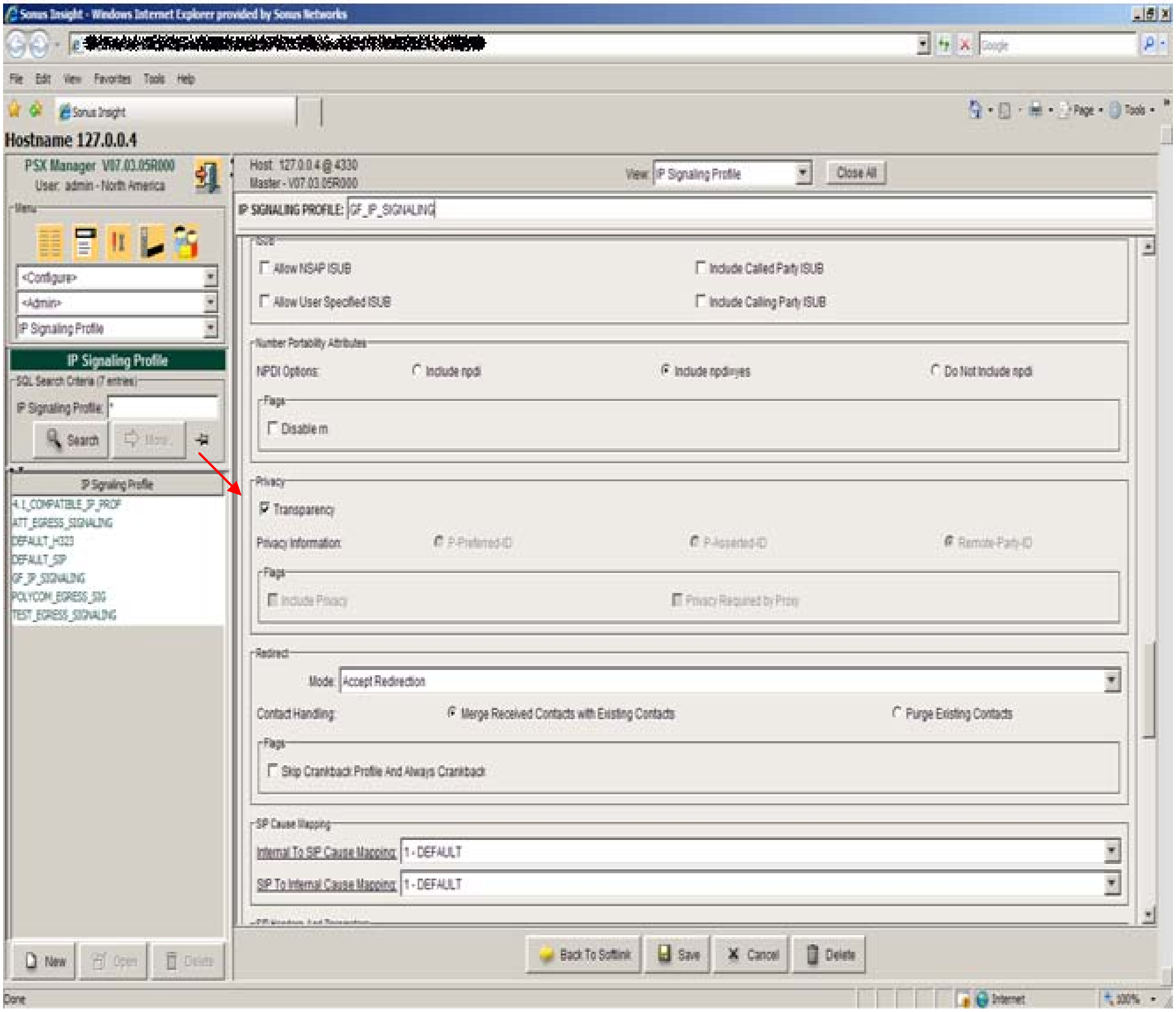


Screenshot 3 / 7

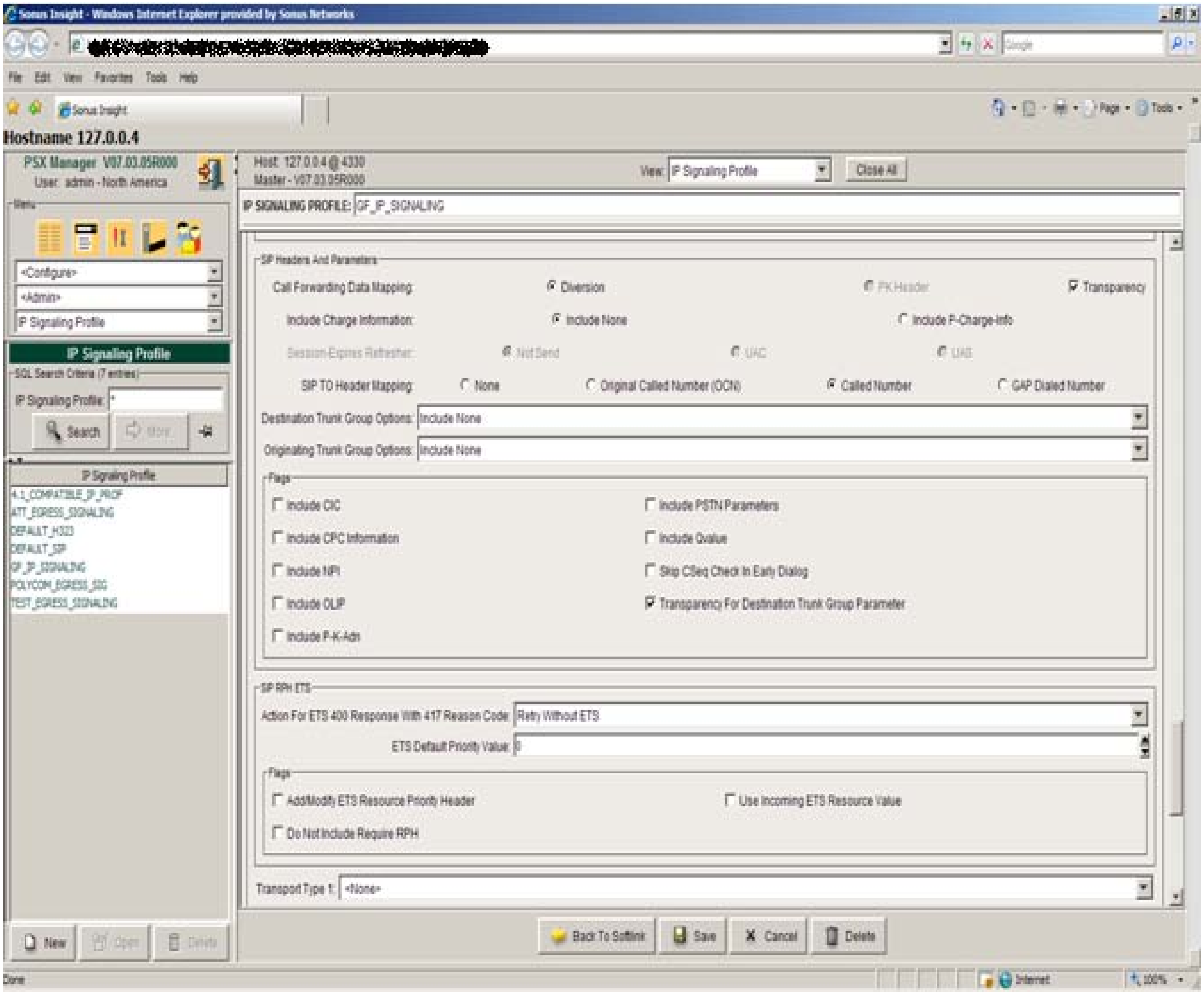


Screenshot 4 / 7

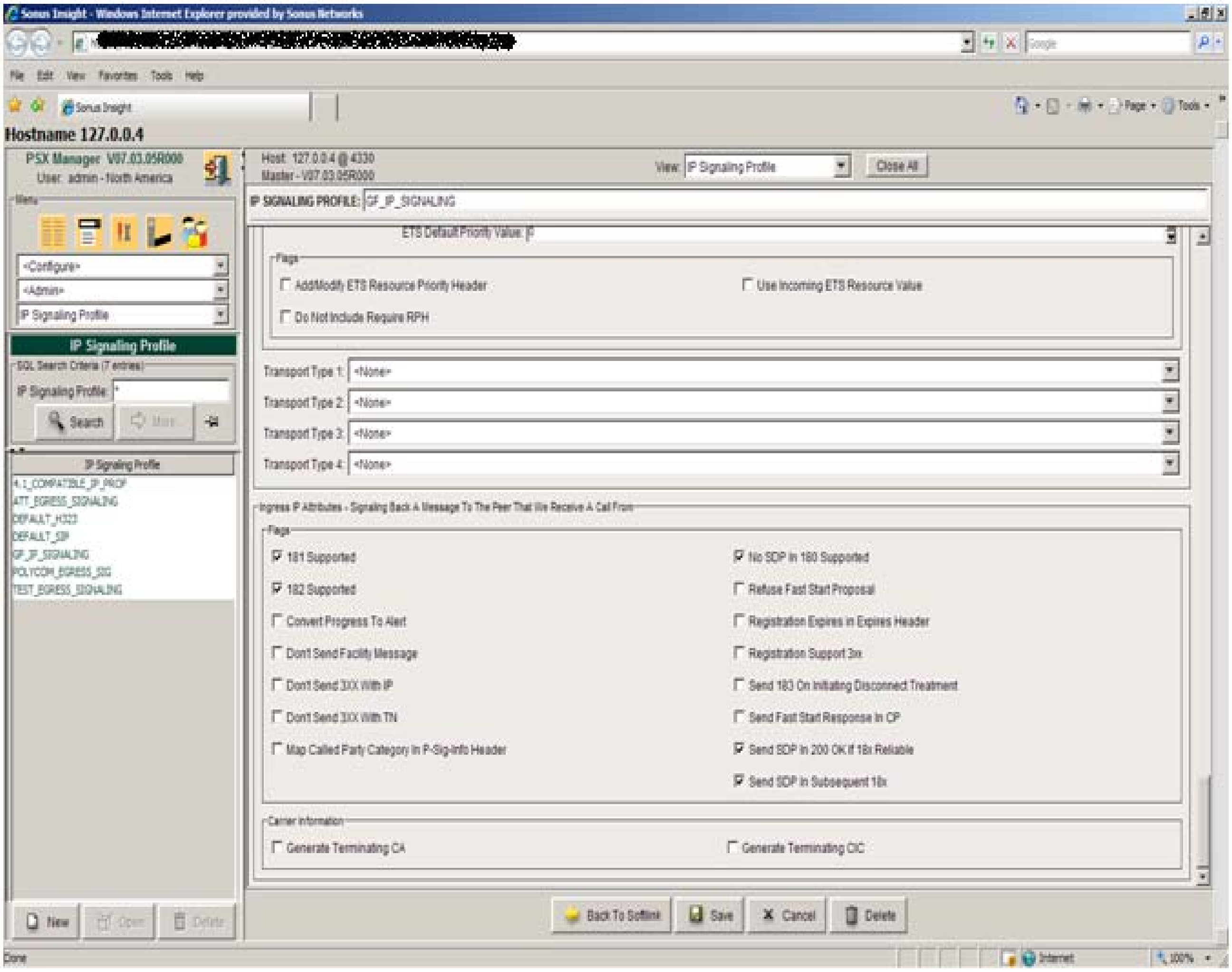
Set Privacy Transparancy flag



Screenshot 5 / 7

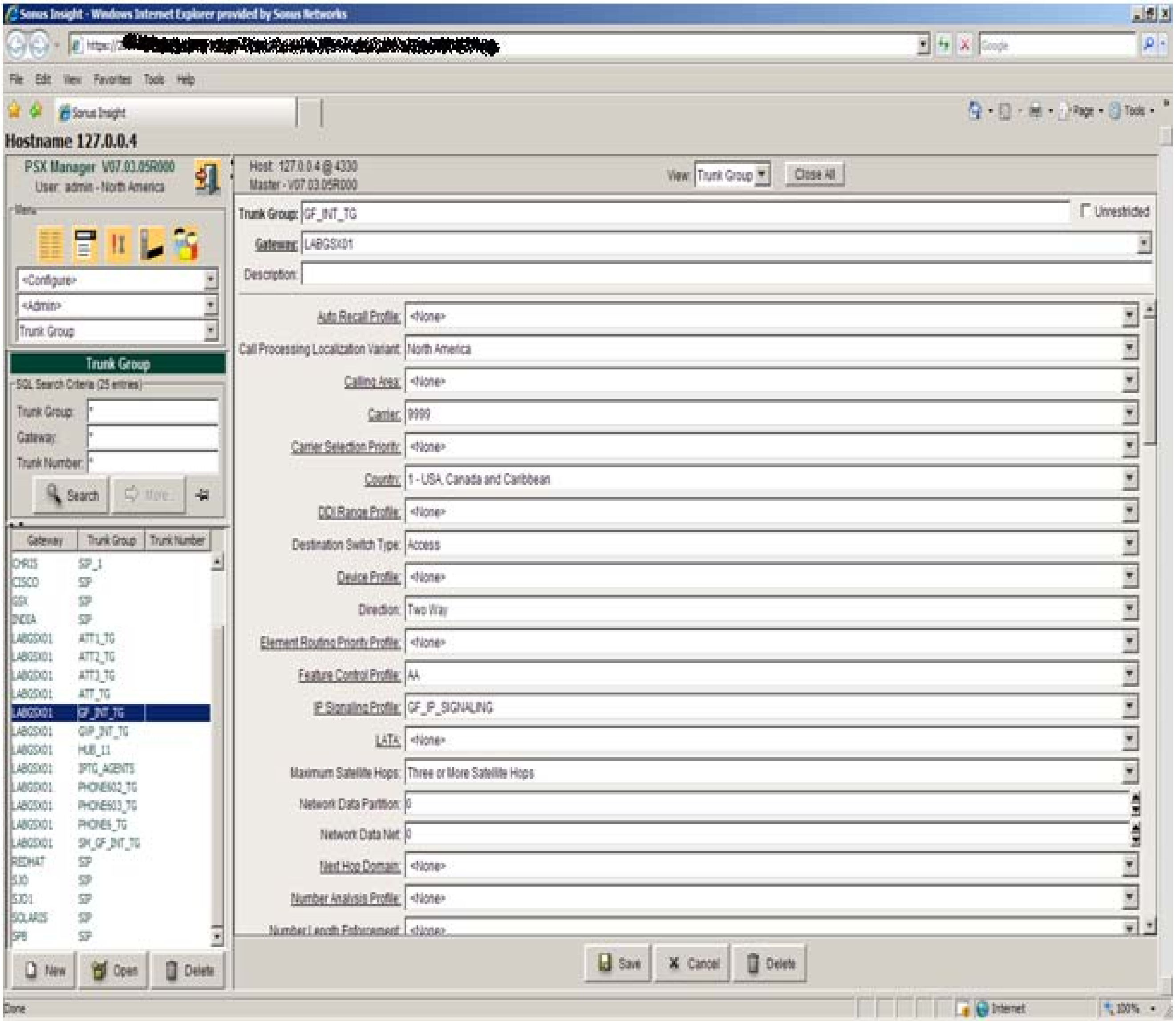


Screenshot 6 / 7

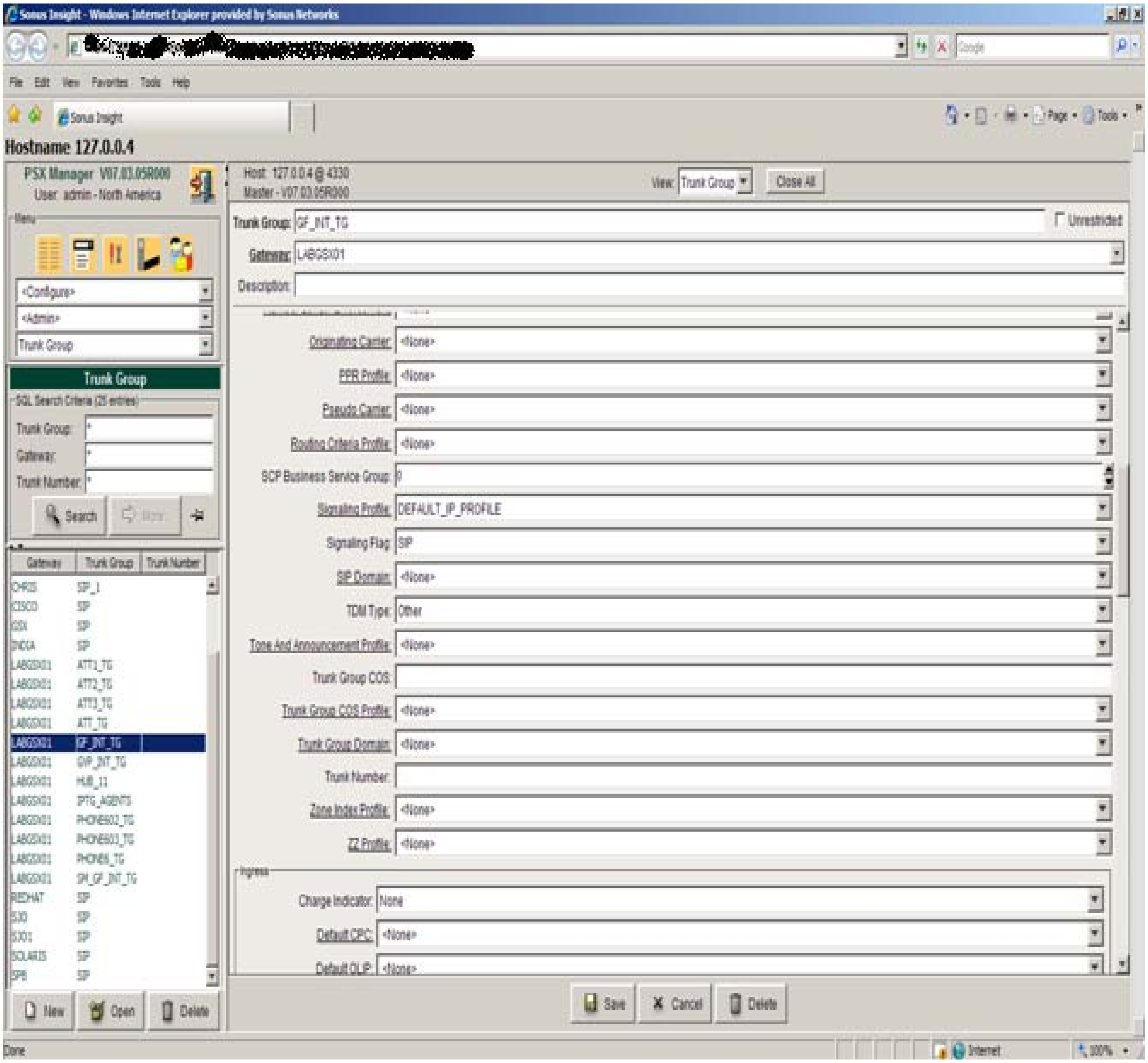


Screenshot 7 / 7

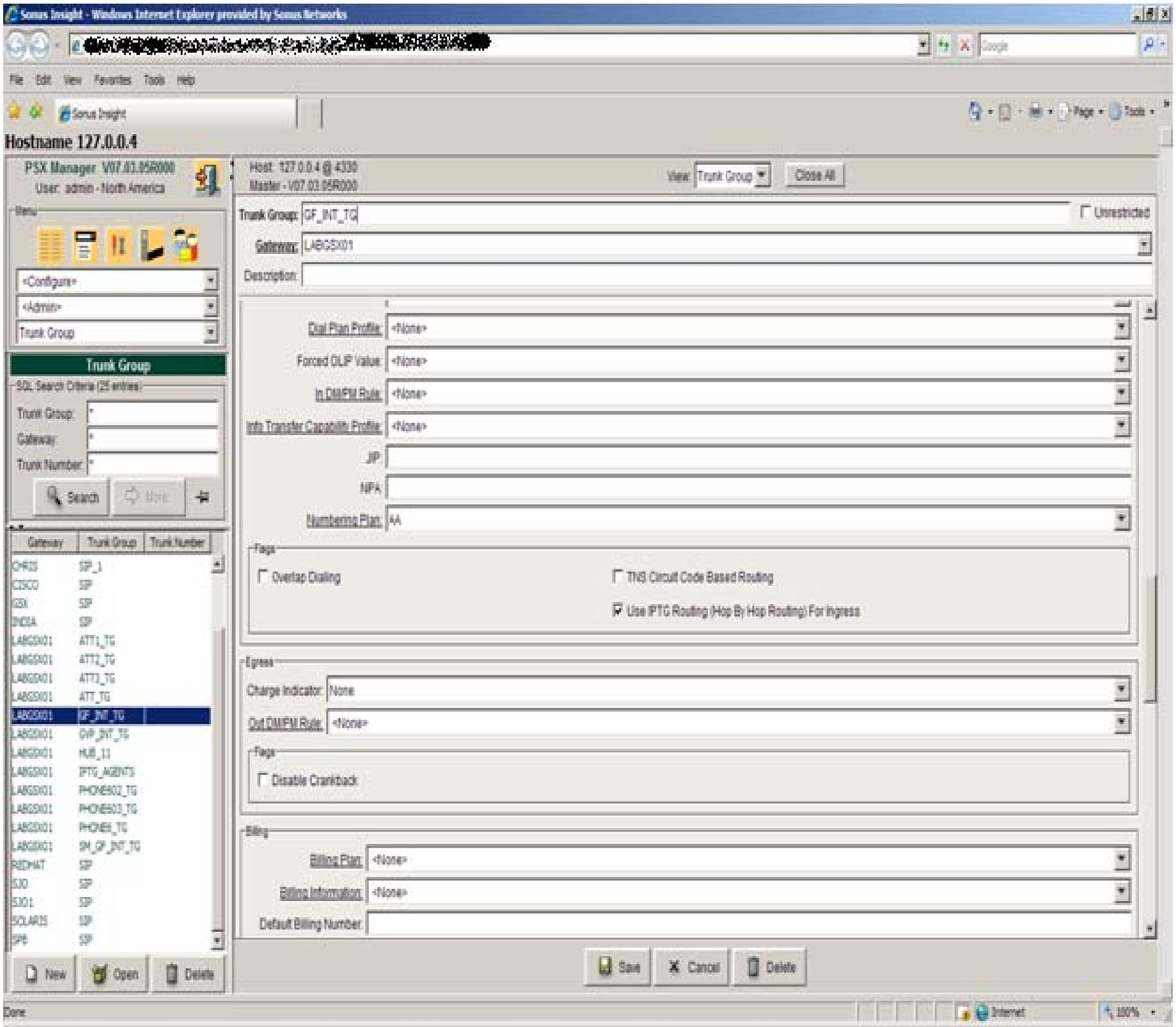
Step 4: Build Trunkgroup (assign PSP, IPSP and IP Sig Peer from Steps 1-3)



