GENBAND SBC and GENBAND AS
Configuration Guide
For Use with AT&T's
IP Flexible Reach -Enhanced Features Service
Using MIS, PNT or AT&T Virtual Private Network

Version 1.0 / Issue 1
1 September 2015

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1 Introduction

This configuration guide describes how to configure GENBAND SBC and GENBAND AS for connectivity to AT&T's IP Flexible Reach service. Testing was performed in accordance with the test plan for the AT&T IPFR trunking. While not detailing the results of the testing performed this configuration guide provides the essential configurations required for SIP interoperability with GENBAND and the AT&T IP Flexible Reach service using AT&T Virtual Private Network (AVPN) transport.

2 Special Notes

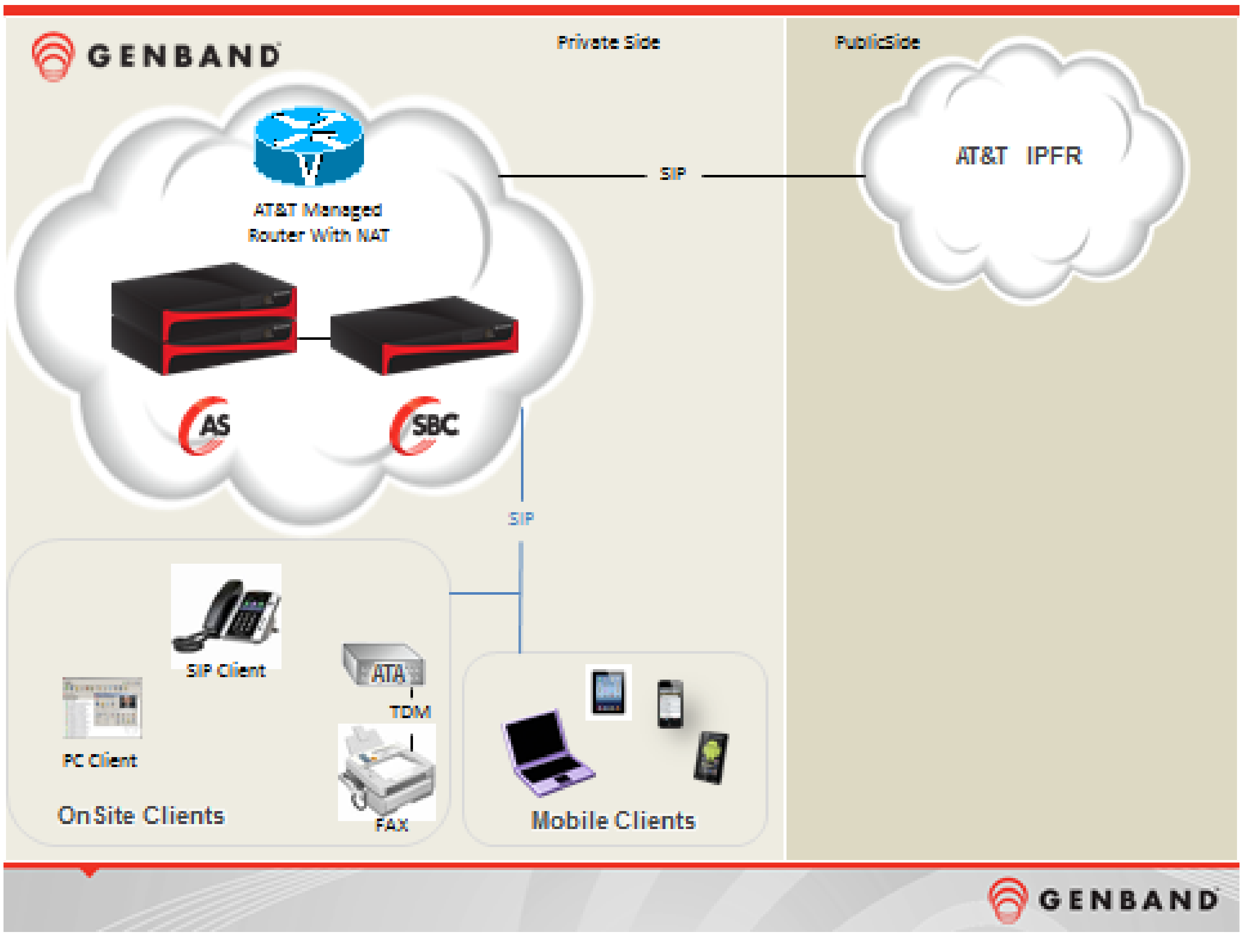
- AT&T requires RTP ports 16384-32767 be used for media (these are configured in the GENBAND SBC used for AT&T trunk connectivity)
- Preferred configuration is to handle SIP Refer messages on the GENBAND AS. This is accomplished by setting the SIP Profile for the Node associated to the AT&T trunk.
- "Use Calling Party as From" and "Add Diversion Header" should be checked in the SIP Profile associated to the AT&T trunk.
- "p-called-party-id" header should be removed from the SIP Profile associated to the AT&T trunk.
- Charge ID needs to be one of the DIDs provided by AT&T. AT&T uses a screening service for access to its network, unless unscreened ANI is requested. They require an authorized DID be in the SIP From, P-Aid, Remote Party ID, or Diversion header