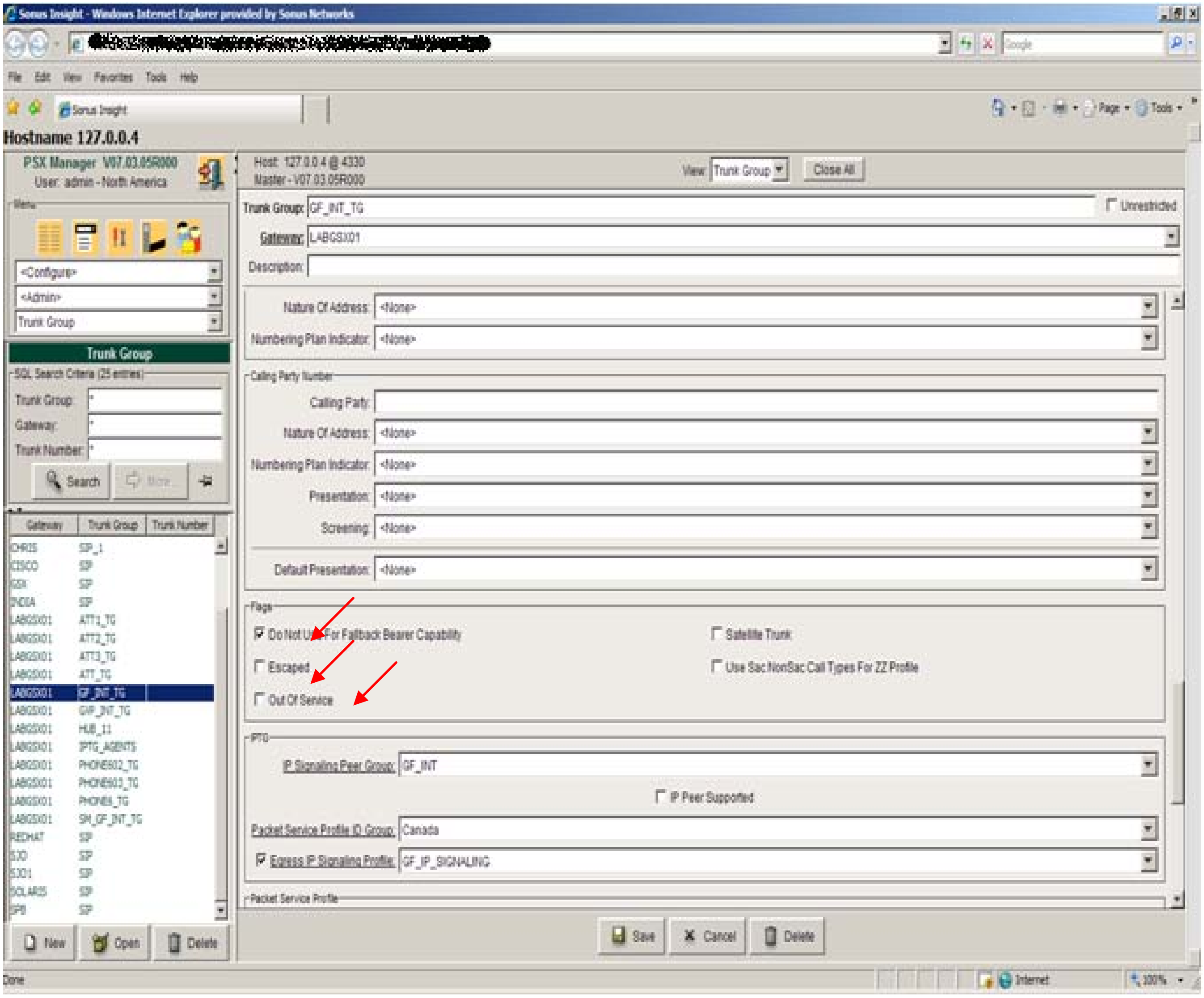
Screenshot 1/5



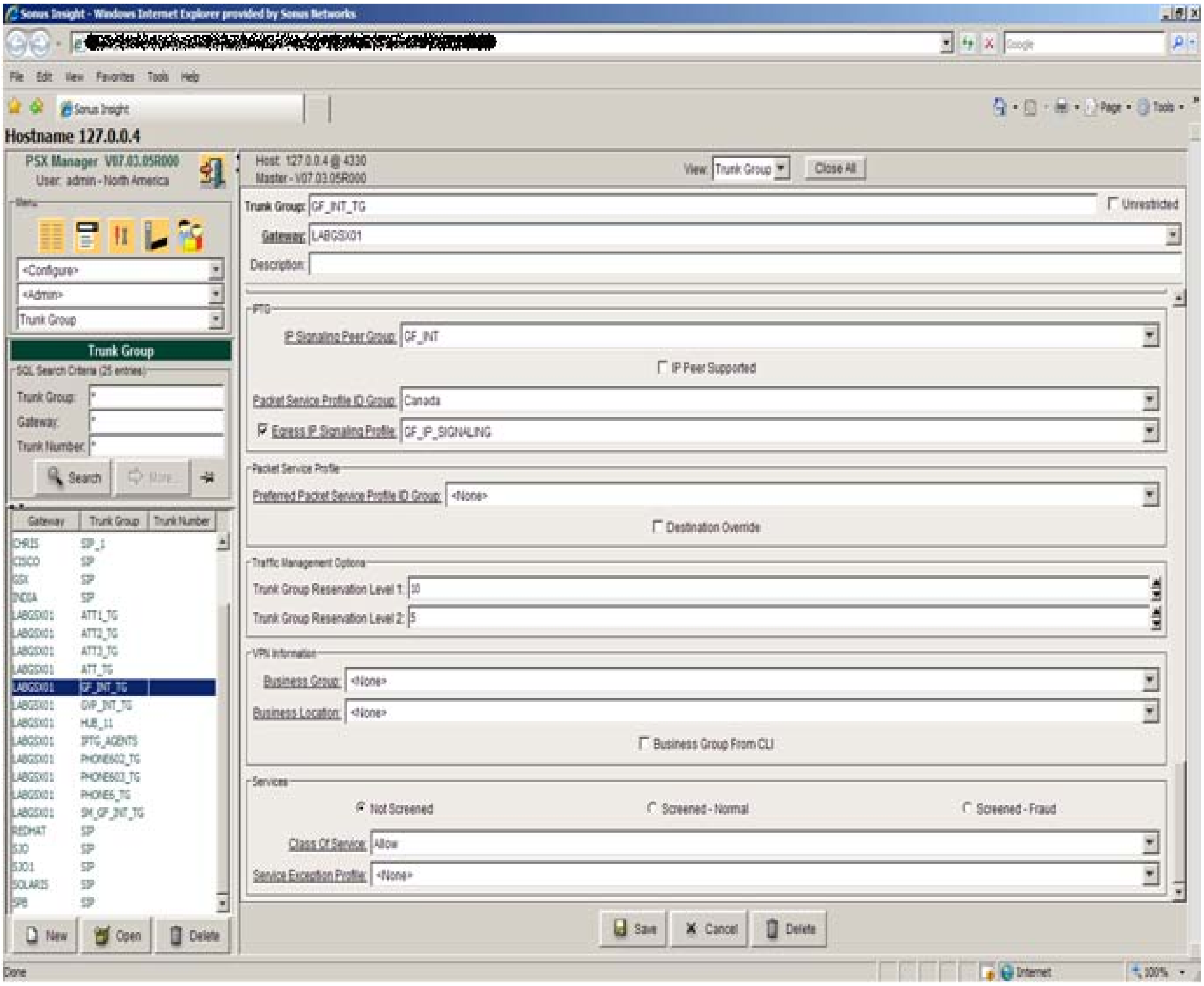
Screenshot 2 / 5



Screenshot 3/5

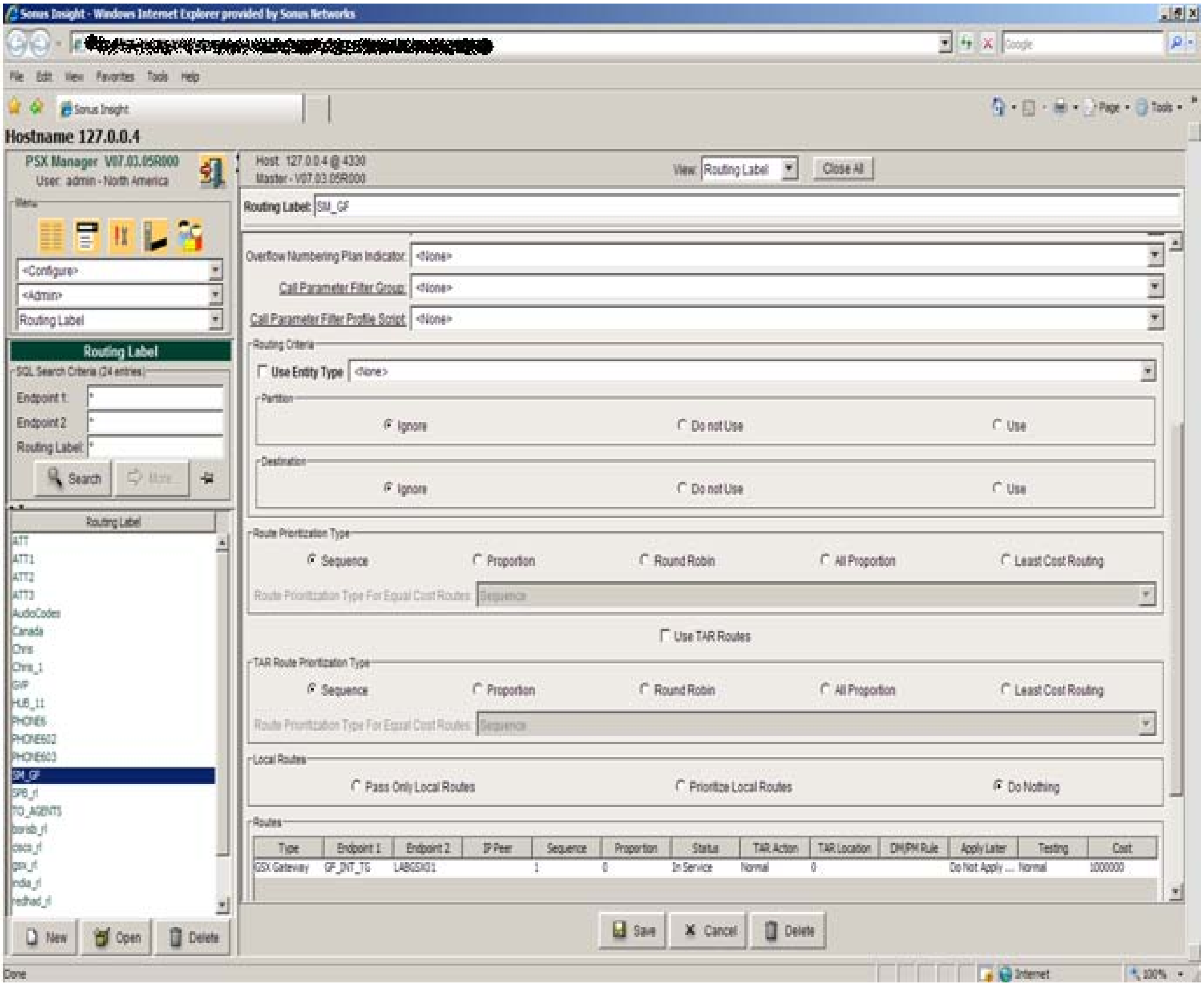


Screenshot 4 / 5



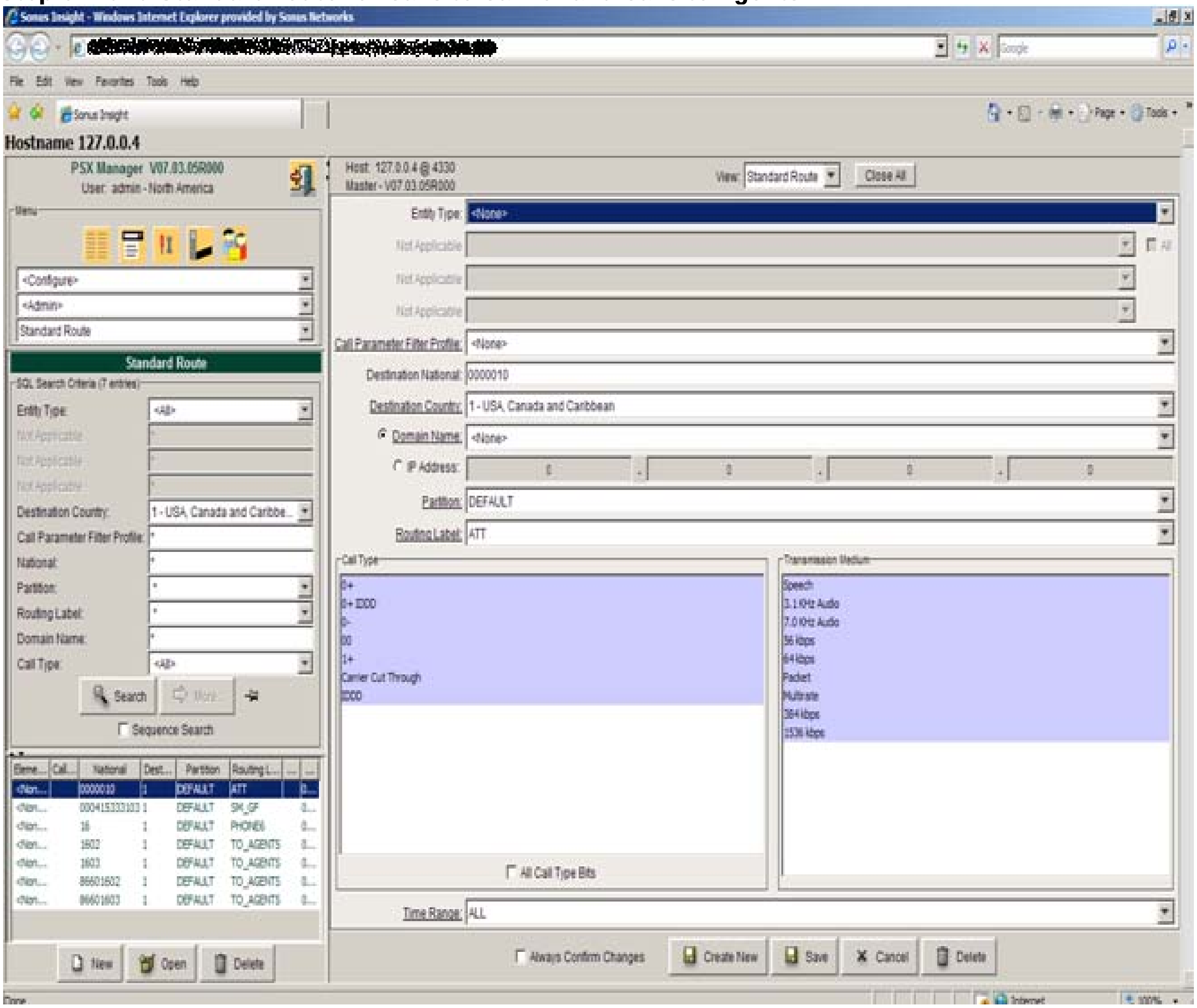
Screenshot 5 / 5

Step 5: Build Route Label to Trunkgroup



Screenshot 1 / 1

Step 6: Build Standard route for Calls to Carrier and Calls to Agents



Screenshot 1 / 1

3.4.6 GSX CLI commands for creating AT&T SIP Server IPTG

At a high level, following configuration is performed on the GSX to connect Sonus to AT&T.

1. Create Network Selector Table for ingress TG determination.
2. Create TG AT&T_TG (same name in PSX TG) - Config with Network selector Table AT&T.
3. Create SIP Service under TG AT&T_TG - Config Sig Zone, Nif Group, Session Timer and Out Adaptor.
4. Fix in SONUS00106060 sip frag body support in Notify to support NOTIFY sent by AT&T.

Note:- AT&T inbound calls use a separate IP Trunkgroup AT&T_TG "external" IP interface. Sonus is configured to route traffic from "external" interface to Sonus "Internal" interface IPTG INT_GF_TG. The NBS "internal/trusted interface" is configured to communicate with Genesys SIP Server

```
##############################################################
# Create Network Selector Table for ingress TG determination #
##############################################################


CREATE IP NETWORK SELECTOR TABLE AT&T
CONFIGURE IP NETWORK SELECTOR TABLE AT&T ADD NUMBER 2XX.2XX.2XX.0 MASK
255.255.255.0
% SHOW IP NETWORK SELECTOR TABLE AT&T ADMIN
Node: labgsx01                               Date: 2010/12/06 13:57:29  GMT
                                             Zone: GMTMINUS05-EASTERN-US

Table Name                      Network Number    Network Mask
-----------------------------------------------------------------------
AT&T                             2XX.2XX.2XX.0   255.255.255.0


###############################################################
#         Create TG AT&T_TG (same name in PSX TG)             #
#         Config with Network selector Table AT&T             #
###############################################################

CREATE TRUNK GROUP AT&T_TG
CONFIGURE TRUNK GROUP AT&T_TG NETWORK SELECTOR TABLE AT&T

% SHOW TRUNK GROUP AT&T_TG ADMIN
Node: labgsx01                               Date: 2010/12/06 13:59:40  GMT
                                             Zone: GMTMINUS05-EASTERN-US


Local Trunk Name: AT&T_TG
----------------------------------------------------------------------------
 State                                            ENABLED
 Inbound Reserve (percent)                        0
 Mode                                             INSERVICE
 Action                                           DRYUP
 Timeout (min)                                    5
 Circuit Reservation State                        DISABLED
 Reserved Priority Calls (circuits)               1
 Reserved Incoming Calls (circuits)               1
 Reserved Outgoing Calls (percent)                10
 Alternate Trunk Group Name
 Trunk Group Rename Timer (sec)                   10
 SILC State                                       DISABLED
 SILC Congestion Level 1 Calls Allowed (percent) 075
 SILC Congestion Level 2 Calls Allowed (percent) 025
 Trunk Group Type                                 IPSELECTED
```

```
 IP Trunk Group Direction                           BOTHWAYS
 Parent IP Trunk Group
 IP Network Selection Table                         AT&T
 IP Call Limit                                      UNLMT
 IP Bandwidth Limit                                 UNLMT
 Packet Outage Detection Minimum Duration           6000
 Packet Outage Detection Minimum Calls              1000
 Packet Outage Detection Bandwidth Limit Reduct     50
 Packet Outage Detection State                      DISABLED
 Packet Outage Detection Interval (minutes)         15
 Master Trunk Group Name
 Calls Requested Per MTRG Request                   100
 Bandwidth Requested Per MTRG Request (1K bps)      12400
 Maximum Ingress Sustained Call Rate                0
 Maximum Ingress Call Burst Size                    0
 Maximum Ingress Sustained SIP nonInvite Rate       0
 Maximum Ingress SIP nonInvite Burst Size           0
 Maximum Egress Sustained Call Rate                 0
 Maximum Egress Call Burst Size                     0
 Maximum Egress Sustained SIP nonInvite Rate        0
 Maximum Egress SIP nonInvite Burst Size            0
 Ingress NonPriority Call Threshold                 0
 Egress NonPriority Call Threshold                  0
 HPC Profile Name                                   defaultintipqueuing
 HPC Early ACM or SIP-18X                           USEDEFAULT
 HPC IP Oversubscription 0verride                   DISABLED
 HPC IP Oversubscription Factor                     10
 Emergency IP Oversubscription Factor               10
 Local Policy Trunk Profile
 IP Registration Limit                              UNLMT
 IP Estimated Child Registrations                   1


########################################################################
#         Create SIP Service under TG AT&T_TG                           #
#         Config Sig Zone, Nif Group, Session Timer and Out Adaptor    #
########################################################################

CREATE SIP SERVICE AT&T_SG
CONFIGURE SIP SERVICE AT&T_SG SIGNALING ZONE SZ_OUTSIDE
CONFIGURE SIP SERVICE AT&T_SG MEDIA NIFGROUP NG_OUT
CONFIGURE SIP SERVICE AT&T_SG TIMER SESSIONKEEPALIVE 0
CONFIGURE SIP SERVICE AT&T_SG OUT_ADAPTOR PROFILE NOTIFY_SIPFRAG_IN

% SHOW SIP SERVICE AT&T_SG ADMIN
Node: labgsx01                                    Date: 2010/12/06 14:04:29  GMT
                                                  Zone: GMTMINUS05-EASTERN-US

-------------------------------------------------------------------------
 SIP Service                           : AT&T_SG
-------------------------------------------------------------------------
 Admin State                           : ENABLED
 Mode                                  : INSERVICE
 Action                                : DRYUP
 Dryup Timeout (min)                   : 5
 Trunk Group                           : AT&T_TG
 Disc Treatment                        : sipDefault
 Tone Package                          : default
 Source Address Filtering              : DISABLED
 Ans Supervision Timeout               : 300
 Ans Supervision Timeout Action        : RELEASE
 Signaling Zone                        : SZ_OUTSIDE  # untrusted signaling
 Media Zone                            : INTERNAL
 Media NIF Group                       : NG_OUT      # untrusted media
```

```
NAPT for Signaling                        : DISABLED
NAPT for Media                            : DISABLED
NAPT QualificationTable name              :
Parse Embedded BGID                       : DISABLED
Congestion Reject Method                  : RELEASE
Congestion Retry Timer Min (sec)          : 10
Congestion Retry Timer Max (sec)          : 30
Congestion Release Timeout (sec)          : 0
SIP Timer T1 (msec)                       : 500
SIP Timer T2 (msec)                       : 4000
SIP Session Keepalive Timer (sec)         : 0
SIP Session Term Delta Time (sec)         : 0
SIP Minimum Session Timer (sec)           : 90
Retry Count for SIP Messages              : 7
Retry Count for INVITE Message            : 6
Retry Count for RE-INVITE Message         : 0
Retry Count for BYE Message               : 3
Retry Count for CANCEL Message            : 3
Session keepalive retry  on 422           : 5
Session keepalive retry  on 491           : 5
Use Route Set                             : DISABLED
OPTIONS                                   : ALLOW
REFER                                     : ALLOW
SUBSCRIBE                                 : ALLOW
NOTIFY                                    : ALLOW
INFO                                      : ALLOW
REGISTER                                  : ALLOW
MESSAGE                                   : ALLOW
PUBLISH                                   : ALLOW
Address Reachability Service Profile      :
REGISTER redirection method               : NONE
Registration                              : NONE
Registrant CAC Profile                    :
Use CallingParty from PAI(priority1)      : ENABLED
Use CallingParty from PPI(priority2)      : ENABLED
Use CallingParty from RPI(priority3)      : ENABLED
Use CallingParty from FROM(priority4)     : ENABLED
Registrar Minimum Expires (sec)           : 3600
Use CPC Param Received in                 : DEFAULT
Relay ISUP MIME Body                      : DISABLED
Privacy Param Restricted                  : DEFAULT
Long Duration Call Timeout (mins)         : 0
Long Duration Call Action                 : NOACTION
Long Duration Call Release Cause          : 41
Long Duration Call Emergency Calls        : EXCLUDE
Resource Priority Header Profile          : defaultSipResPriorProf
Variant Type                              : SONUS
Trusted Source flag                       : ENABLED
COMEDIA connection role                   : NONE
Crank Back Profile                        : 1
Skip Crank Back Profile                   : DISABLED
DNS Support                               : A-ONLY
Receive Side Filter Profile               :
Direct Media Allowed                      : DISABLED
TCP Retransmit Interval in Seconds        : 6
SCTP Retransmit Interval in Seconds       : 6
Registration Max-Expires NON-NAT          : 3600
Registration Max-Expires NAT-TCP          : 240
Registration Max-Expires NAT-UDP          : 60
Call Redirection                          : ENABLED
Transport Protocol Preference #1          : NONE
Transport Protocol Preference #2          : NONE
Transport Protocol Preference #3          : NONE
Transport Protocol Preference #4          : NONE
```

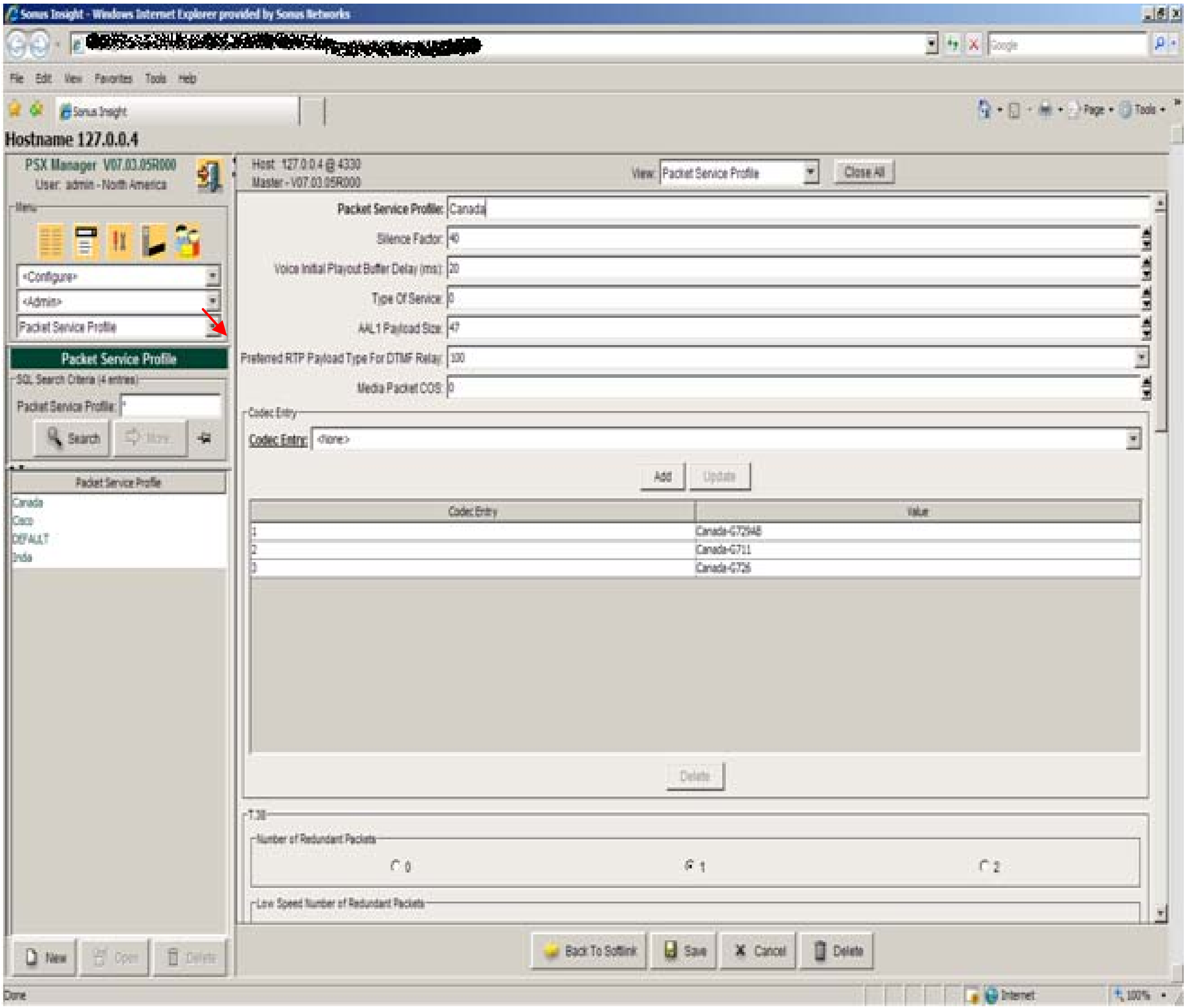
```
 Factor Value for Hop Counter          : 1
 Max Fwds Hdr Default                  : 70
 Route Msg Validation                  : NOVALIDATION
 Overlap Addressing Support            : DISABLED
 Overlap Min Digits For Query          : 0
 Overlap Timer Digit Collection        : 10
 Overlap Timer IOW3                    : 4
 Timer IOW2                            : 0
 Inter Operator ID                     :
 URI PRESENTATION PREFERENCE           : NONE
 Additional Headers Transmit Profile   :
 Strict Parse                          : DISABLED
 TMR Unrestricted 64kbit/s             : DISABLED
 Include Application Headers           : DISABLED
 Transmit Preconditions                : NONE
 Receive Preconditions                 : NONE
 DataPathMode Passthru                 : DISABLED
 CPC to SIP Cause Map Profile Index    : 0
 SIP to CPC Cause Map Profile Index    : 0
 Set NOA to International              : DISABLED
 Relay Non-Invite Requests             : DISABLED
 Default MaxPtime                      : 150
 The 100Rel support                    : ENABLED
 Late Media support                    : CONVERT
 Emergency Profile                     :
 Estimated Child Registrations         : 1
 Input Adaptor Profile                 : NOTIFY_SIPFRAG_IN    #This should go
away with SONUS00106060 (see detail for SMM below)
 Output Adaptor Profile                :
 Bckwd Info Msg After Confirmed Dialog: DISABLED
 Use Ingress Originating CA            : DISABLED
 Add Egress Originating CA             : DISABLED
 ISDN SubAddress Preference            : RFC2806
 Peer Overload Throttling              : DISABLED
 Dynamic Blacklist Profile             :
 Send Originating CIC                  : DISABLED
 Use Ingress Charge Info               : DISABLED
 Send Charge Info                      : DISABLED
 Media Recording                       : DISABLED
 Refer Reject Response Code            : 403
 Redirect Disconnect Code              : 503



#########################################################################
#         Fix in SONUS00106060 sip frag body support in Notify          #
#                                                                       #
#########################################################################
# Ingress Rule                                                          #
# Apply this to your ingress.  In short it says:                        #
#  If this is a SIP NOTIFY message, and                                 #
#  If variable VAR1 does not exist, and                                 #
#  If the Content-Type header exists, then                              #
#  Change set VAR1 to 'message/unknown', then                           #
#  set the value of the 'Content-Type' header to VAR1,                  #
#  and add a 'X-Jmac' header                                            #
#                                                                       #
#########################################################################
CREATE SIPADAPTOR PROFILE NOTIFY_SIPFRAG_IN
CONFIGURE SIPADAPTOR PROFILE NOTIFY_SIPFRAG_IN ADD RULE 1
CONFIGURE SIPADAPTOR PROFILE NOTIFY_SIPFRAG_IN RULE 1 ADD CRITERION MESSAGE
CRITERION MESSAGE MESSAGE_TYPES REQUEST METHOD_TYPE NOTIFY CONDITION EXIST
```

```
CONFIGURE SIPADAPTOR PROFILE NOTIFY_SIPFRAG_IN RULE 1 ADD CRITERION VARIABLE
CRITERION VARIABLE VARIABLE_ID VAR1 CONDITION ABSENT
CONFIGURE SIPADAPTOR PROFILE NOTIFY_SIPFRAG_IN RULE 1 ADD CRITERION HEADER
CRITERION HEADER NAME "Content-Type" CONDITION EXIST
CONFIGURE SIPADAPTOR PROFILE NOTIFY_SIPFRAG_IN RULE 1 ADD ACTION 1 TYPE
VARIABLE
CONFIGURE SIPADAPTOR PROFILE NOTIFY_SIPFRAG_IN RULE 1 ACTION 1 TO VARIABLE
VAR1
CONFIGURE SIPADAPTOR PROFILE NOTIFY_SIPFRAG_IN RULE 1 ACTION 1 FROM VALUE
"message/unknown"
CONFIGURE SIPADAPTOR PROFILE NOTIFY_SIPFRAG_IN RULE 1 ACTION 1 OPERATION
STORE
CONFIGURE SIPADAPTOR PROFILE NOTIFY_SIPFRAG_IN RULE 1 ADD ACTION 2 TYPE
HEADER
CONFIGURE SIPADAPTOR PROFILE NOTIFY_SIPFRAG_IN RULE 1 ACTION 2 HEADER_INFO
HEADERVALUE
CONFIGURE SIPADAPTOR PROFILE NOTIFY_SIPFRAG_IN RULE 1 ACTION 2 OPERATION
MODIFY
CONFIGURE SIPADAPTOR PROFILE NOTIFY_SIPFRAG_IN RULE 1 ACTION 2 FROM VAR VAR1
CONFIGURE SIPADAPTOR PROFILE NOTIFY_SIPFRAG_IN RULE 1 ACTION 2 TO HEADER_NAME
"Content-Type"
CONFIGURE SIPADAPTOR PROFILE NOTIFY_SIPFRAG_IN RULE 1 ADD ACTION 3 TYPE
HEADER
CONFIGURE SIPADAPTOR PROFILE NOTIFY_SIPFRAG_IN RULE 1 ACTION 3 OPERATION ADD
CONFIGURE SIPADAPTOR PROFILE NOTIFY_SIPFRAG_IN RULE 1 ACTION 3
HEADER_POSITION LAST
CONFIGURE SIPADAPTOR PROFILE NOTIFY_SIPFRAG_IN RULE 1 ACTION 3 HEADER_INFO
HEADERVALUE
CONFIGURE SIPADAPTOR PROFILE NOTIFY_SIPFRAG_IN RULE 1 ACTION 3 TO HEADER_NAME
"X-Jmac"
CONFIGURE SIPADAPTOR PROFILE NOTIFY_SIPFRAG_IN RULE 1 ACTION 3 FROM VALUE
"Tunnel Notify"
CONFIGURE SIPADAPTOR PROFILE NOTIFY_SIPFRAG_IN STATE ENABLE


% SHOW SIPADAPTOR PROFILE NOTIFY_SIPFRAG_IN ADMIN
Node: labgsx01                                Date: 2010/12/06 14:11:38  GMT
                                              Zone: GMTMINUS05-EASTERN-US

SIP Adaptor Profile Name:            NOTIFY_SIPFRAG_IN
SIP Adaptor Profile Index:           3
SIP Adaptor Profile State:           ENABLED

------------------------------------------------------
SIP Manipulation Rules
------------------------------------------------------
Rule Index:                          1
    Apply Match Header:              ONE
    Apply Match Header Range:

------------------------------------------------------
SIP Manipulation Criterions
------------------------------------------------------
Rule 1 MESSAGE Criterion
    Match Condition:               EXIST   Message Types:
REQUEST
    Method Types:                  NOTIFY
Rule 1 HEADER Criterion
    Match Condition:               EXIST
    Header Name:                   Content-Type
    Header Value:
    Header Instance:               ALL
    Header Range:
Rule 1 VARIABLE Criterion
```

```
    Match Condition:               ABSENT
    Variable:                      VAR1
    Match Value:

------------------------------------------------------
SIP Manipulation Actions
------------------------------------------------------
Rule 1 Action 1 Type VARIABLE
    Operation:            STORE
    From Operand:         message/unknown
    To Operand:           VAR1
Rule 1 Action 2 Type HEADER
    Operation:            MODIFY
    Header Position:
    Header Info:          HEADERVALUE
    From Operand:         VAR1
    To Operand:           Content-Type
Rule 1 Action 3 Type HEADER
    Operation:            ADD
    Header Position:      LAST
    Header Info:          HEADERVALUE
    From Operand:         Tunnel Notify
    To Operand:           X-Jmac
```

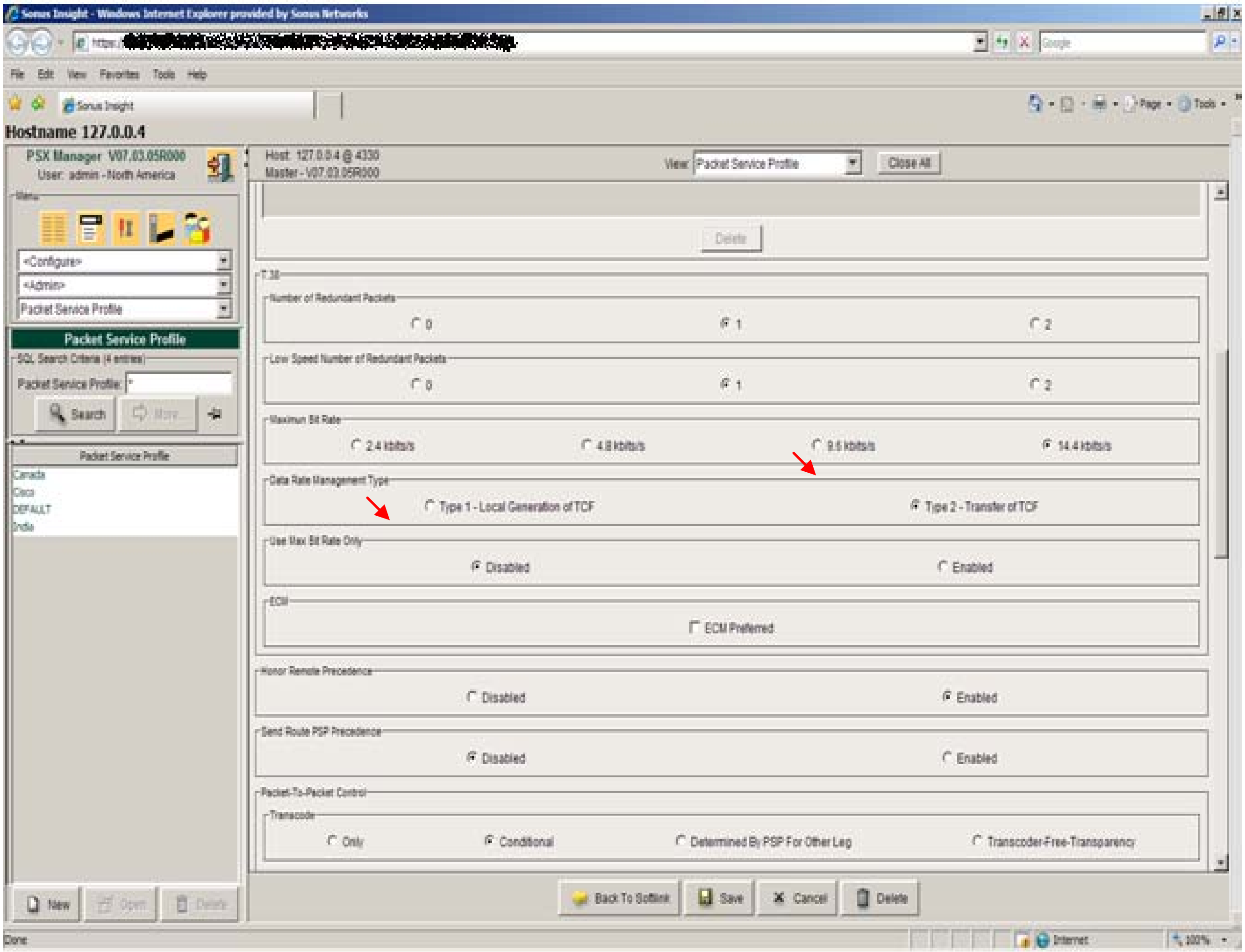
3.4.7 PSX GUI tables needed for Carrier SIP Server IPTG (This Carrier config reflects the settings needed for the AT&T testing and may change for other Carriers)

STEP 1: Create Packet Service Profile (using same one for Genesys)
Packet Service Profile (codecs G729AB, G711 and G726)