Emergency 911/E911 Services Limitations and Restrictions - Although AT&T provides 911/E911 calling capabilities, AT&T does not warrant or represent that the equipment and software (e.g., IP PBX) reviewed in this customer configuration guide will properly operate with AT&T IP Flexible Reach to complete 911/E911 calls; therefore, it is Customer's responsibility to ensure proper operation with its equipment/software vendor.

While AT&T IP Flexible Reach services support E911/911 calling capabilities under certain Calling Plans, there are circumstances when that E911/911 service may not be available, as stated in the Service Guide for AT&T IP Flexible Reach found at http://new.serviceguide.att.com. Such circumstances include, but are not limited to, relocation of the end user's CPE, use of a non-native or virtual telephone number, failure in the broadband connection, loss of electrical power, and delays that may occur in updating the Customer's location in the automatic location information database. Please review the AT&T IP Flexible Reach Service Guide in detail to understand the limitations and restrictions.

3 Overview

With GENBAND SBC and AS using AT&T IPFR trunking service you can replace your traditional trunking with VoIP. Both voice and fax (G.711 and T.38) calls are supported along with auto attendant, voicemail , and Meet-Me conferencing.



- GENBAND SBC (Q10/Q20), version 8.3.8.7
- GENBAND AS, version 10.4, patch 17.0.22.6
- GENBAND GENCom Client, version 10.3.1378
- Polycom VVX series SIP terminals, firmware 5.2.0.8330

4 Configuration Guide

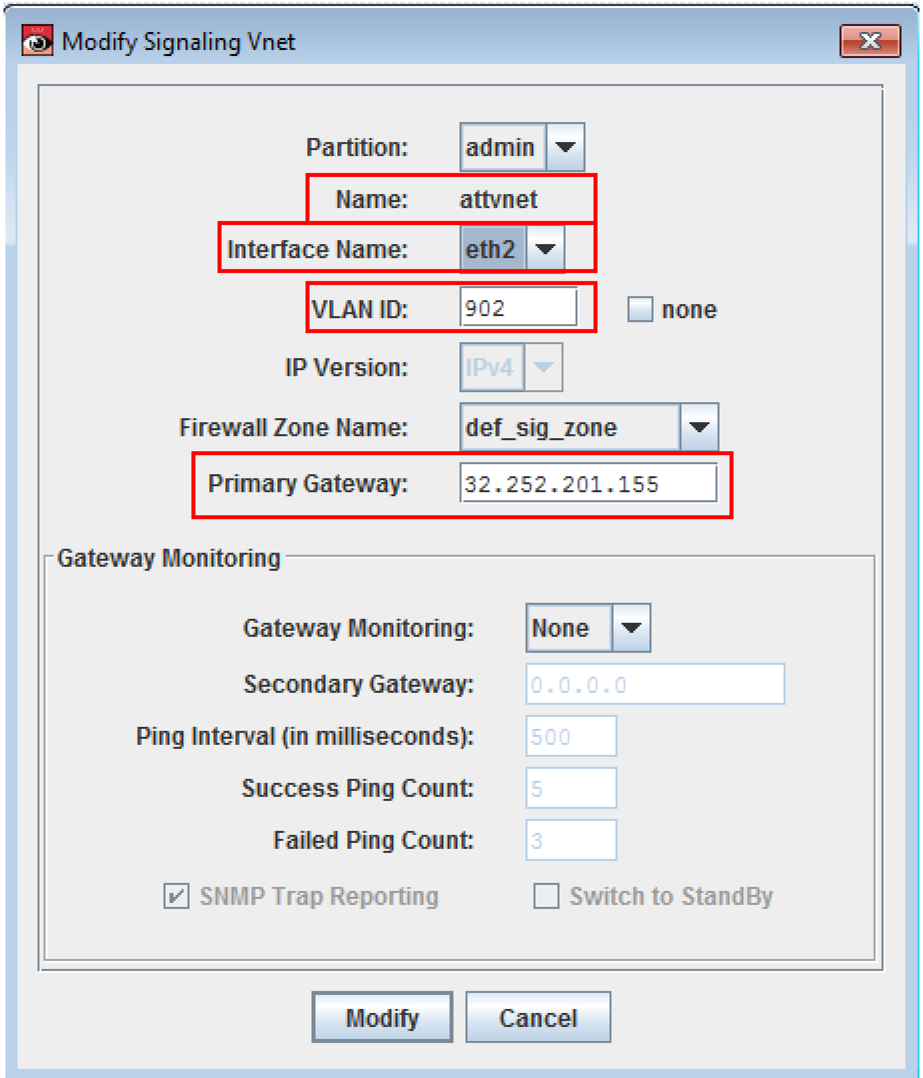
This guide assumes that you have a functioning GENBAND SBC and AS. The starting point is configuration of the link to AT&T IPFR service. Also proper knowledge of GENBAND SBC and GENBAND AS administration and configuration are required.