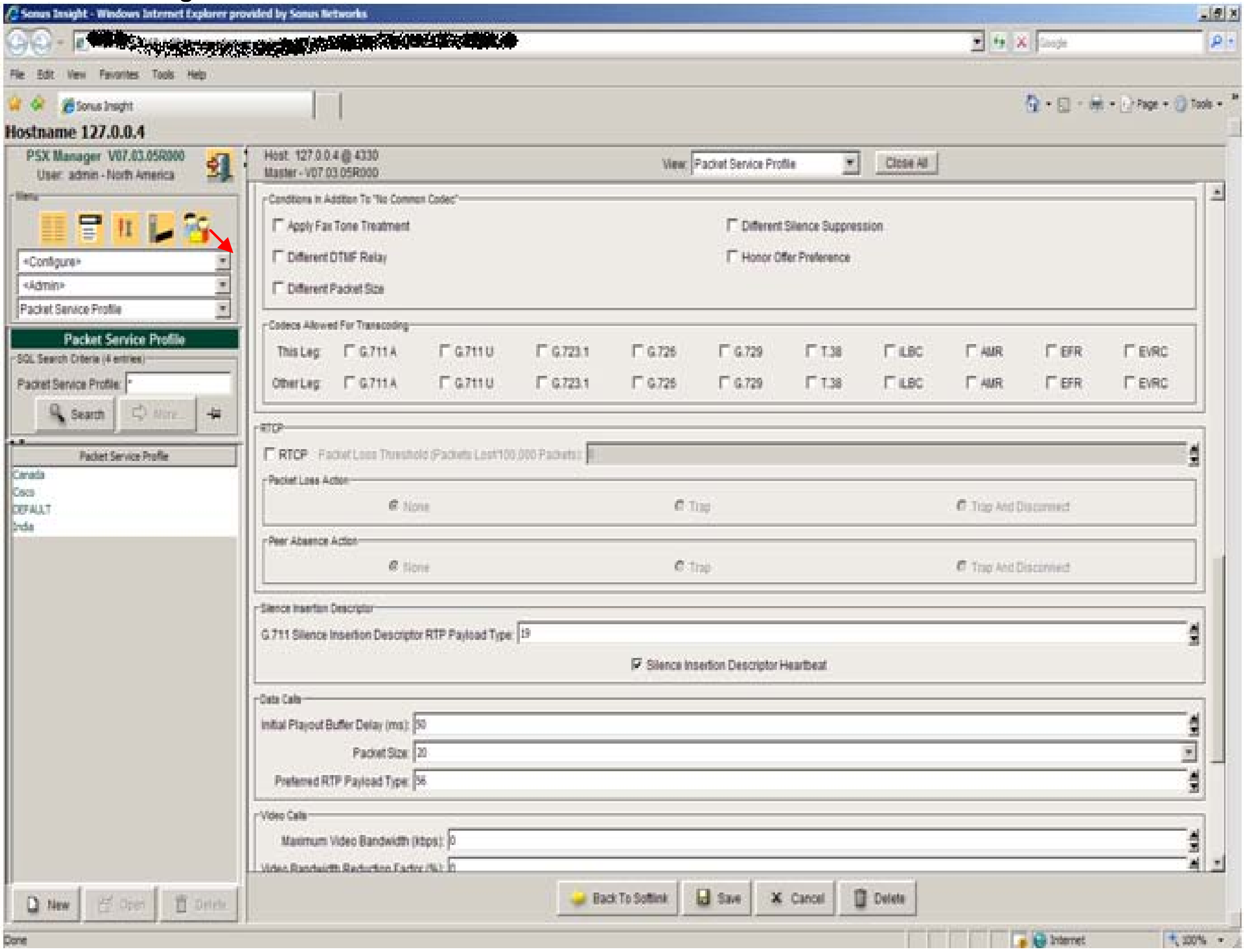
Screenshot 1 / 3

Honor Remote Precedence set to Enabled
Send Route PSP Precedence set to Disabled



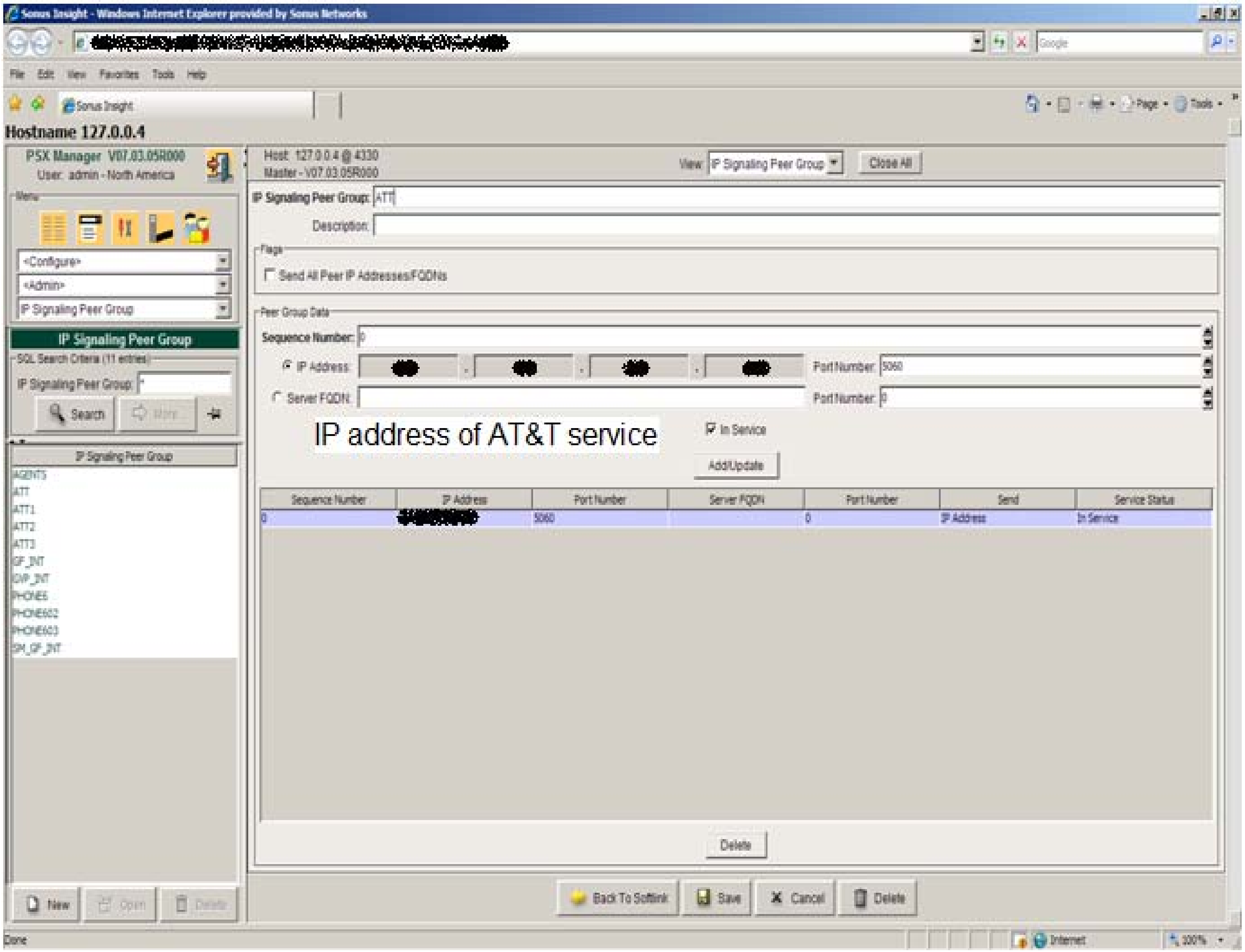
Screenshot 2 / 3

No Transcoding combos are selected



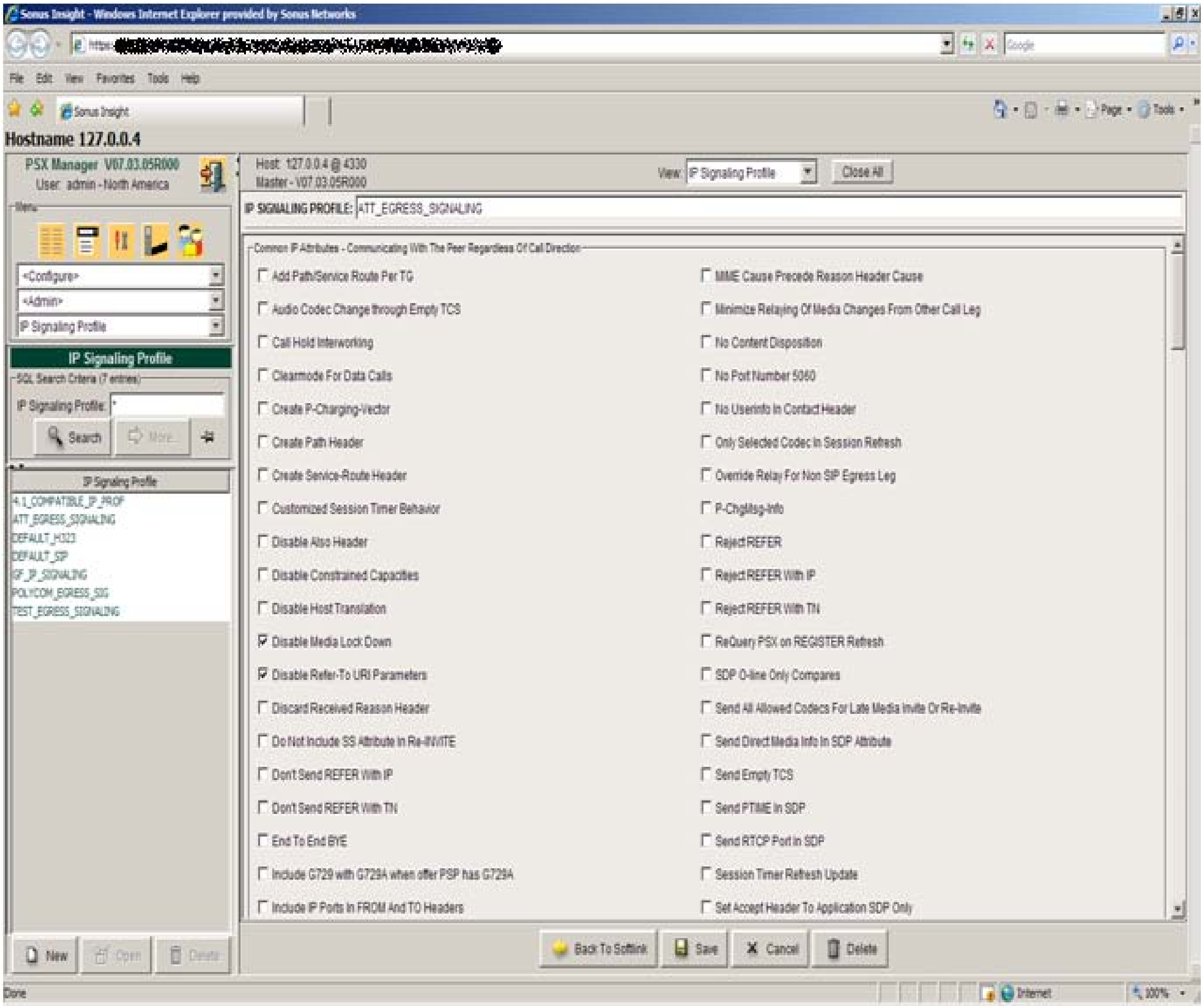
Screenshot 3 / 3

Step 2: IP Signaling Peer Group
Used for routing SIP calls to Carrier



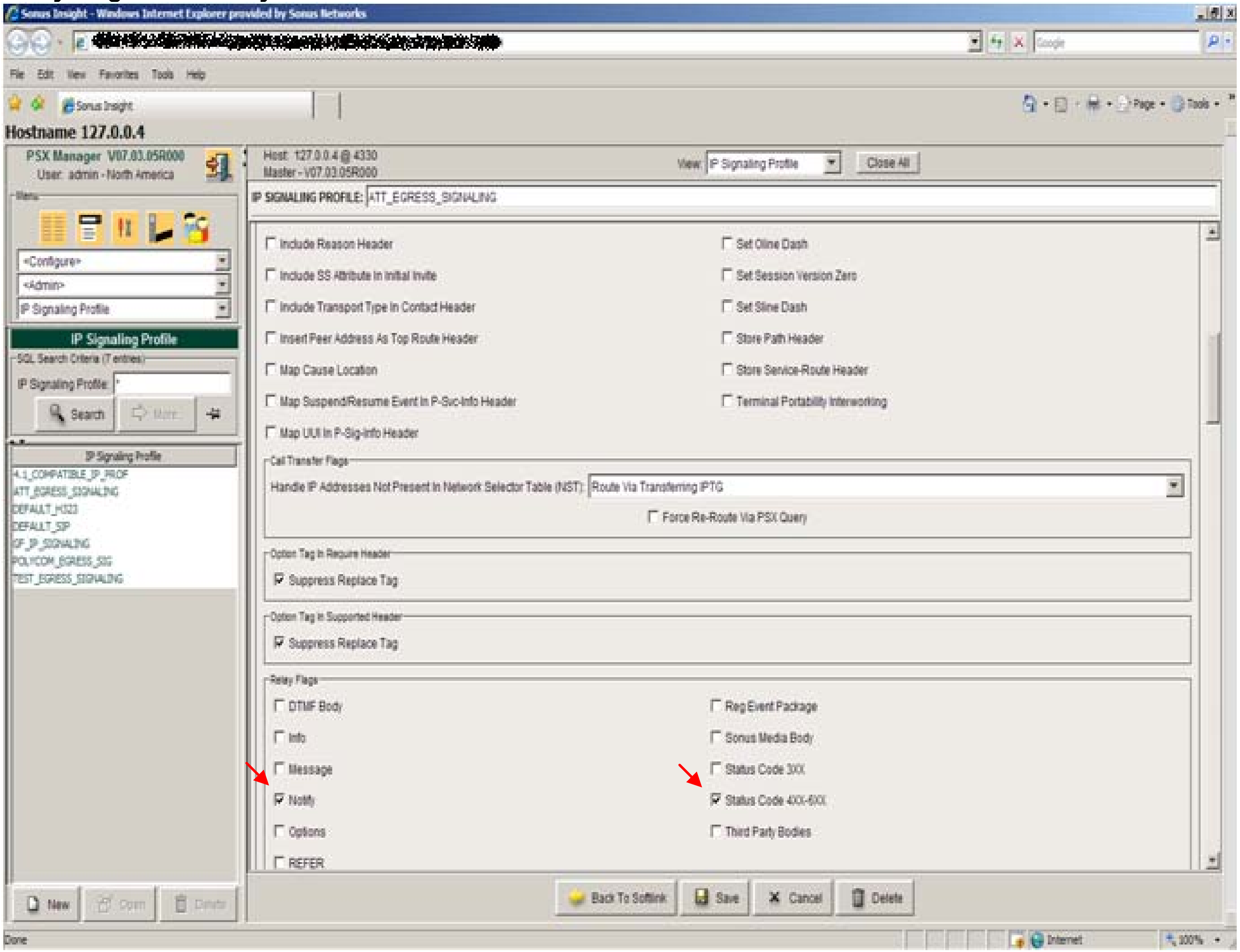
Screenshot 1/1

Step 3: IP Signaling Profile



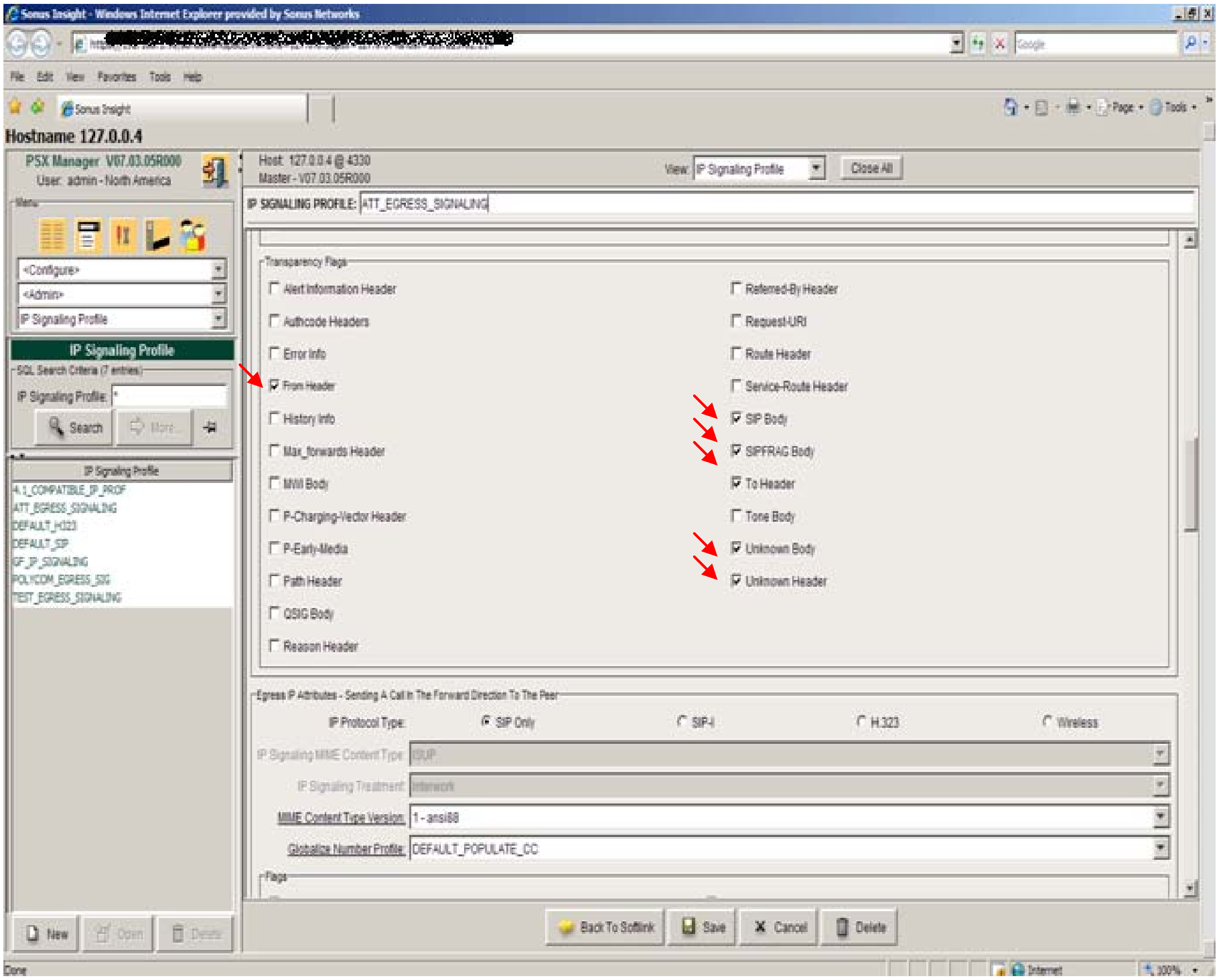
Screenshot 1 / 7

Relay flags set: Notify and 4XX-6XX

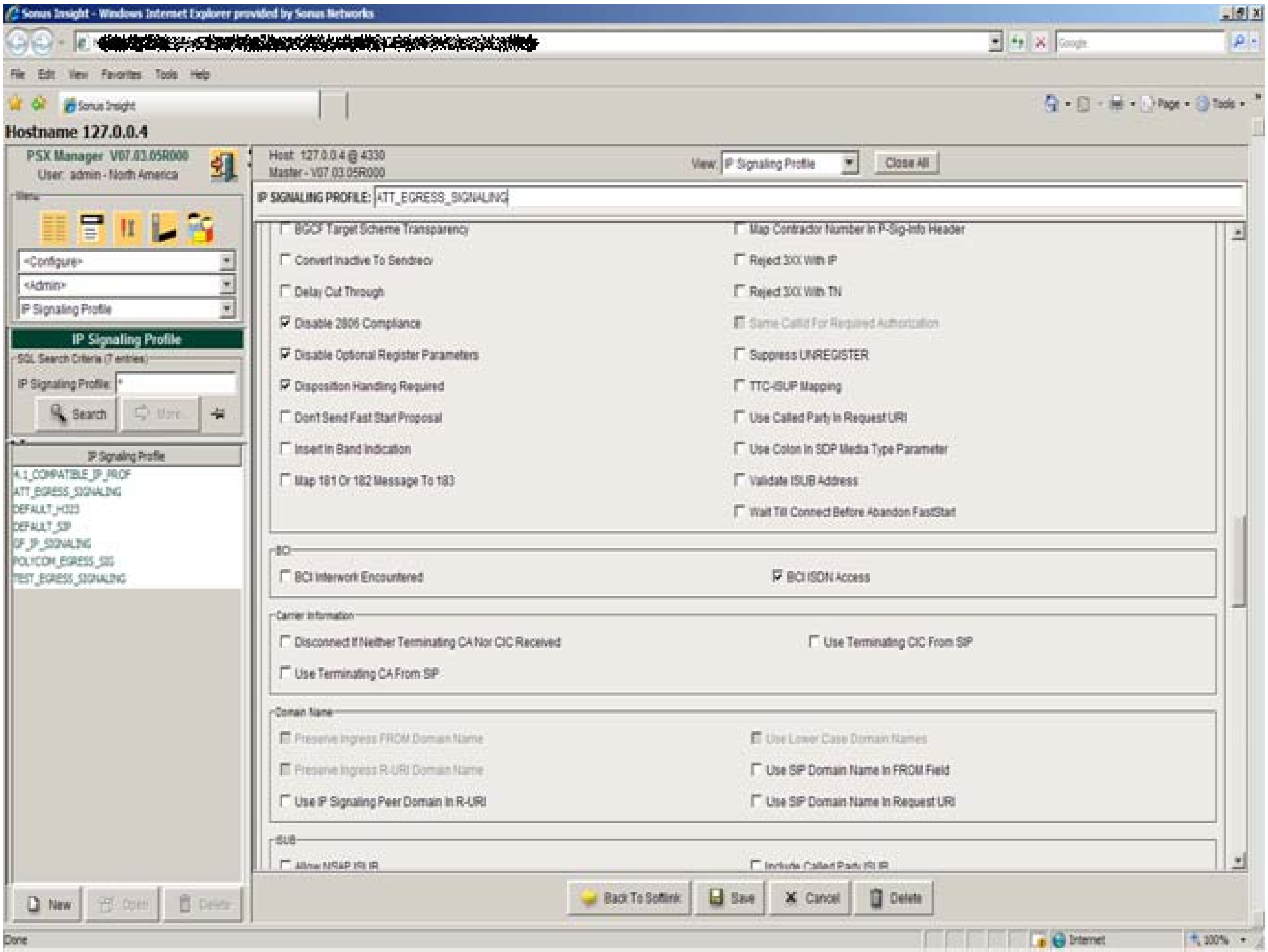


Screenshot 2 / 7

Transparancy Flags set: From Header, SIP Body, SIPFRAG Body, To Header, Unknown Body and Unknown Header.

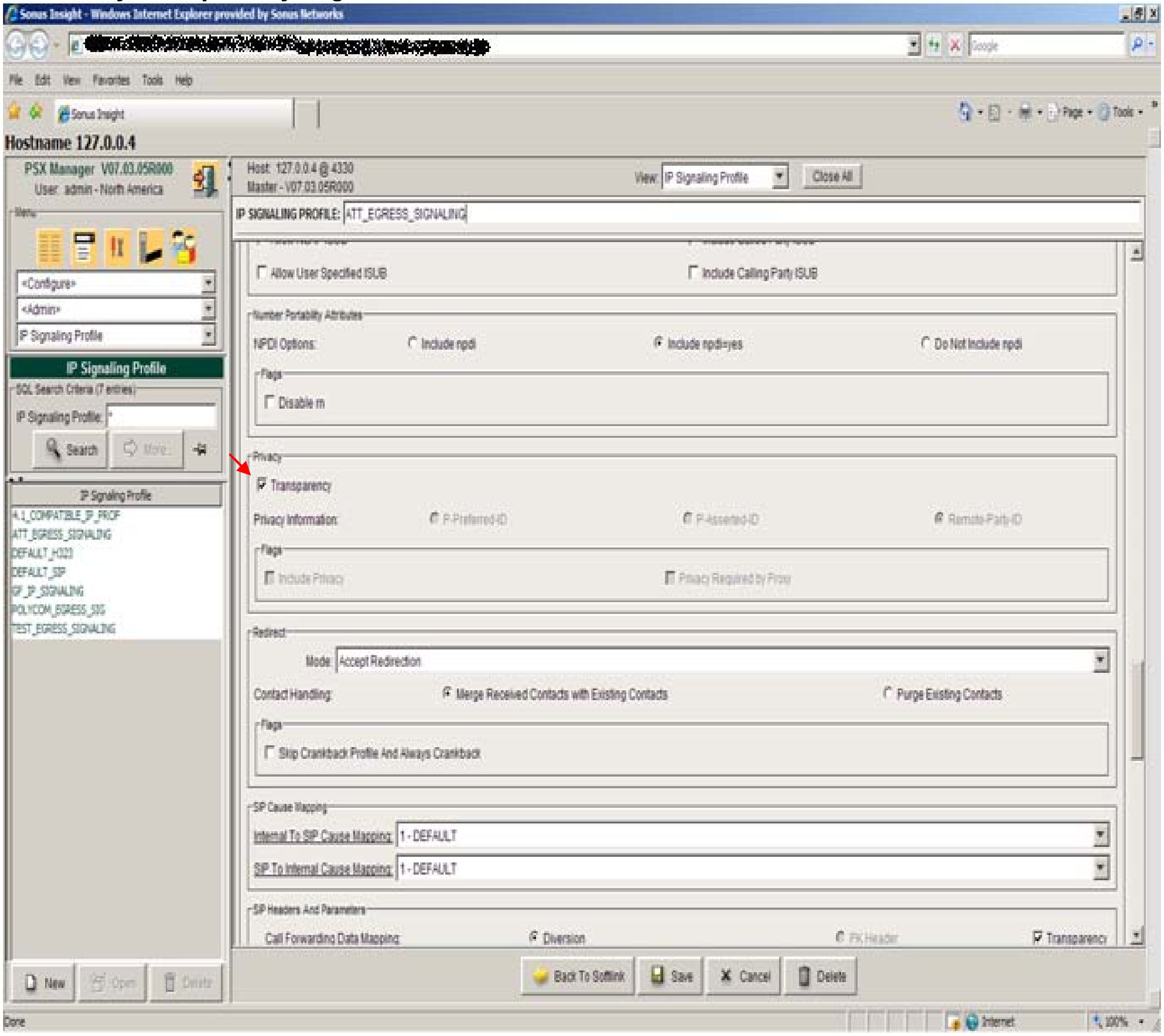


Screenshot 3/7

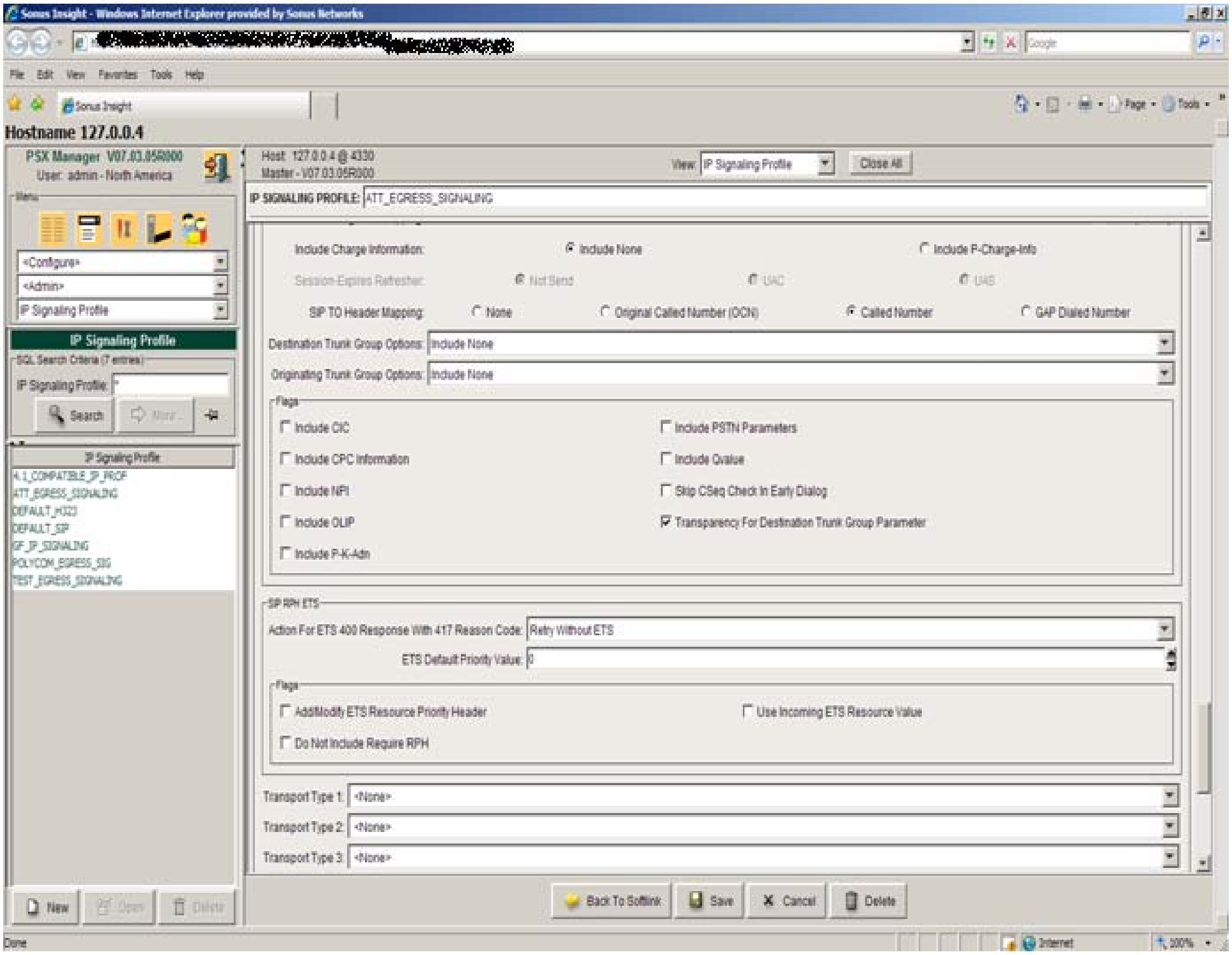


Screenshot 4 / 7

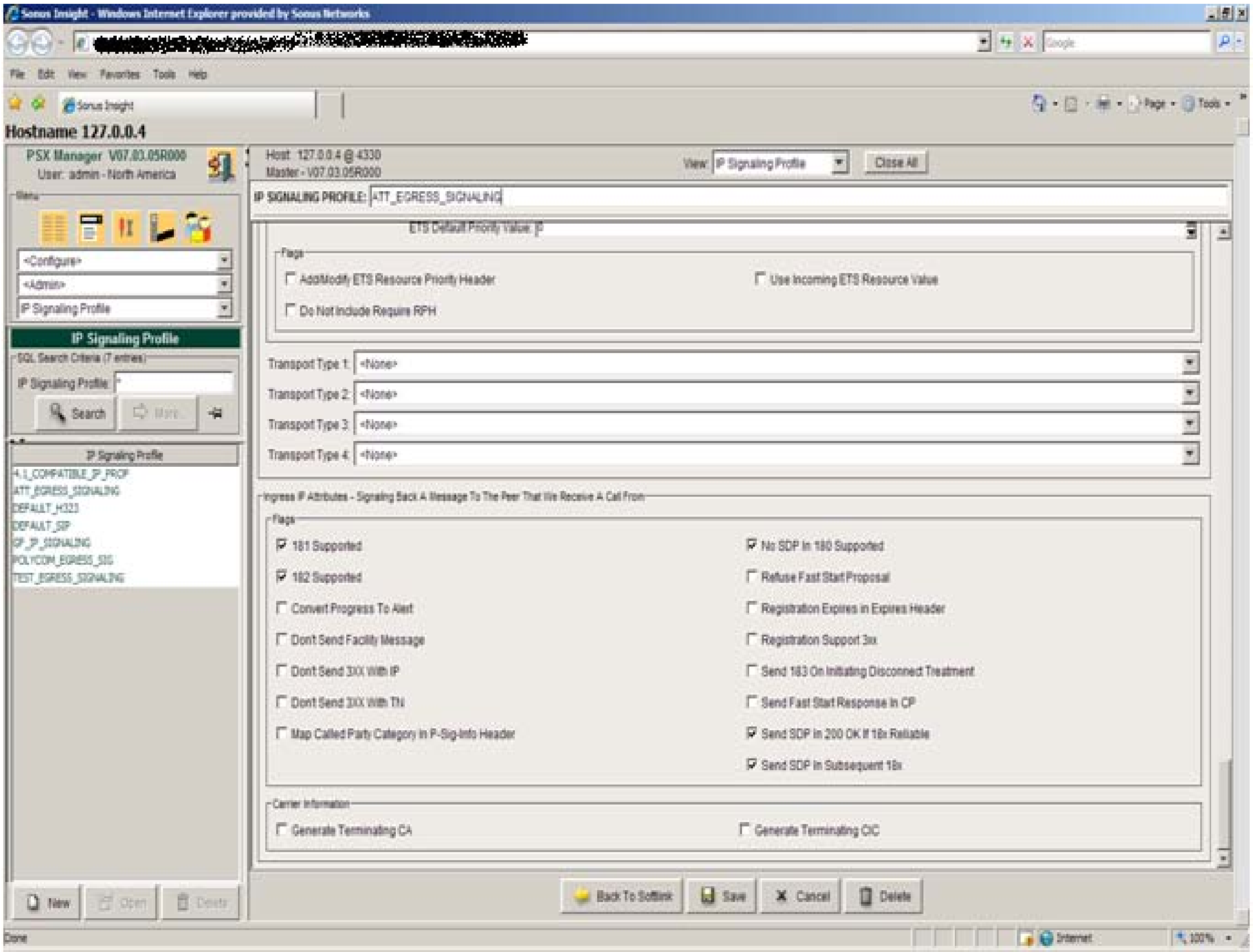
Set Privacy Transparency flag



Screenshot 5 / 7

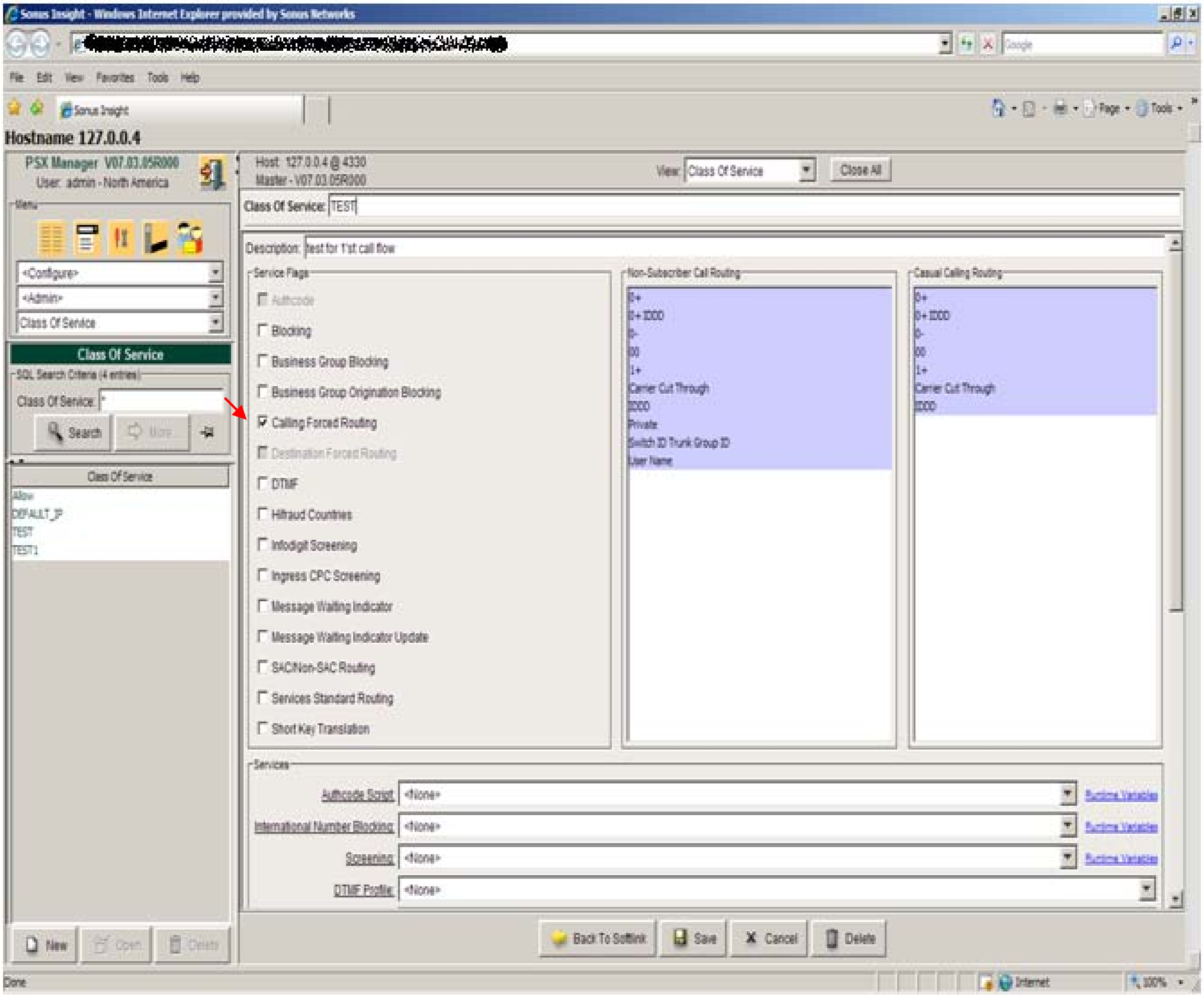


Screenshot 6 / 7

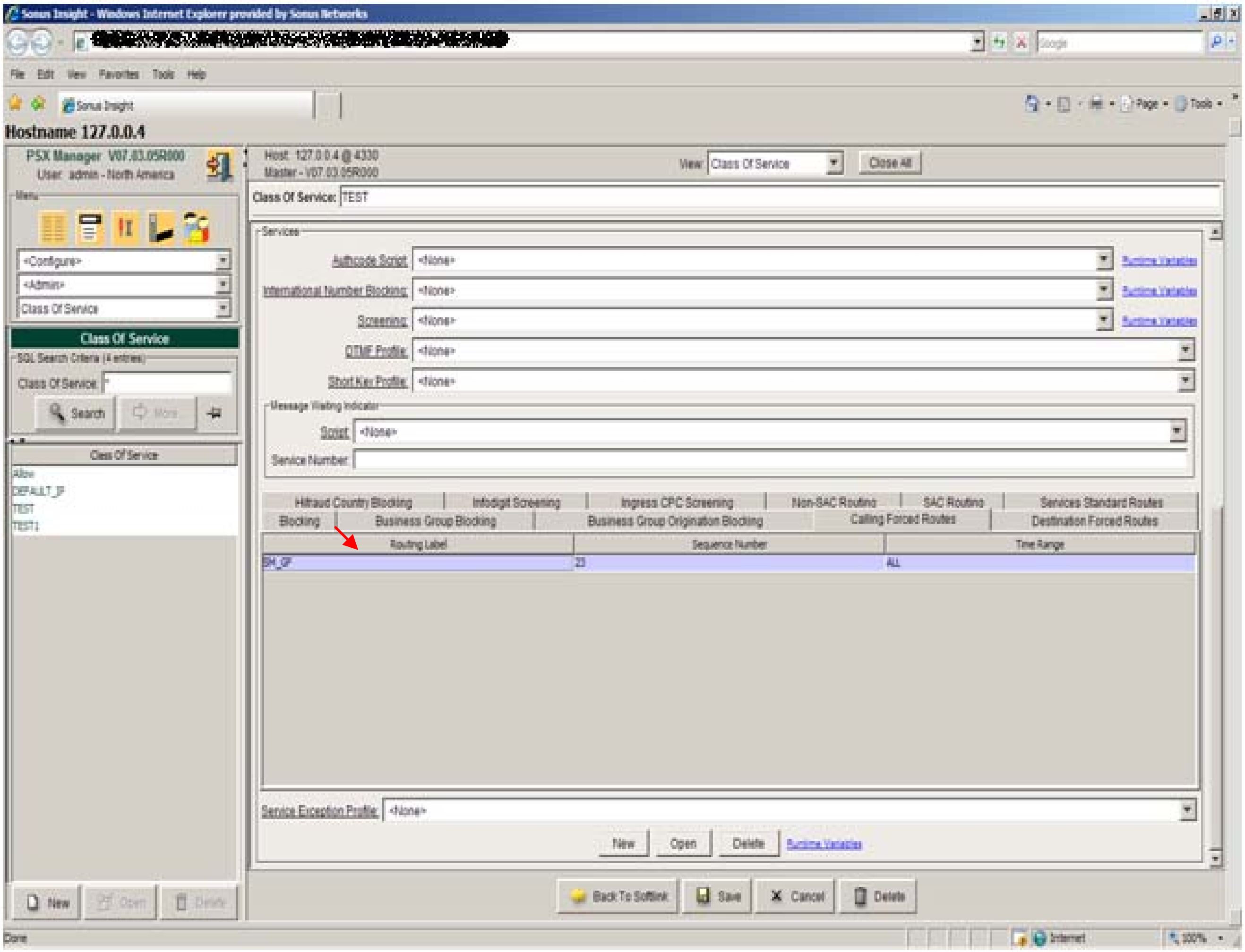


Screenshot 7 / 7

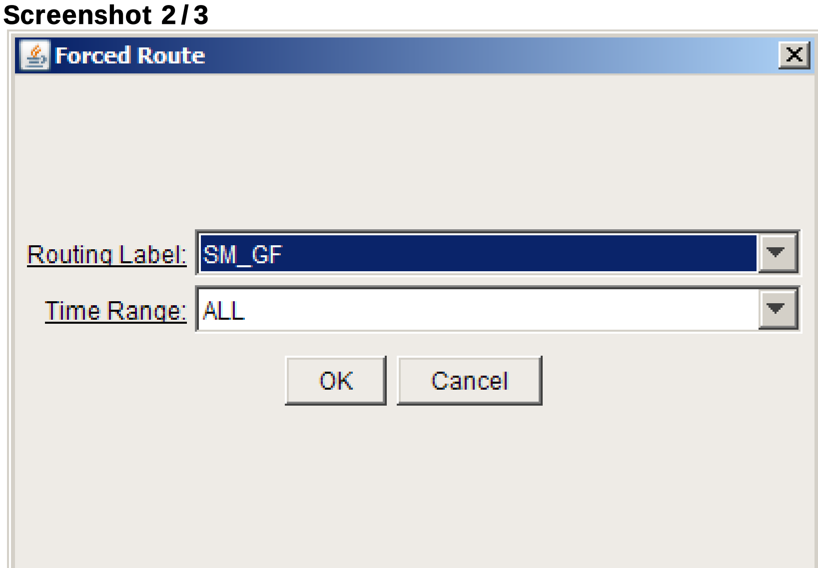
Step 4: Build Class of Service with Calling Forced Routing to Genesys



Screenshot 1 / 3

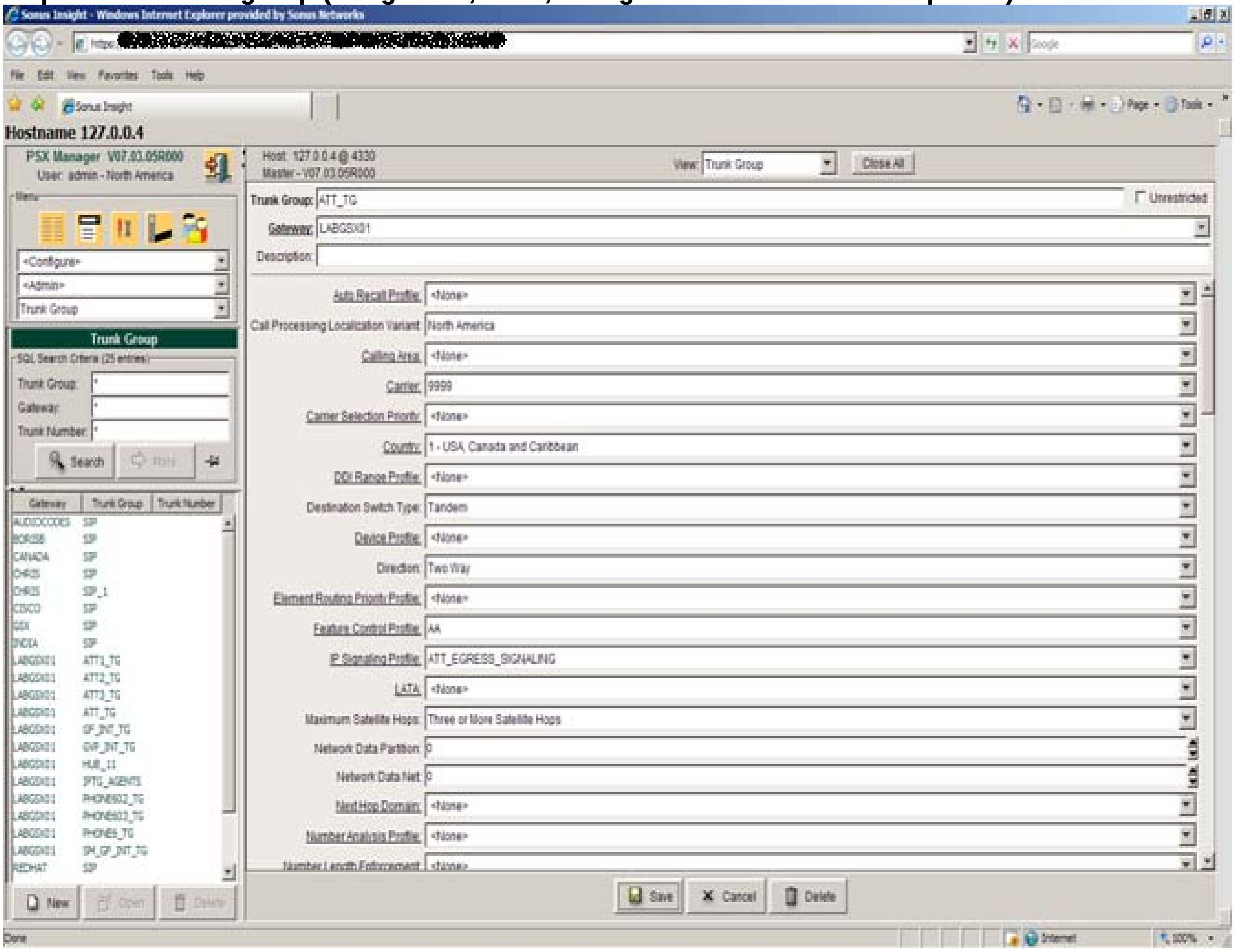


Screenshot 2 / 3

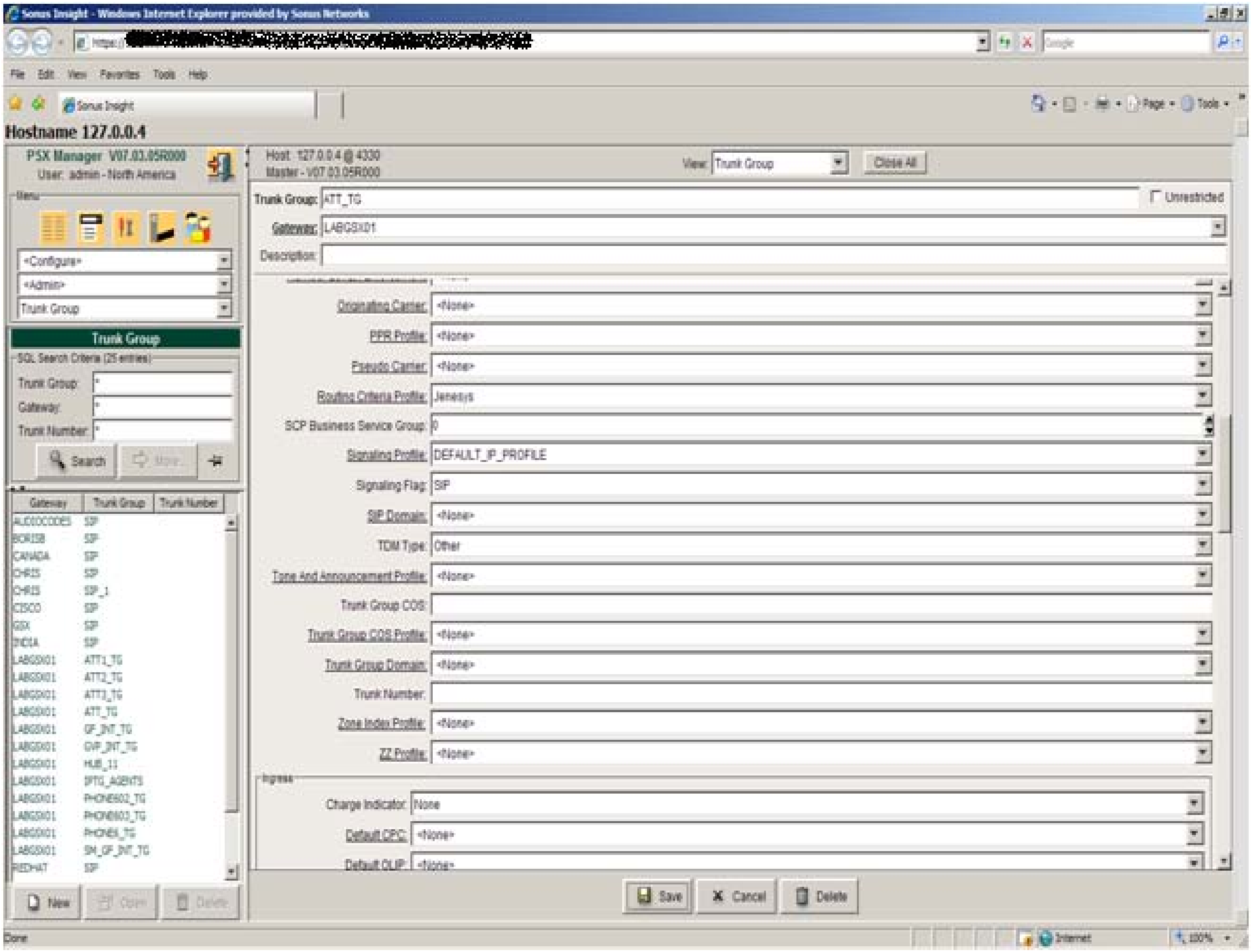


Screenshot 3 / 3

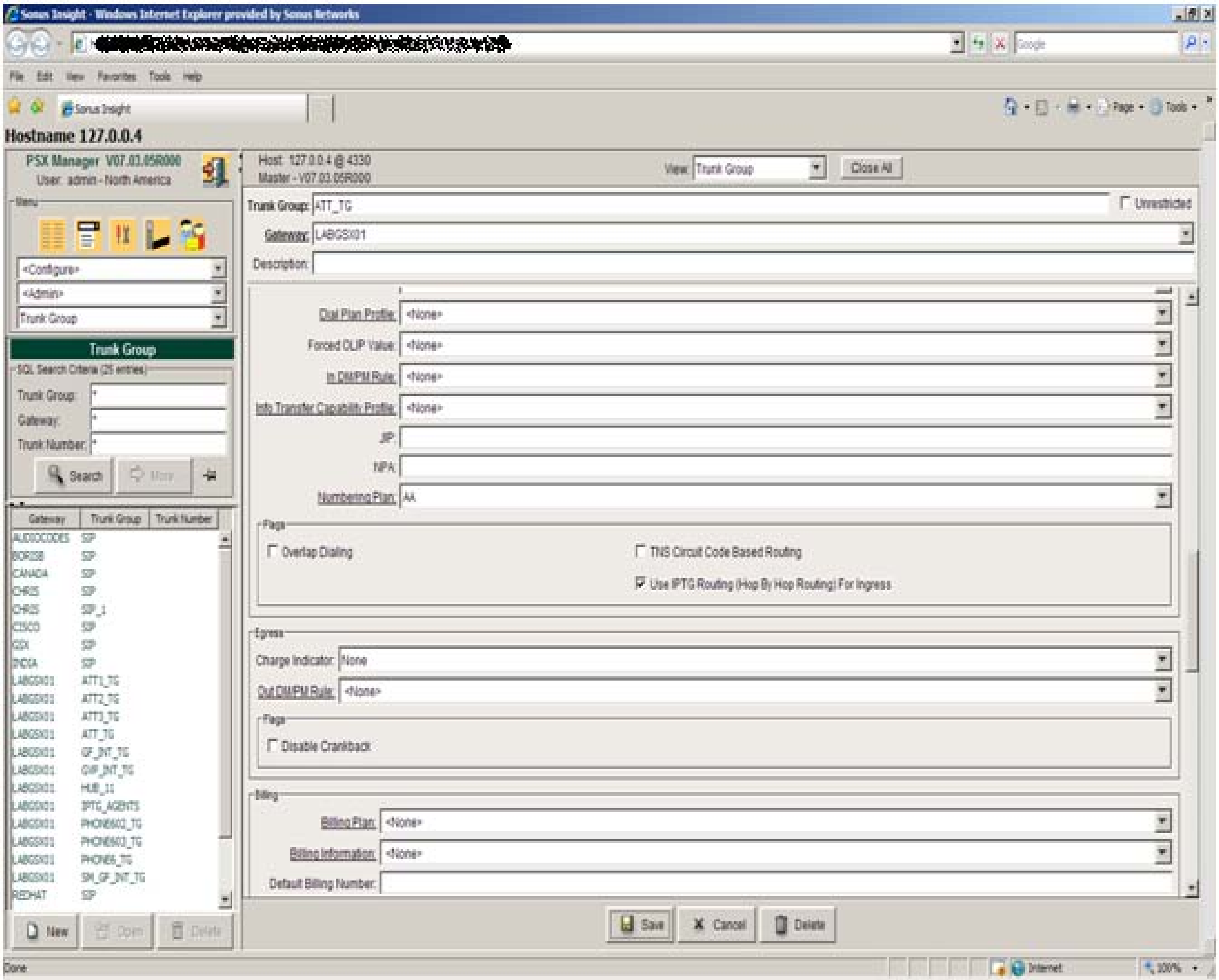
Step 5: Build Trunkgroup (assign PSP, IPSP, IP Sig Peer and COS from Steps 1-4)



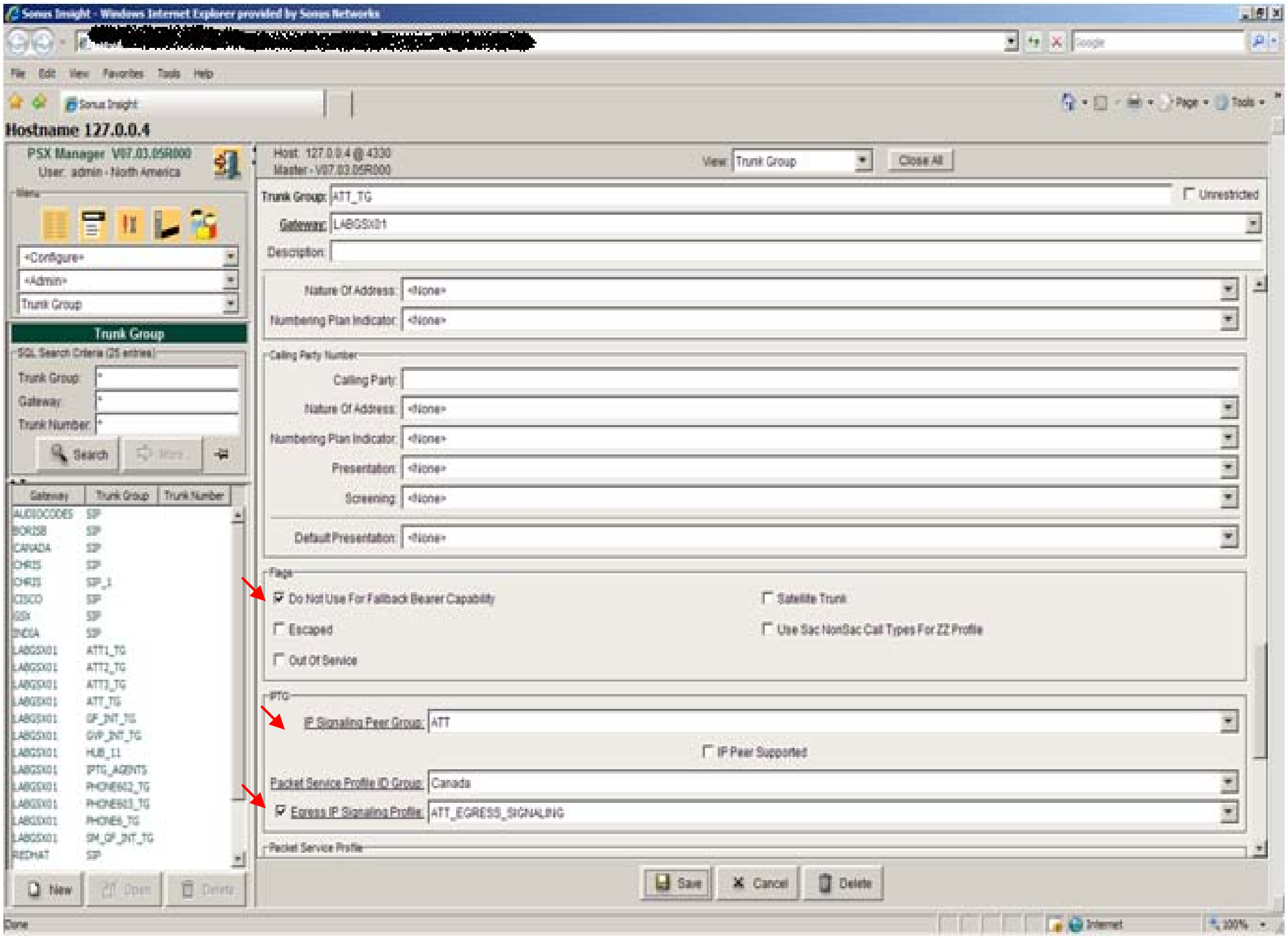
Screenshot 1 / 5



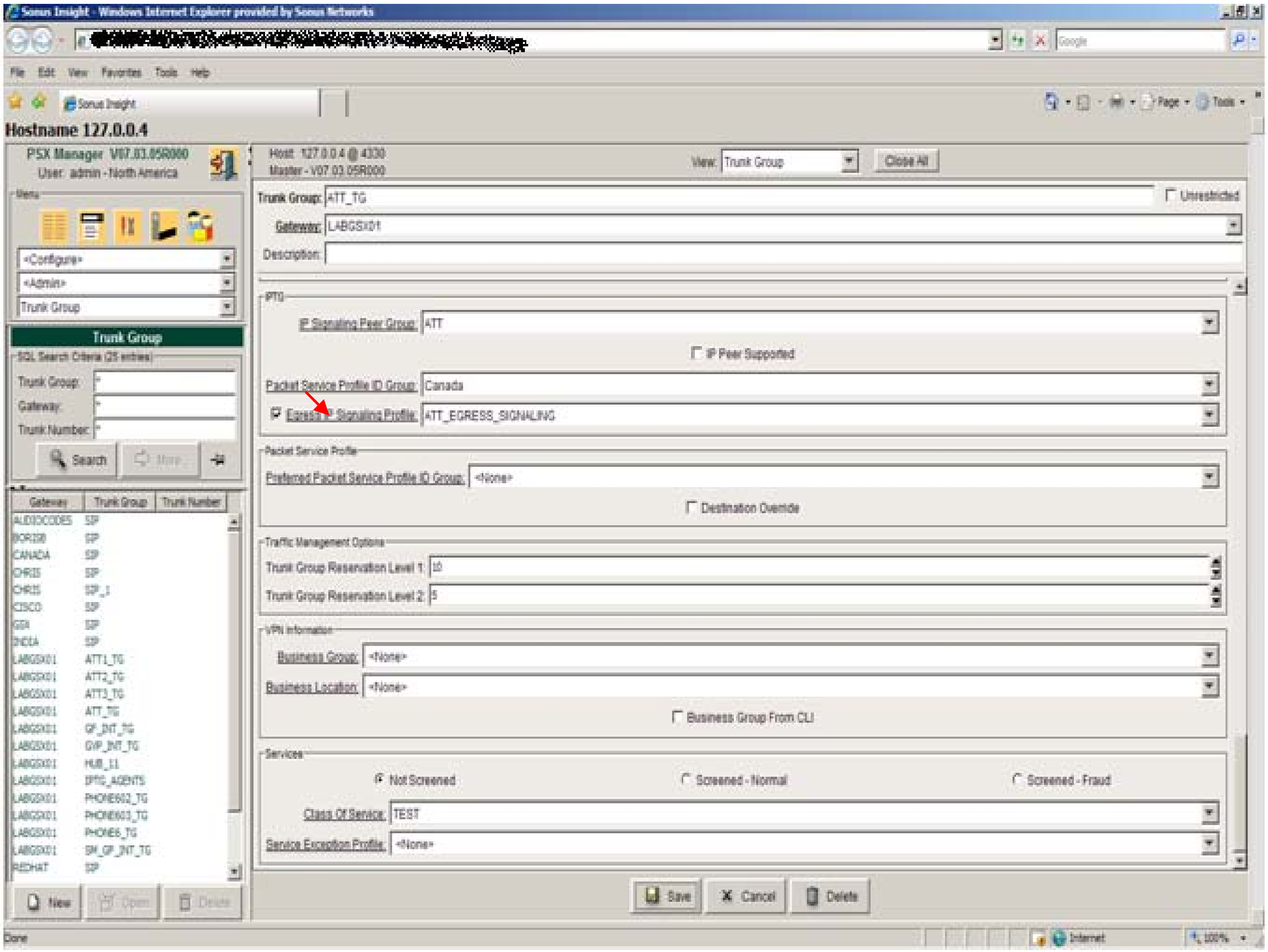
Screenshot 2 / 5



Screenshot 3 / 5



Screenshot 4 / 5



Screenshot 5 / 5

3.4.8 Optional section - GSX CLI commands for creating AgentsSip Server IPTG

This section is required only if SIP end points register to Genesys SIP Server via NBS.

1. Create Network Selector Table for ingress TG determination.
2. CREATE TRUNK GROUP IPTG_AGENTS.
3. Create SIP Service under TG IPTG_AGENTS - Config Sig Zone, Nif Group, Session Timer and Out Adaptor.