4.1 SBC Configuration

From the SBC cli verify the version and the following 2 global parameters are set.

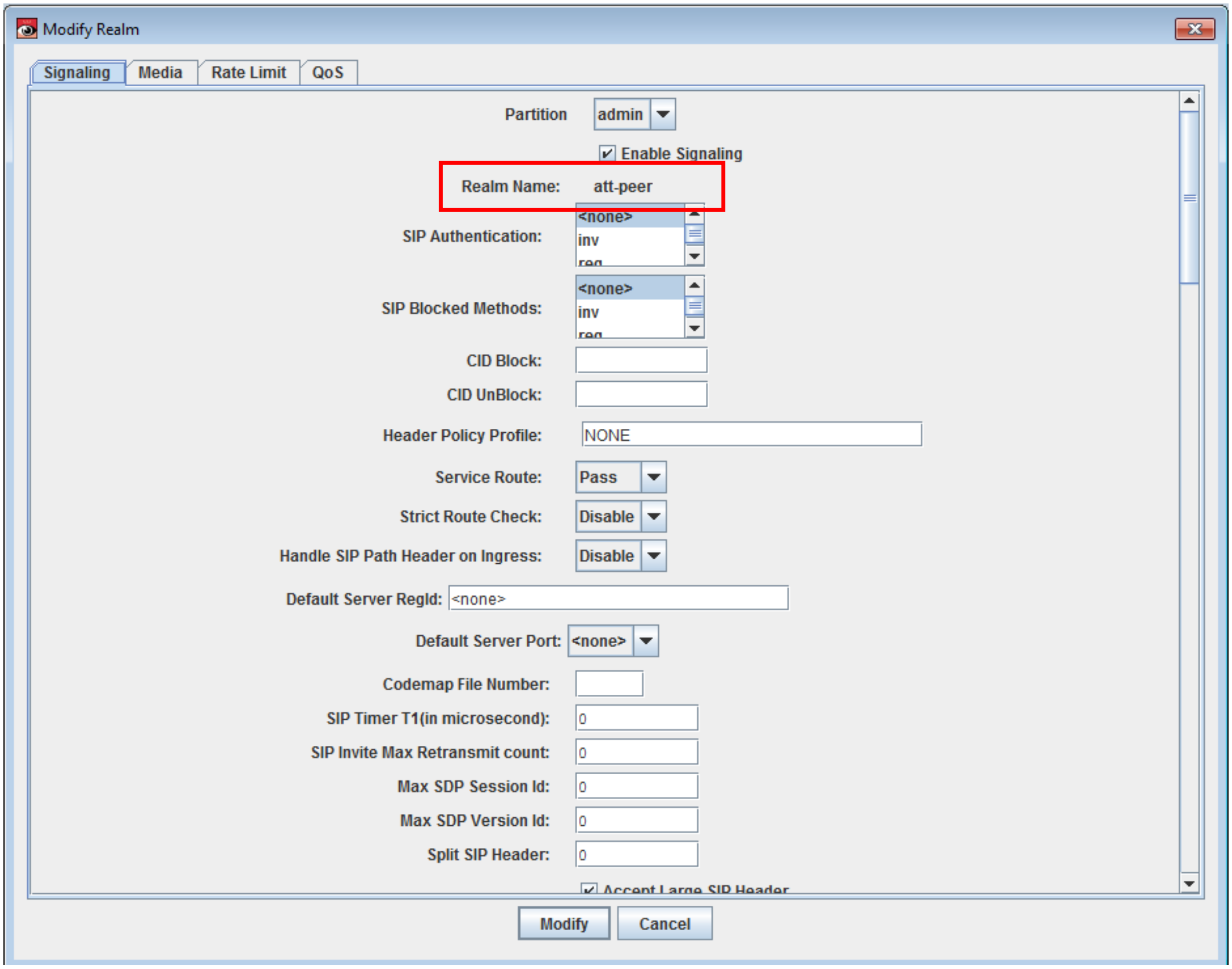
- # gis –v
  GENBAND GIS Directory Server v8.3.8.7, 05-12-2015
  Copyright (c) 1998-2015 GENBAND Inc.
- # nxconfig.pl –S
  gis calls hide-src-fqdn 1
  gis sip replace-history-host-with-rsa 1