Note: If some SIP end points register to Genesys SIP Server through the NBS and not directly to the SIP Server, then the end points register to IP Trunkgroup IPTG_AGENTS on “external” IP interface. Sonus is configured to route traffic from “external” interface to Sonus “Internal” interface IPTG INT_GF_TG. The NBS “internal/trusted interface” is configured to communicate with Genesys SIP Server.

```
##############################################################
# Create Network Selector Table for ingress TG determination #
##############################################################


CREATE IP NETWORK SELECTOR TABLE AGENTS
CONFIGURE IP NETWORK SELECTOR TABLE AGENTS ADD NUMBER 1X.1X.1X.X MASK
255.255.255.0

% SHOW IP NETWORK SELECTOR TABLE AGENTS ADMIN
Node: labgsx01                               Date: 2010/12/06 15:25:51  GMT
                                             Zone: GMTMINUS05-EASTERN-US

Table Name                      Network Number    Network Mask
-----------------------------------------------------------------------
AGENTS                          1X.1X.1X.X      255.255.255.0

CREATE TRUNK GROUP IPTG_AGENTS
CONFIGURE TRUNK GROUP IPTG_AGENTS NETWORK SELECTOR TABLE AGENTS

% SHOW TRUNK GROUP IPTG_AGENTS ADMIN
Node: labgsx01                               Date: 2010/12/06 15:27:58  GMT
                                             Zone: GMTMINUS05-EASTERN-US


Local Trunk Name: IPTG_AGENTS
----------------------------------------------------------------------------
 State                                             ENABLED
 Inbound Reserve (percent)                         0
 Mode                                              INSERVICE
 Action                                            DRYUP
 Timeout (min)                                     5
 Circuit Reservation State                         DISABLED
 Reserved Priority Calls (circuits)                1
 Reserved Incoming Calls (circuits)                1
 Reserved Outgoing Calls (percent)                 10
 Alternate Trunk Group Name
 Trunk Group Rename Timer (sec)                    10
 SILC State                                        DISABLED
 SILC Congestion Level 1 Calls Allowed (percent) 075
 SILC Congestion Level 2 Calls Allowed (percent) 025
 Trunk Group Type                                  IPSELECTED
 IP Trunk Group Direction                          BOTHWAYS
 Parent IP Trunk Group
 IP Network Selection Table                        AGENTS
 IP Call Limit                                     UNLMT
 IP Bandwidth Limit                                UNLMT
```

```
 Packet Outage Detection Minimum Duration        6000
 Packet Outage Detection Minimum Calls           1000
 Packet Outage Detection Bandwidth Limit Reduct  50
 Packet Outage Detection State                   DISABLED
 Packet Outage Detection Interval (minutes)      15
 Master Trunk Group Name
 Calls Requested Per MTRG Request                100
 Bandwidth Requested Per MTRG Request (1K bps)   12400
 Maximum Ingress Sustained Call Rate             0
 Maximum Ingress Call Burst Size                 0
 Maximum Ingress Sustained SIP nonInvite Rate    0
 Maximum Ingress SIP nonInvite Burst Size        0
 Maximum Egress Sustained Call Rate              0
 Maximum Egress Call Burst Size                  0
 Maximum Egress Sustained SIP nonInvite Rate     0
 Maximum Egress SIP nonInvite Burst Size         0
 Ingress NonPriority Call Threshold              0
 Egress NonPriority Call Threshold               0
 HPC Profile Name                                defaultintipqueuing
 HPC Early ACM or SIP-18X                        USEDEFAULT
 HPC IP Oversubscription 0verride                DISABLED
 HPC IP Oversubscription Factor                  10
 Emergency IP Oversubscription Factor            10
 Local Policy Trunk Profile
 IP Registration Limit                           UNLMT
 IP Estimated Child Registrations                1

########################################################################
#         Create SIP Service under TG IPTG_AGENTS                      #
#         Config Sig Zone, Nif Group, Session Timer and Out Adaptor    #
########################################################################

CREATE SIP SERVICE SIP_AGENTS
CONFIGURE SIP SERVICE SIP_AGENTS SIGNALING ZONE SZ_OUTSIDE
CONFIGURE SIP SERVICE SIP_AGENTS MEDIA NIFGROUP NG_OUT
CONFIGURE SIP SERVICE SIP_AGENTS TIMER SESSIONKEEPALIVE 0

% SHOW SIP SERVICE SIP_AGENTS ADMIN
Node: labgsx01                                Date: 2010/12/06 15:29:57  GMT
                                              Zone: GMTMINUS05-EASTERN-US

-------------------------------------------------------------------------
 SIP Service                           : SIP_AGENTS
-------------------------------------------------------------------------
 Admin State                           : ENABLED
 Mode                                  : INSERVICE
 Action                                : DRYUP
 Dryup Timeout (min)                   : 5
 Trunk Group                           : IPTG_AGENTS
 Disc Treatment                        : sipDefault
 Tone Package                          : default
 Source Address Filtering              : DISABLED
 Ans Supervision Timeout               : 300
 Ans Supervision Timeout Action        : RELEASE
 Signaling Zone                        : SZ_OUTSIDE #untrusted signaling
 Media Zone                            : INTERNAL
 Media NIF Group                   : NG_OUT    #untrusted media
 NAPT for Signaling                    : DISABLED
 NAPT for Media                        : DISABLED
 NAPT QualificationTable name          :
 Parse Embedded BGID                   : DISABLED
 Congestion Reject Method              : RELEASE
 Congestion Retry Timer Min (sec)      : 10
 Congestion Retry Timer Max (sec)      : 30
```

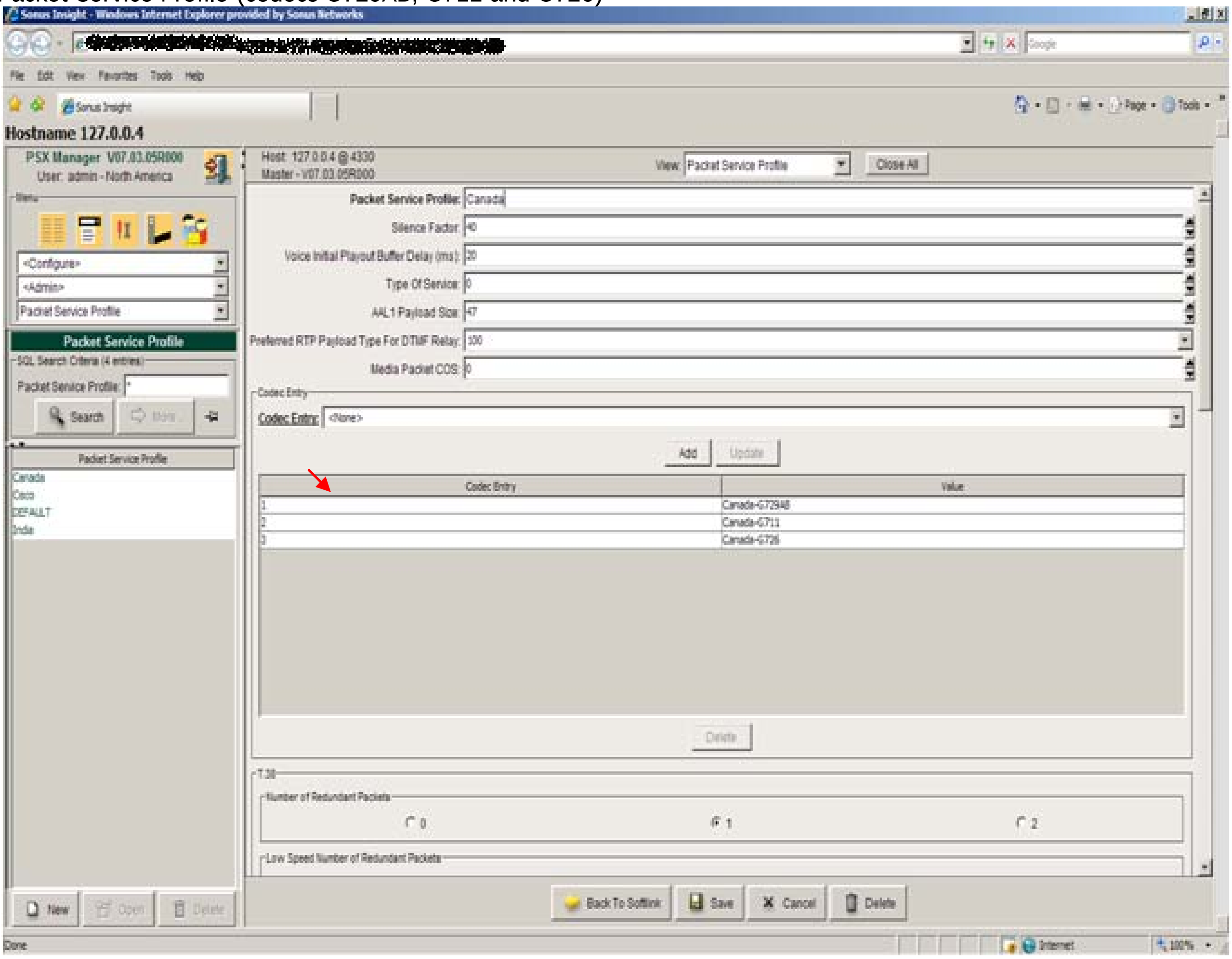
```
Congestion Release Timeout (sec)      : 0
SIP Timer T1 (msec)                   : 500
SIP Timer T2 (msec)                   : 4000
SIP Session Keepalive Timer (sec)     : 0
SIP Session Term Delta Time (sec)     : 0
SIP Minimum Session Timer (sec)       : 90
Retry Count for SIP Messages          : 7
Retry Count for INVITE Message        : 6
Retry Count for RE-INVITE Message     : 0
Retry Count for BYE Message           : 3
Retry Count for CANCEL Message        : 3
Session keepalive retry  on 422       : 5
Session keepalive retry  on 491       : 5
Use Route Set                         : DISABLED
OPTIONS                               : ALLOW
REFER                                 : ALLOW
SUBSCRIBE                             : ALLOW
NOTIFY                                : ALLOW
INFO                                  : ALLOW
REGISTER                              : ALLOW
MESSAGE                               : ALLOW
PUBLISH                               : ALLOW
Address Reachability Service Profile :
REGISTER redirection method           : NONE
Registration                          : SUPPORTED #requiring registration
Registrant CAC Profile                :
Use CallingParty from PAI(priority1) : ENABLED
Use CallingParty from PPI(priority2) : ENABLED
Use CallingParty from RPI(priority3) : ENABLED
Use CallingParty from FROM(priority4): ENABLED
Registrar Minimum Expires (sec)       : 3600
Use CPC Param Received in             : DEFAULT
Relay ISUP MIME Body                  : DISABLED
Privacy Param Restricted              : DEFAULT
Long Duration Call Timeout (mins)     : 0
Long Duration Call Action             : NOACTION
Long Duration Call Release Cause      : 41
Long Duration Call Emergency Calls    : EXCLUDE
Resource Priority Header Profile      : defaultSipResPriorProf
Variant Type                          : SONUS
Trusted Source flag                   : ENABLED
COMEDIA connection role               : NONE
Crank Back Profile                    : 1
Skip Crank Back Profile               : DISABLED
DNS Support                           : A-ONLY
Receive Side Filter Profile           :
Direct Media Allowed                  : DISABLED
TCP Retransmit Interval in Seconds    : 6
SCTP Retransmit Interval in Seconds   : 6
Registration Max-Expires NON-NAT      : 3600
Registration Max-Expires NAT-TCP      : 240
Registration Max-Expires NAT-UDP      : 60
Call Redirection                      : ENABLED
Transport Protocol Preference #1      : NONE
Transport Protocol Preference #2      : NONE
Transport Protocol Preference #3      : NONE
Transport Protocol Preference #4      : NONE
Factor Value for Hop Counter          : 1
Max Fwds Hdr Default                  : 70
Route Msg Validation                  : NOVALIDATION
Overlap Addressing Support            : DISABLED
Overlap Min Digits For Query          : 0
Overlap Timer Digit Collection        : 10
Overlap Timer IOW3                    : 4
```

```
Timer IOW2                              : 0
Inter Operator ID                       :
URI PRESENTATION PREFERENCE             : NONE
Additional Headers Transmit Profile     :
Strict Parse                            : DISABLED
TMR Unrestricted 64kbit/s               : DISABLED
Include Application Headers             : DISABLED
Transmit Preconditions                  : NONE
Receive Preconditions                   : NONE
DataPathMode Passthru                   : DISABLED
CPC to SIP Cause Map Profile Index      : 0
SIP to CPC Cause Map Profile Index      : 0
Set NOA to International                : DISABLED
Relay Non-Invite Requests               : DISABLED
Default MaxPtime                        : 150
The 100Rel support                      : ENABLED
Late Media support                      : CONVERT
Emergency Profile                       :
Estimated Child Registrations           : 1
Input Adaptor Profile                   :
Output Adaptor Profile                  :
Bckwd Info Msg After Confirmed Dialog: DISABLED
Use Ingress Originating CA              : DISABLED
Add Egress Originating CA               : DISABLED
ISDN SubAddress Preference              : RFC2806
Peer Overload Throttling                : DISABLED
Dynamic Blacklist Profile               :
Send Originating CIC                    : DISABLED
Use Ingress Charge Info                 : DISABLED
Send Charge Info                        : DISABLED
Media Recording                         : DISABLED
Refer Reject Response Code              : 403
Redirect Disconnect Code                : 503
```

3.4.9 Optional section - PSX GUI tables needed for Agents Server IPTG (The agent configuration had agents registering on their own IPTG through NBS to Genesys SIP Server)

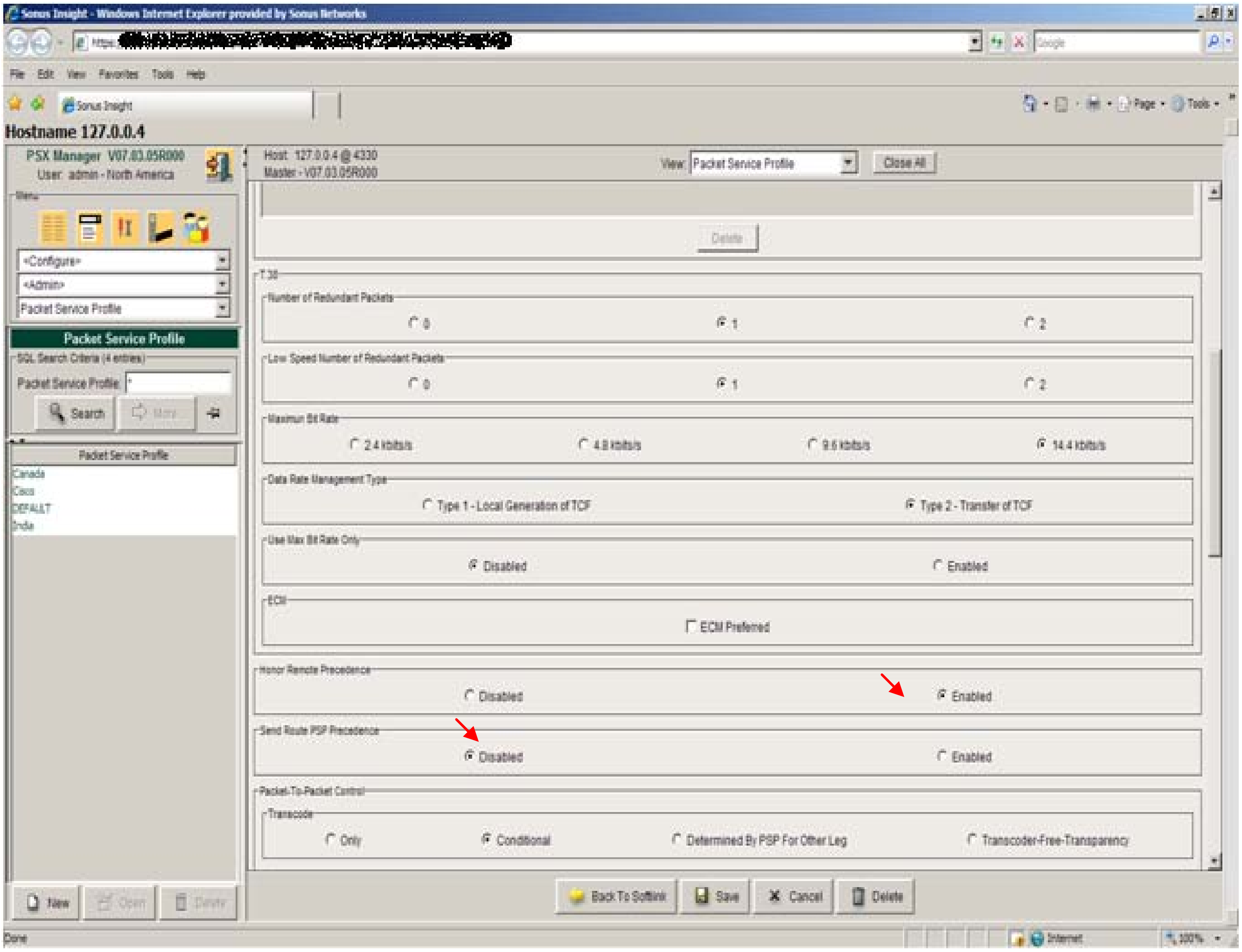
STEP 1: Create Packet Service Profile (using same one for Genesys)

Packet Service Profile (codecs G729AB, G711 and G726)



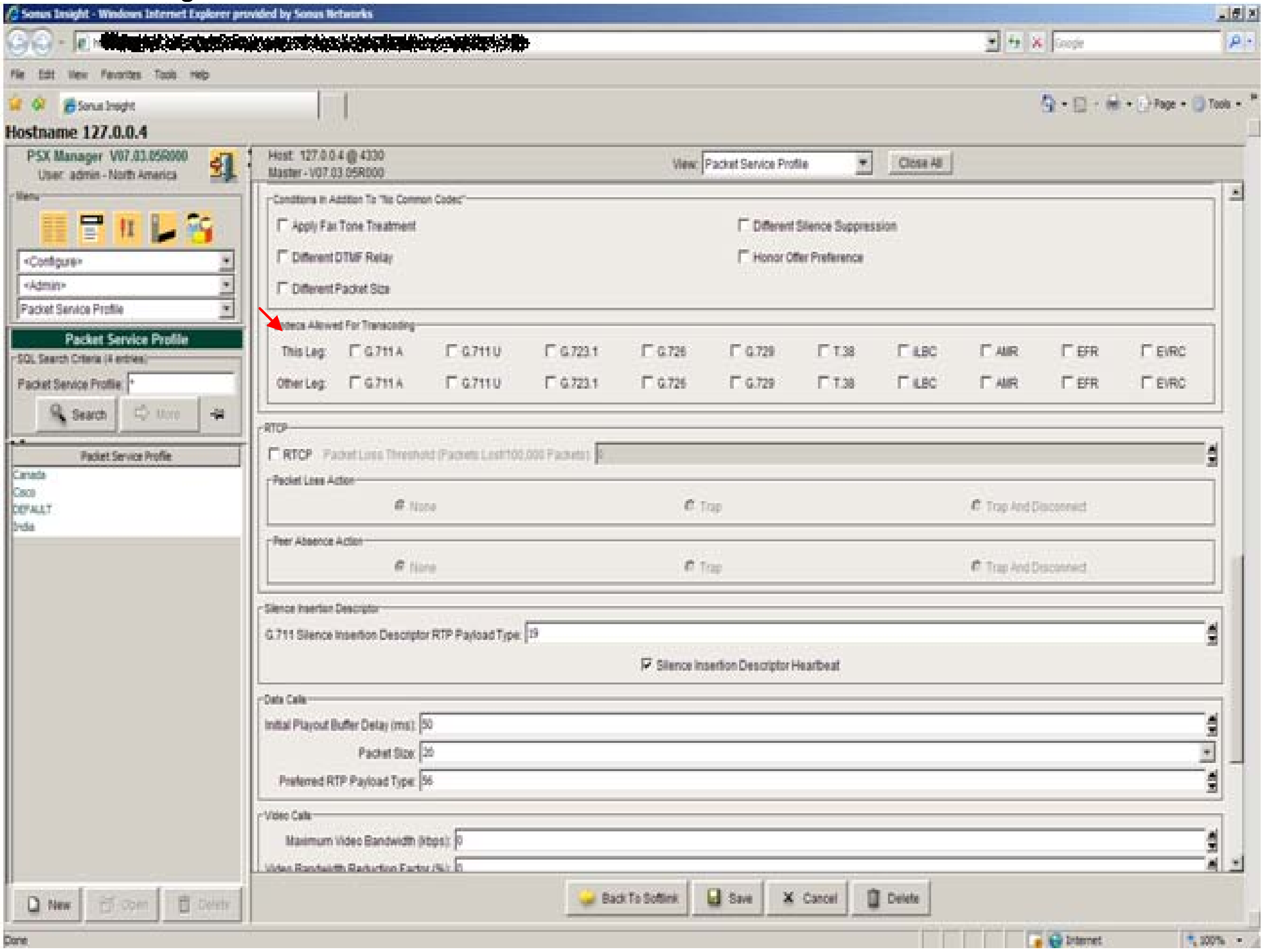
Screenshot 1/3

Honor Remote Precedence set to Enabled
Send Route PSP Precedence set to Disabled



Screenshot 2 / 3

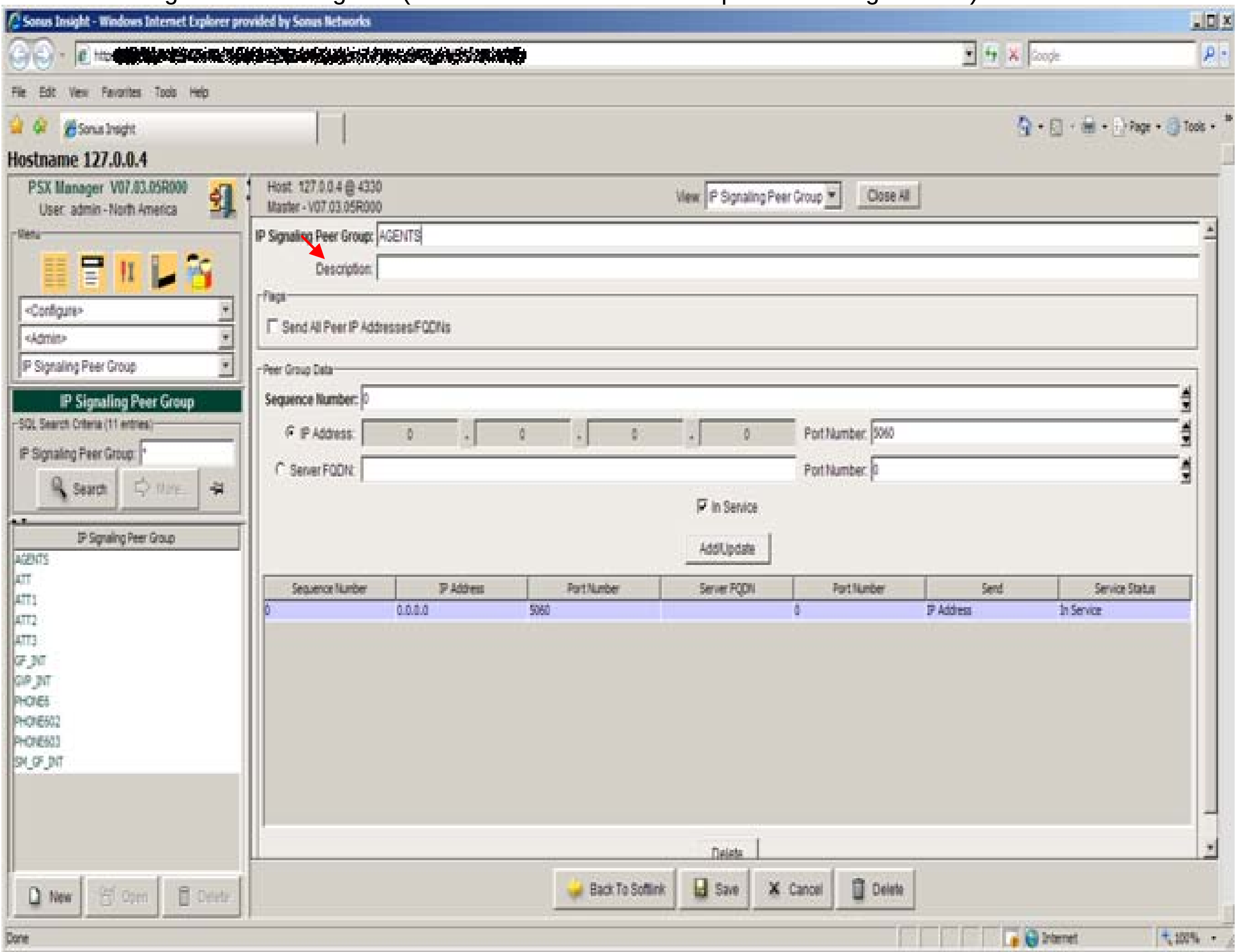
No Transcoding combos are selected



Screenshot 3/3

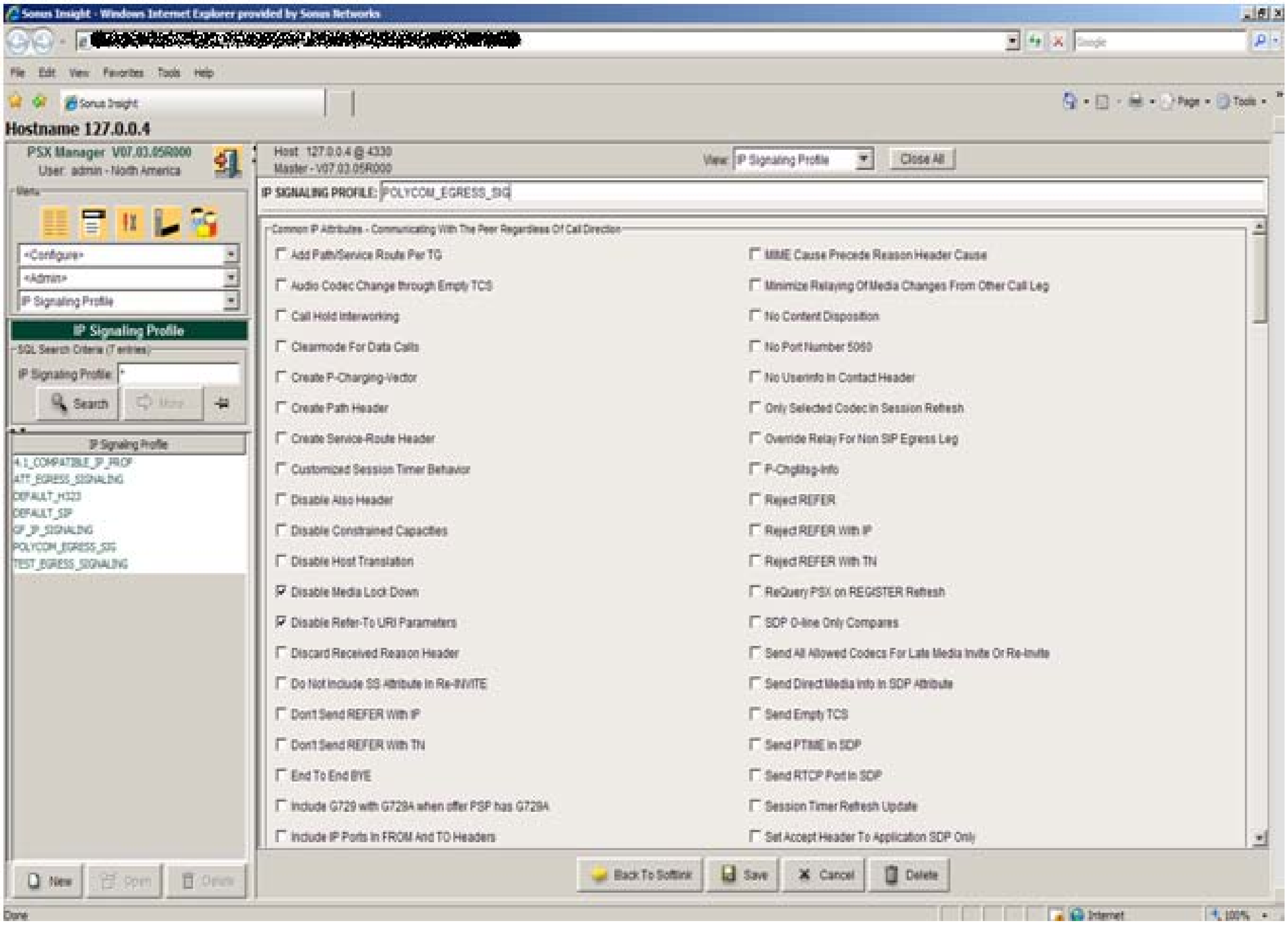
Step 2: IP Signaling Peer Group

Used for routing SIP calls to Agents (0.0.0.0 is defined when endpoints use registration)