Open the RSM console and Add a Signaling VNet

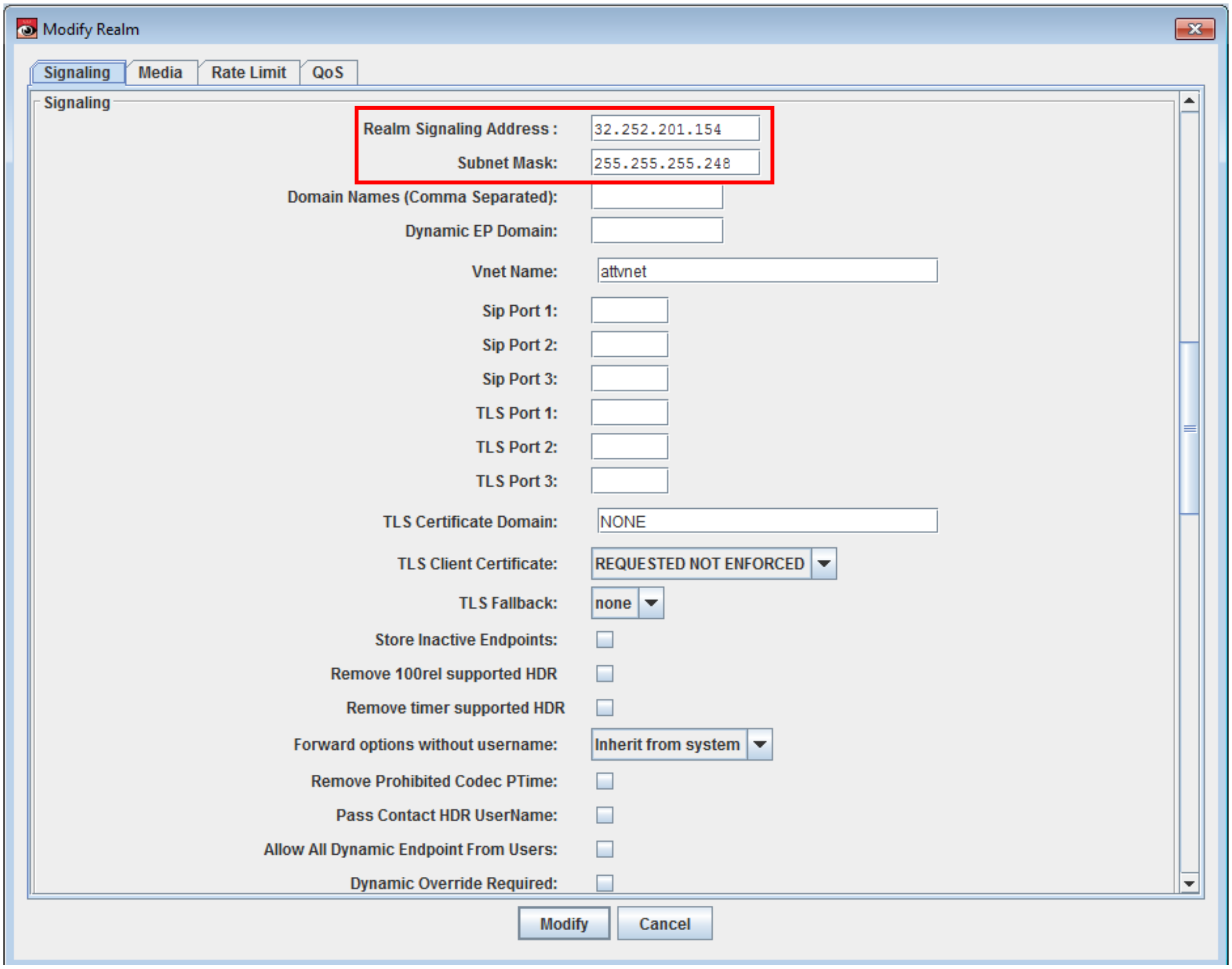


Add a Name, select the Interface Name, VLAN ID, and Primary Gateway.