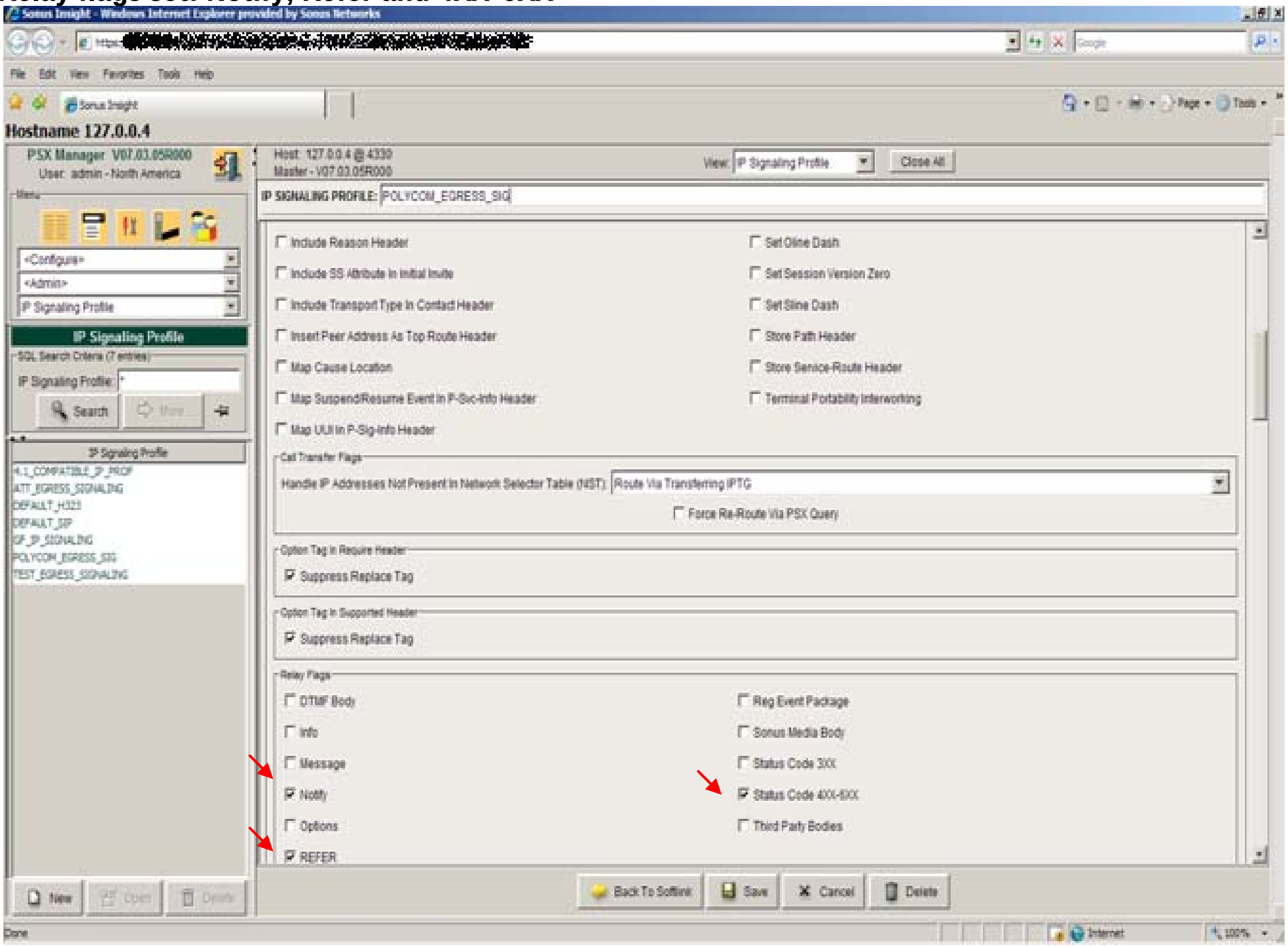
Screenshot 1 / 1

Step 3: IP Signaling Profile



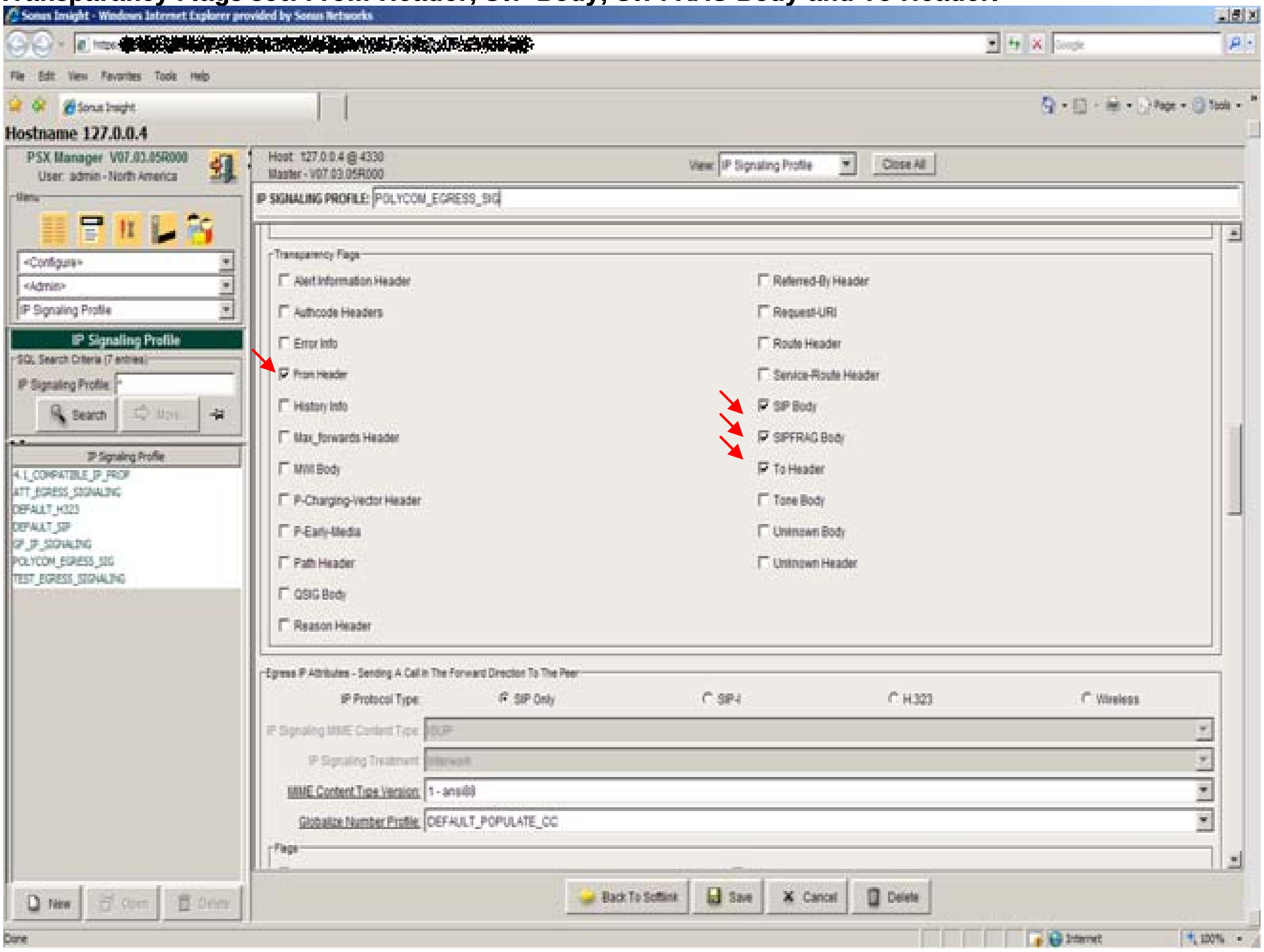
Screenshot 1 / 7

Relay flags set: Notify, Refer and 4XX-6XX

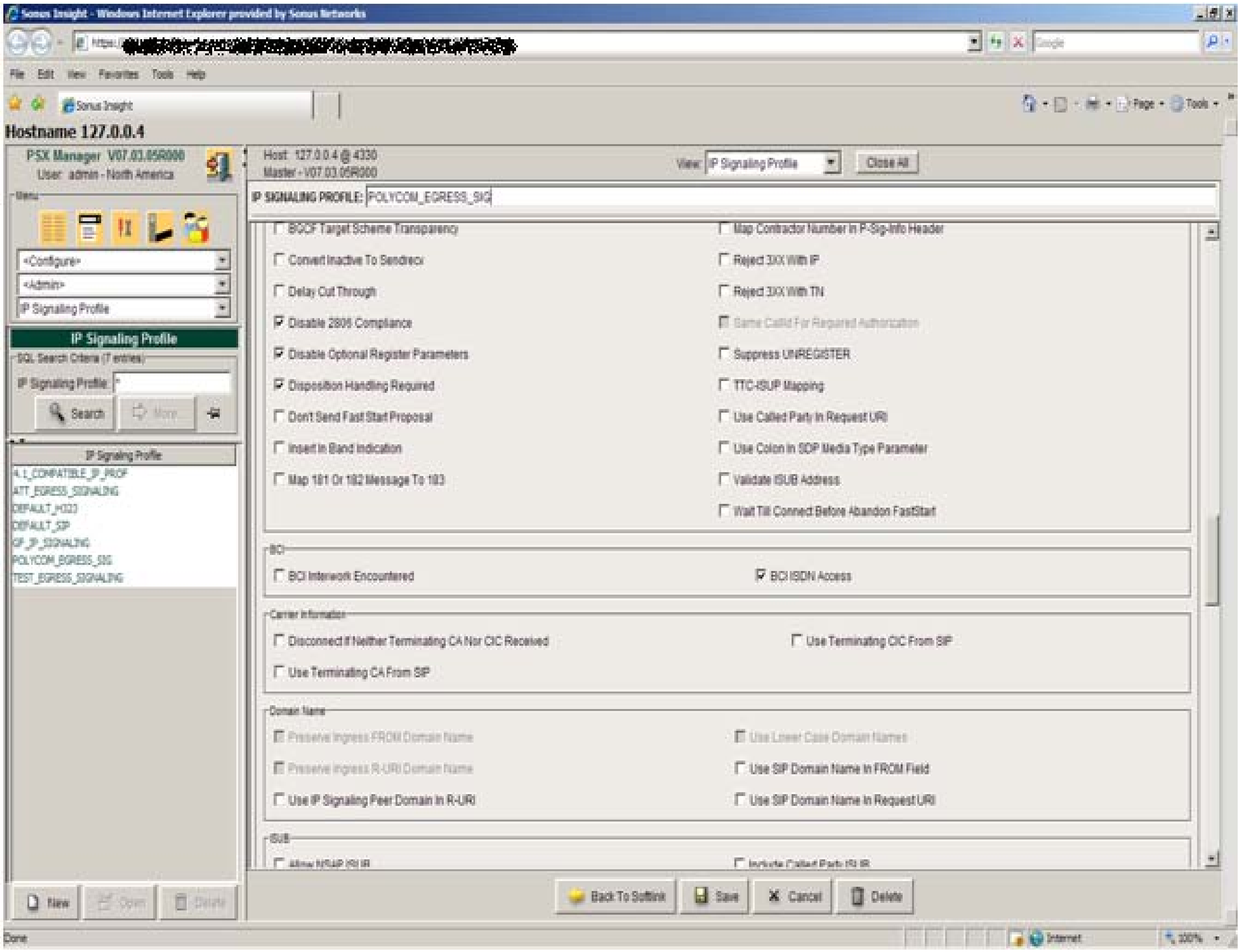


Screenshot 2 / 7

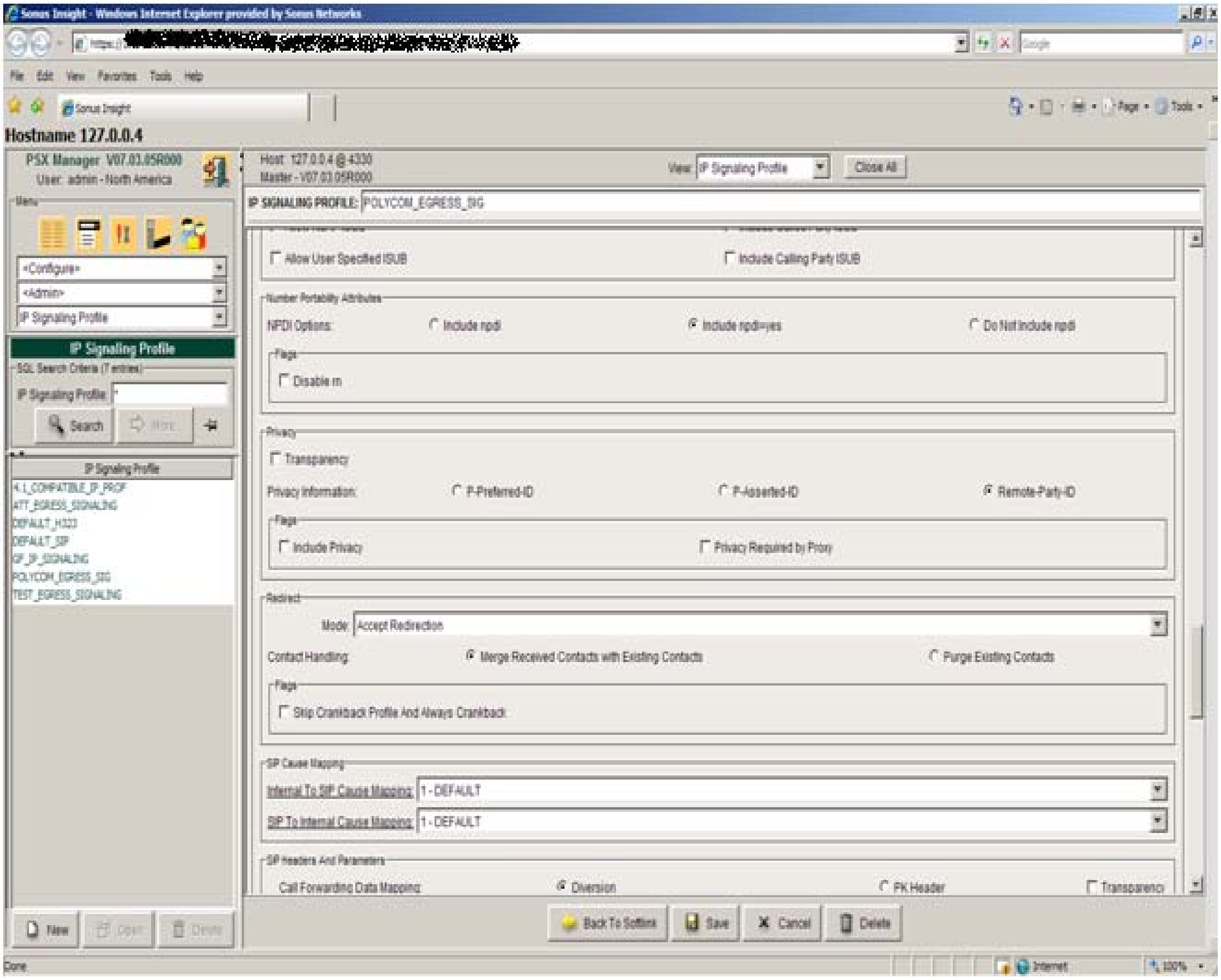
Transparancy Flags set: From Header, SIP Body, SIPFRAG Body and To Header.



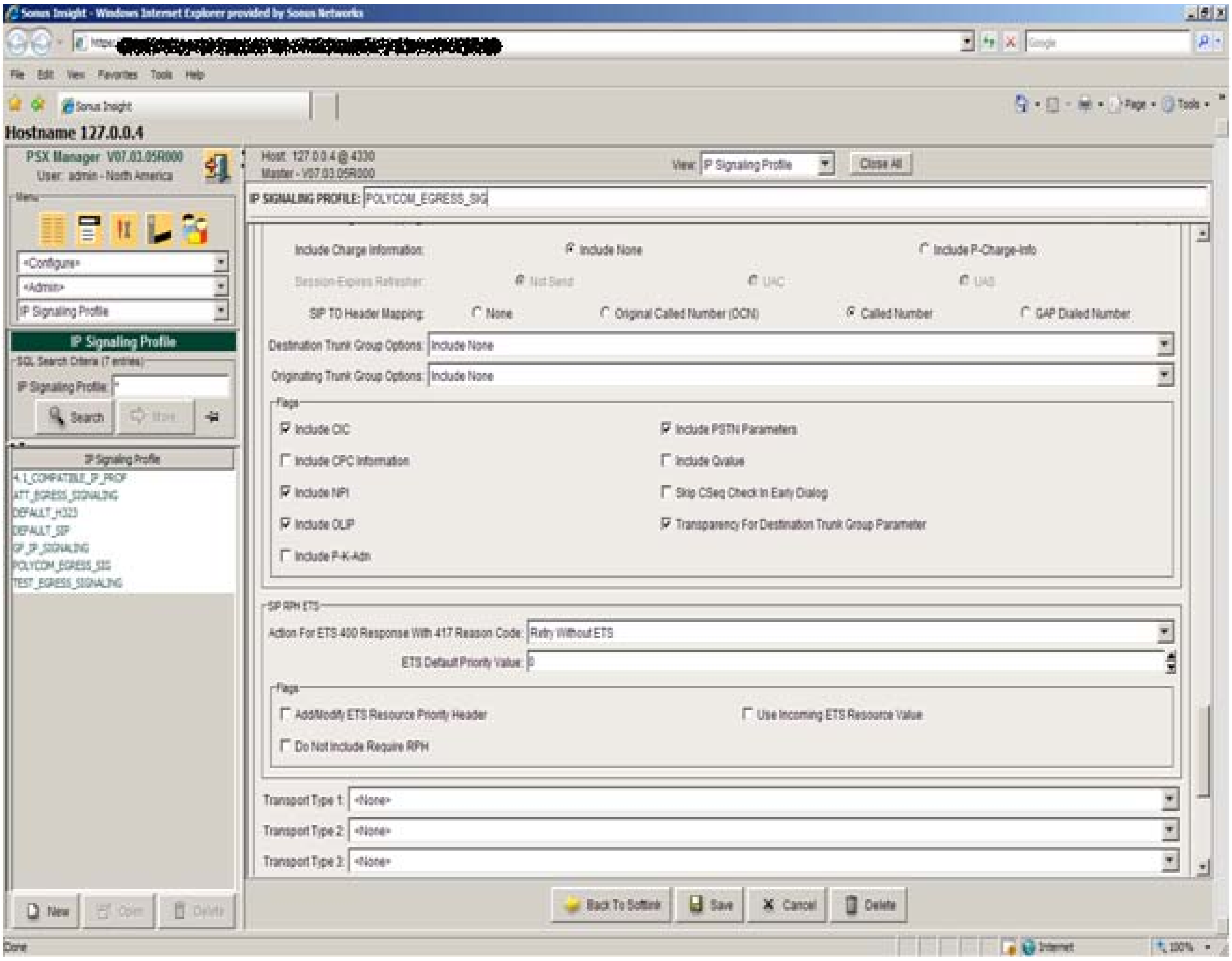
Screenshot 3 / 7



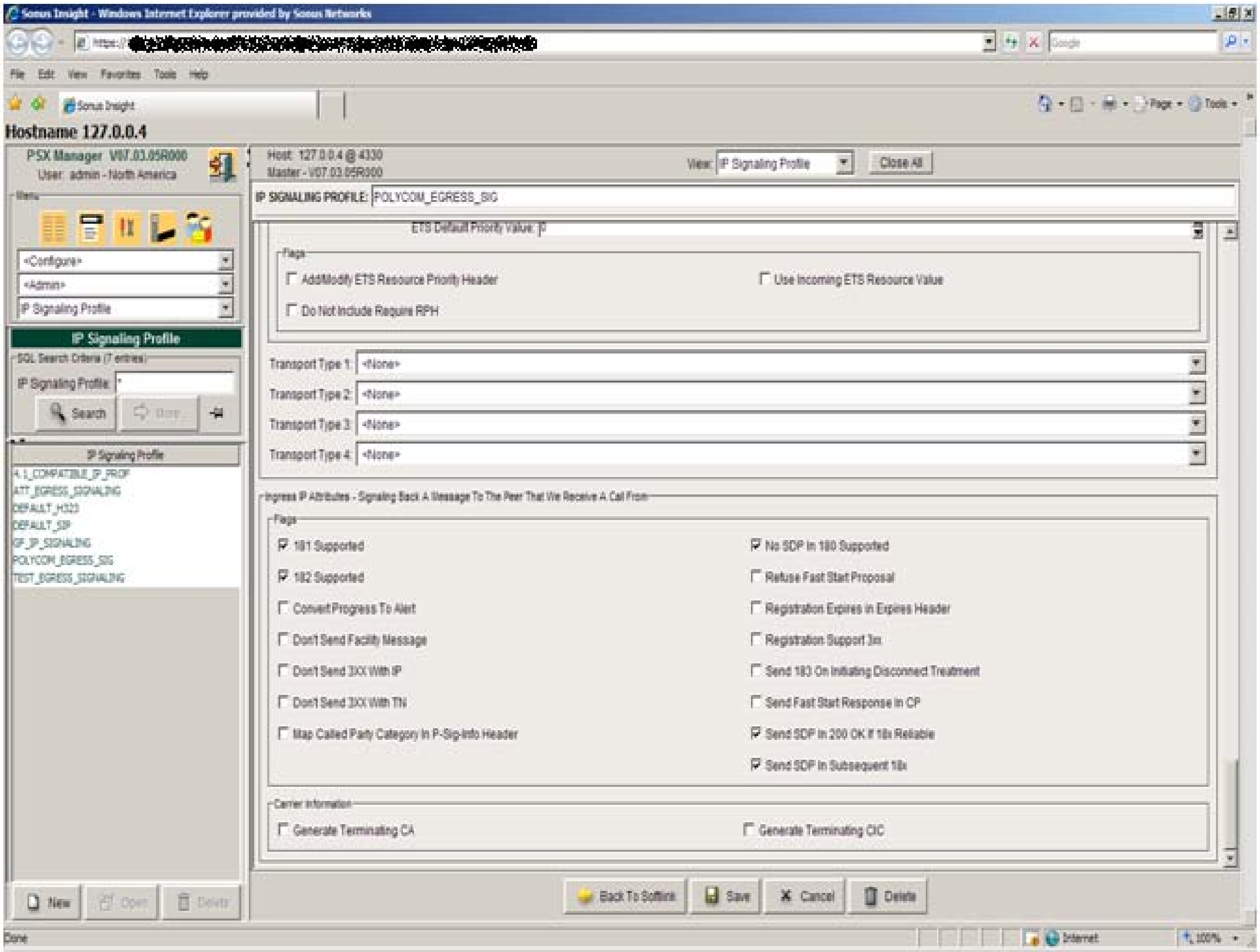
Screenshot 4 / 7



Screenshot 5 / 7



Screenshot 6 / 7

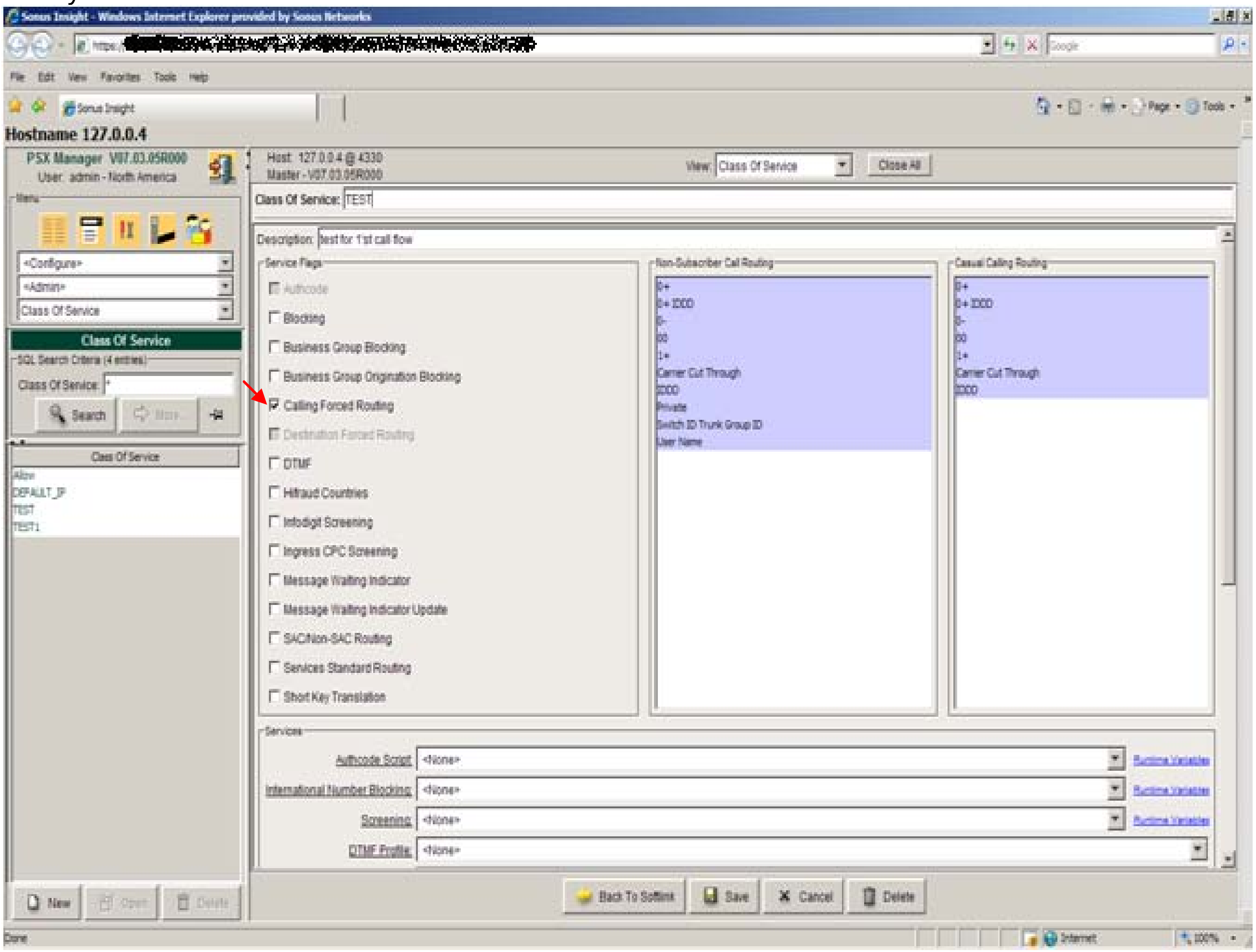


Screenshot 7 / 7

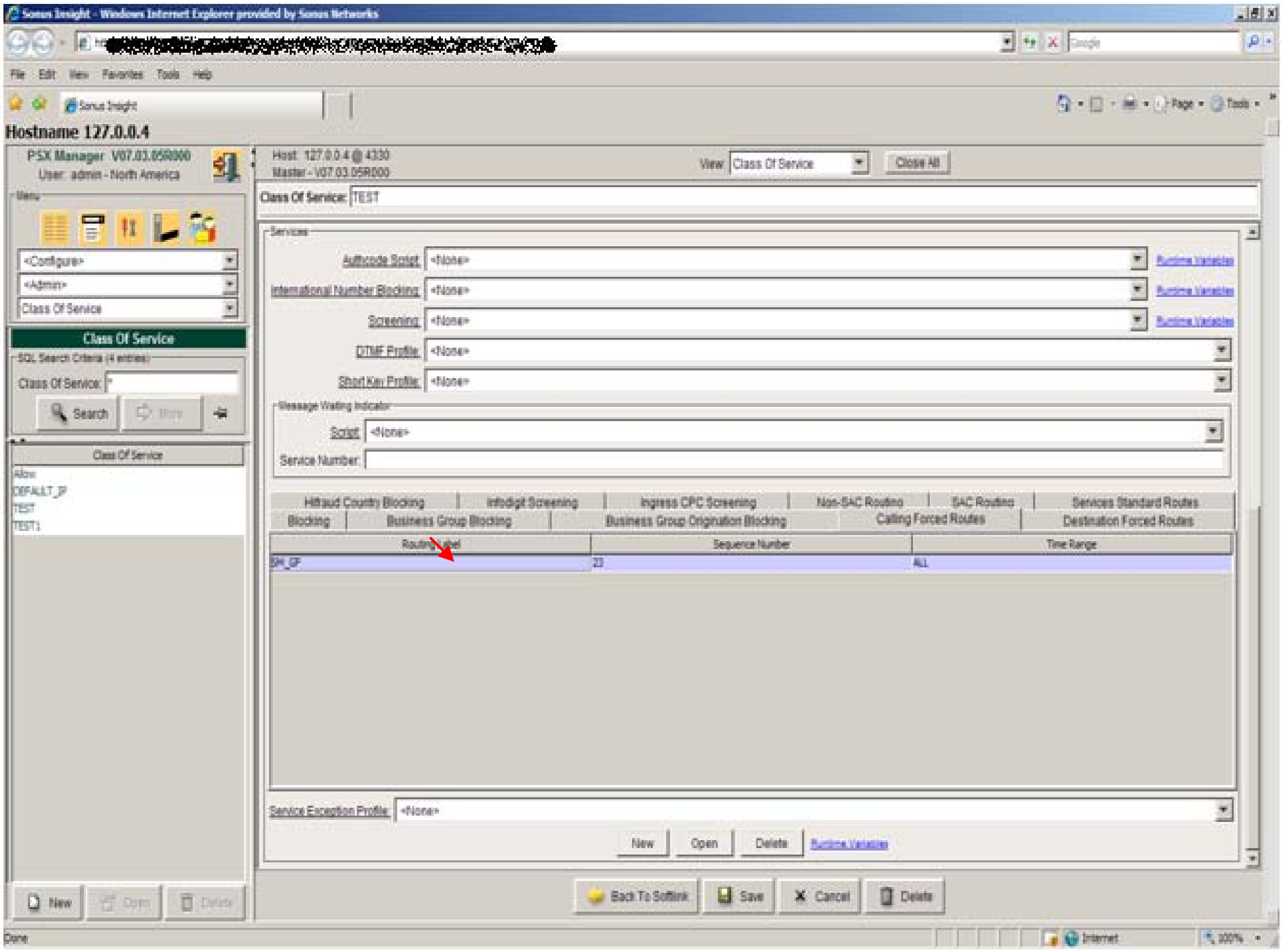
Step 4: Build Class of Service with Calling Forced Routing to Genesys

Note: Configuration is similar to Carrier PSX COS

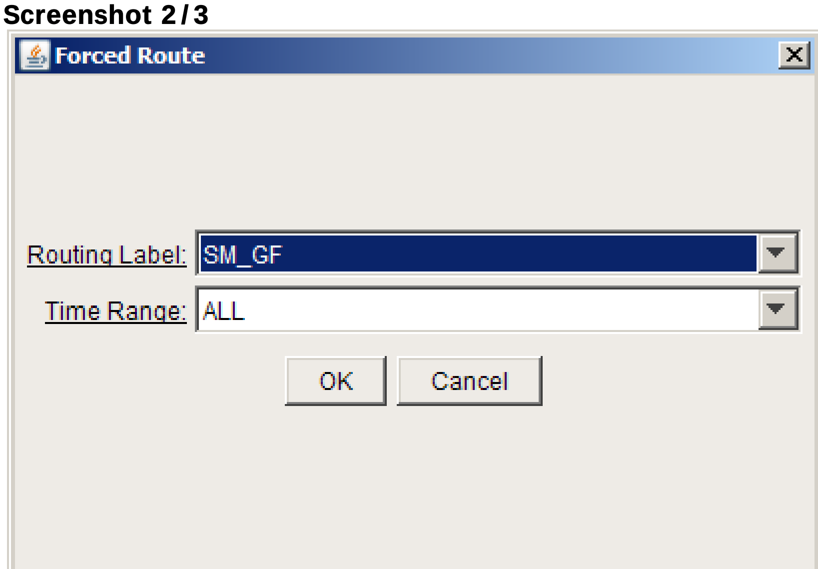
Any SIP end points that register to SIP Server via NBS, have their REGISTRATIONS use a forced Route to Genesys SIP Server.



Screenshot 1 / 3

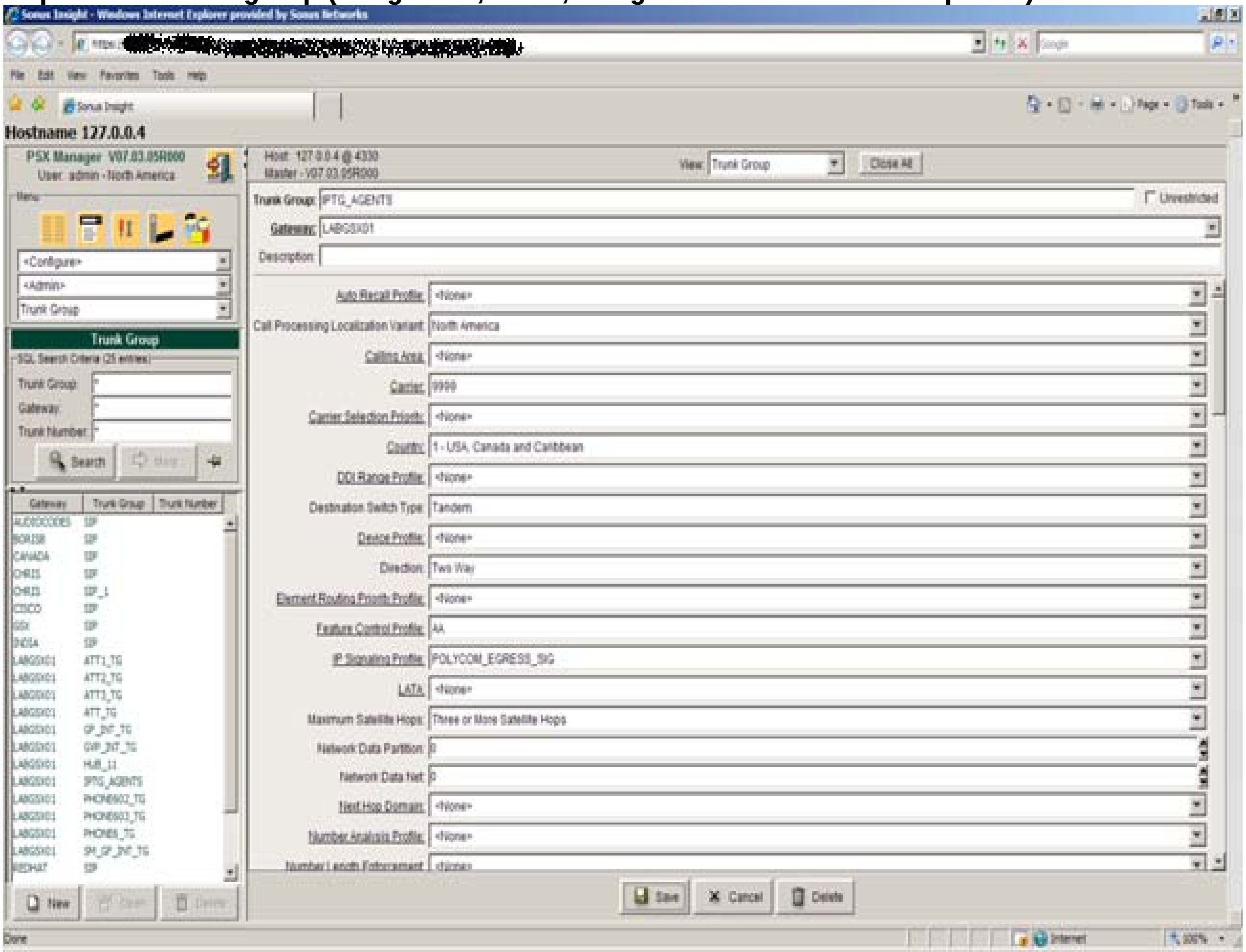


Screenshot 2 / 3

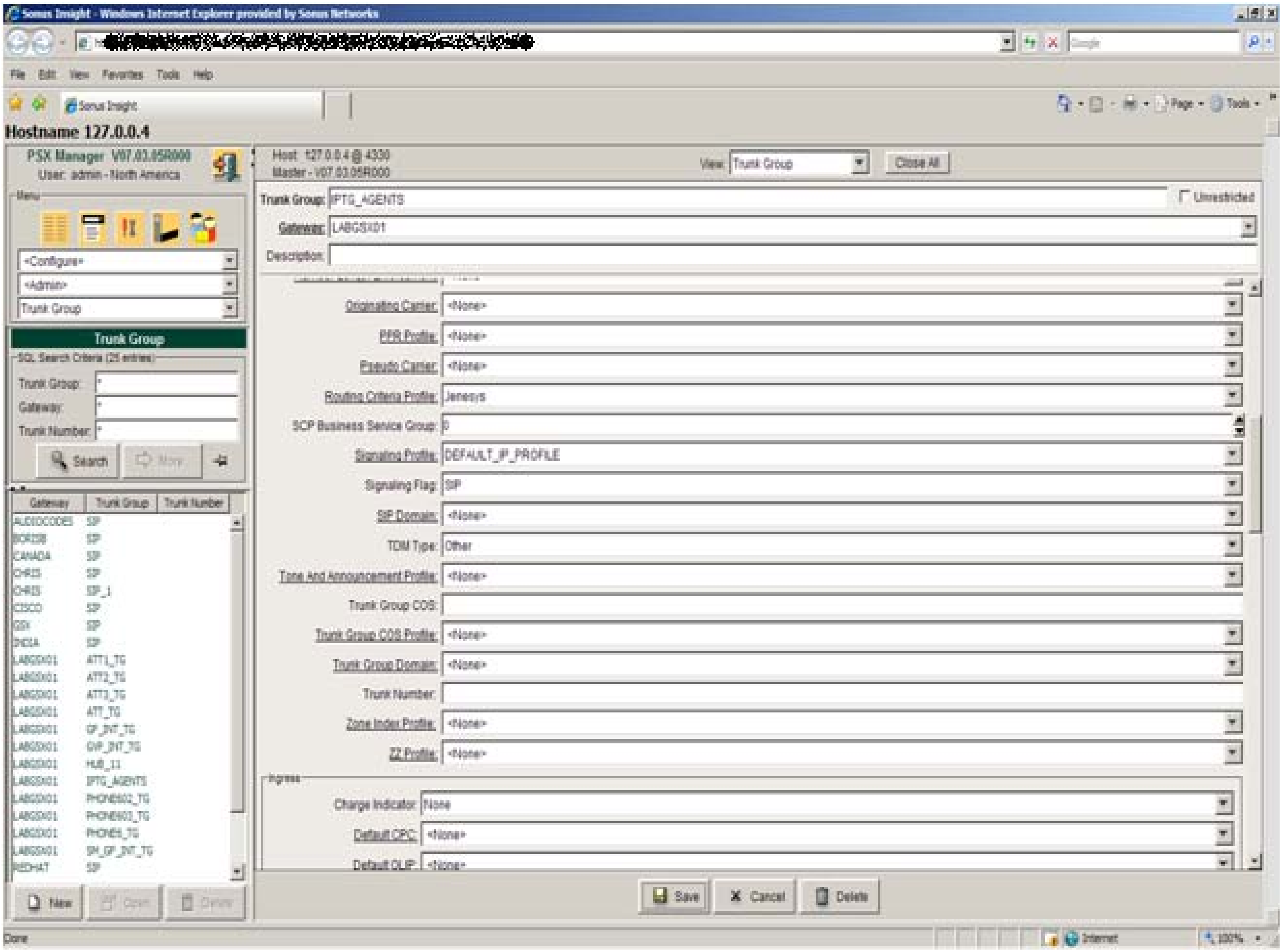


Screenshot 3 / 3

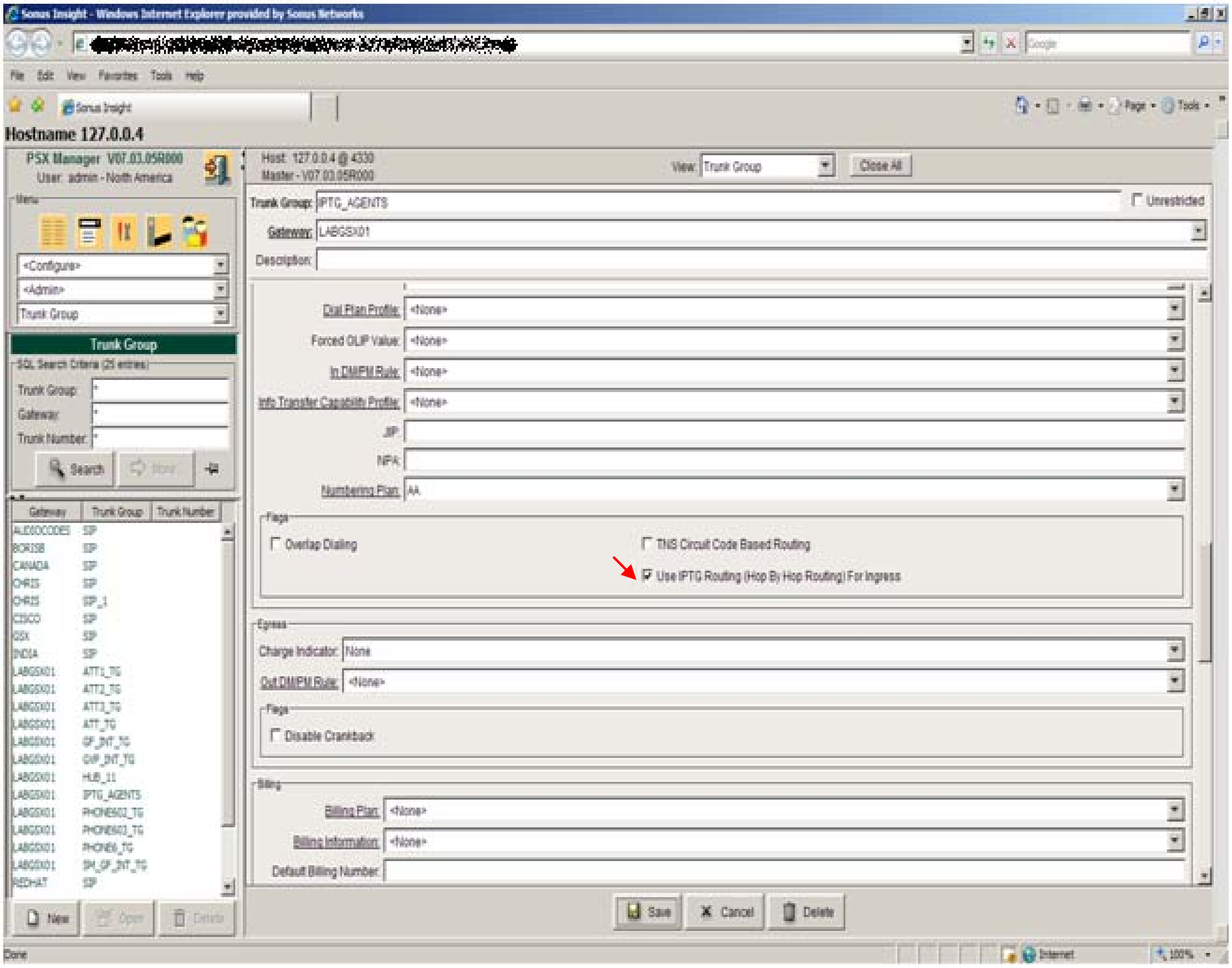
Step 5: Build Trunkgroup (assign PSP, IPSP ,IP Sig Peer and COS from Steps 1-4)



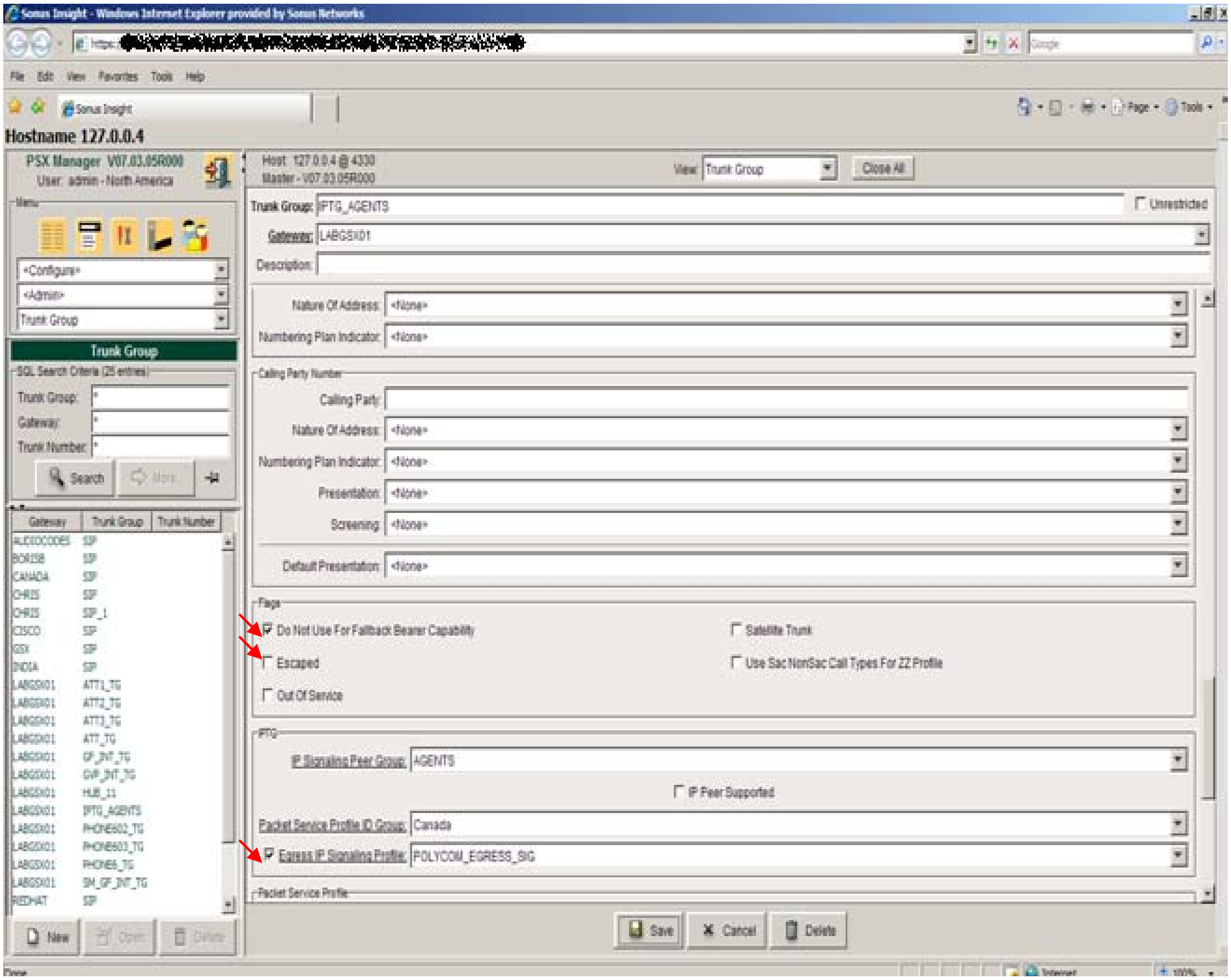
Screenshot 1/5



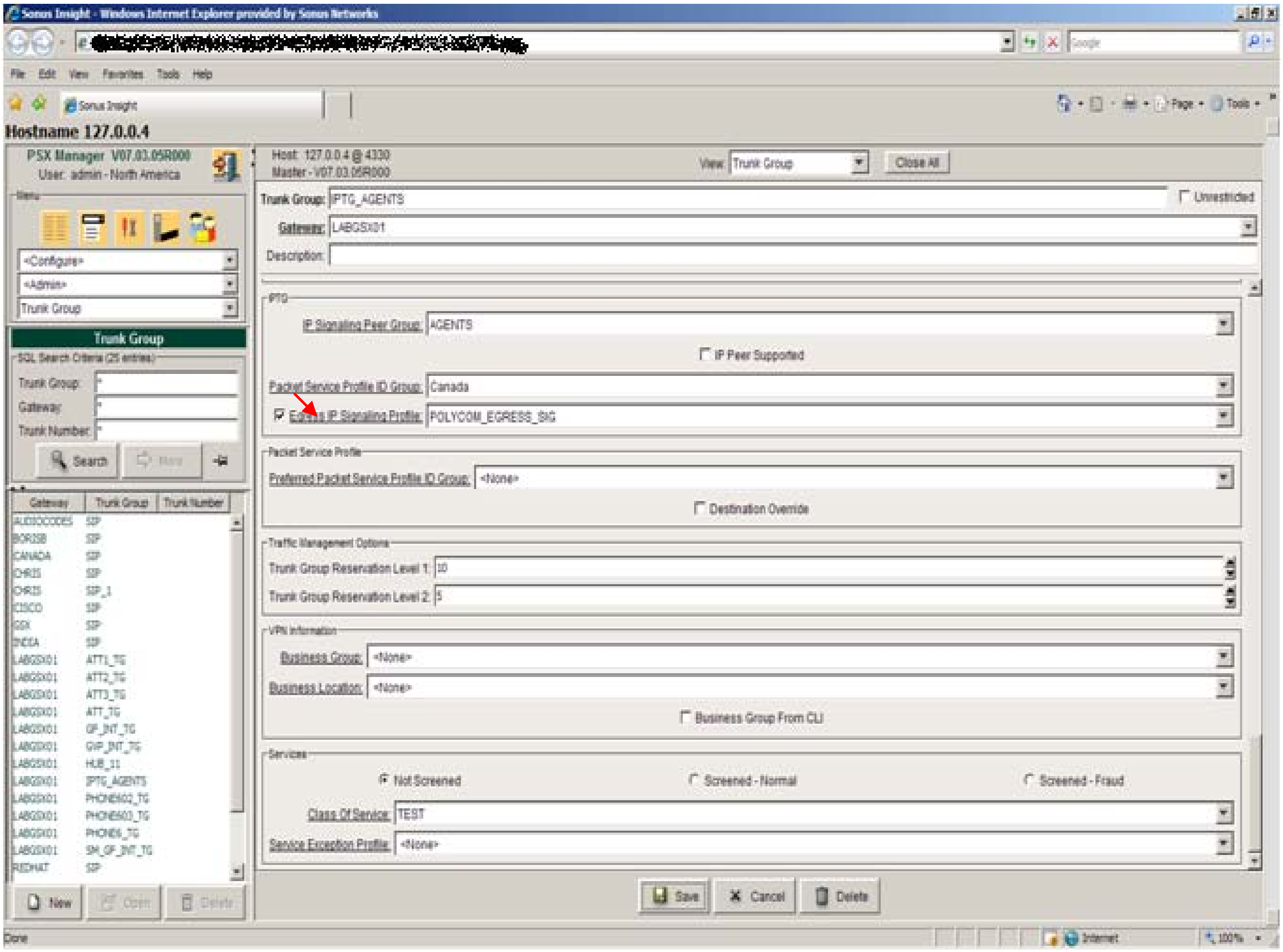
Screenshot 2 / 5



Screenshot 3 / 5



Screenshot 4 / 5



Screenshot 5 / 5

4. Final Notes

The interoperability testing between GVP 8.1.4, Sonus NBS 7.3.5 and AT&T IPTF and IPXC service was certified successfully.

The SIP end point used for the testing was Polycom SoundPoint IP 550.