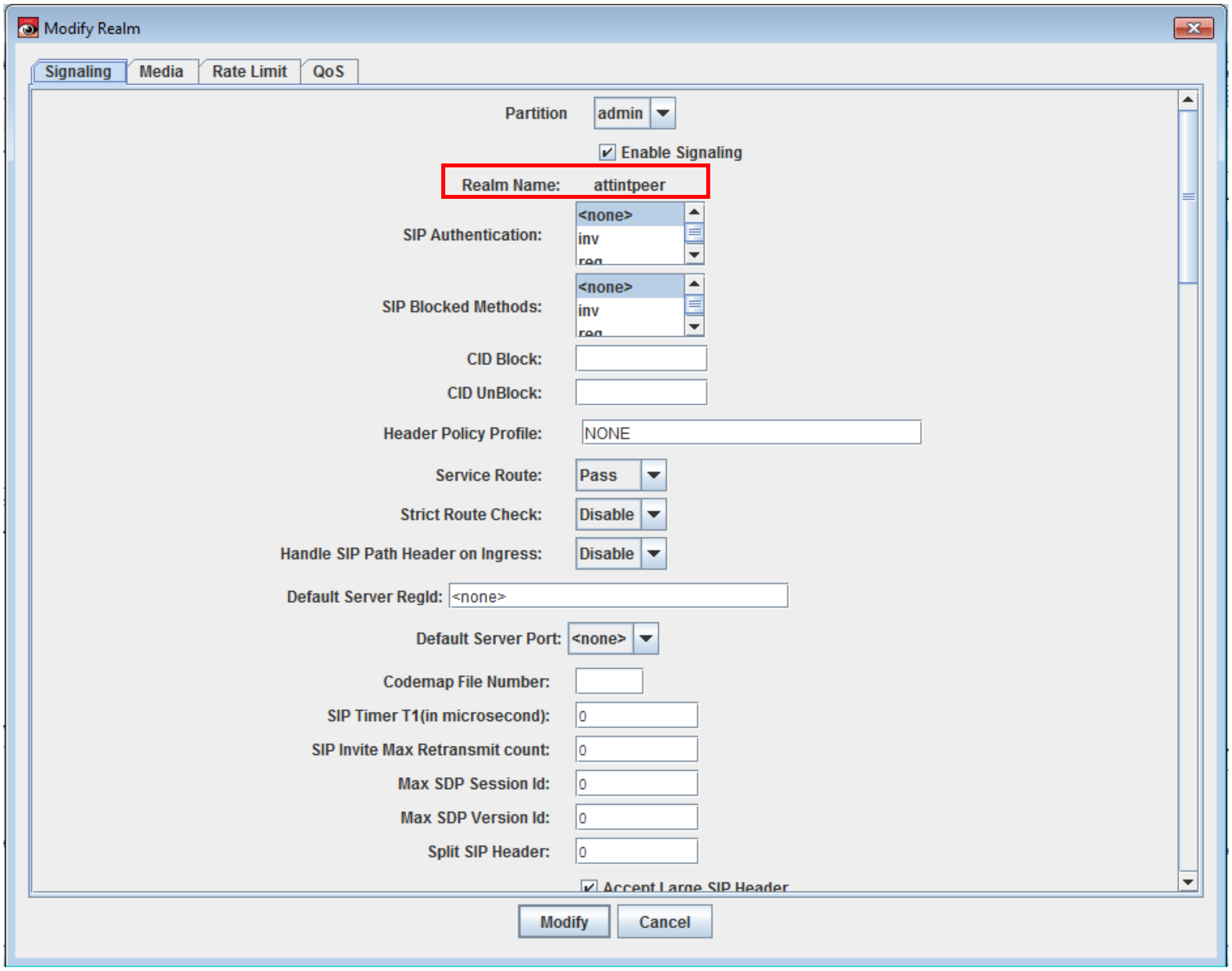
Next build the Realms for both public and private interface. The public will interface to the AT&T equipment while the private will interface to the GENBAND AS>

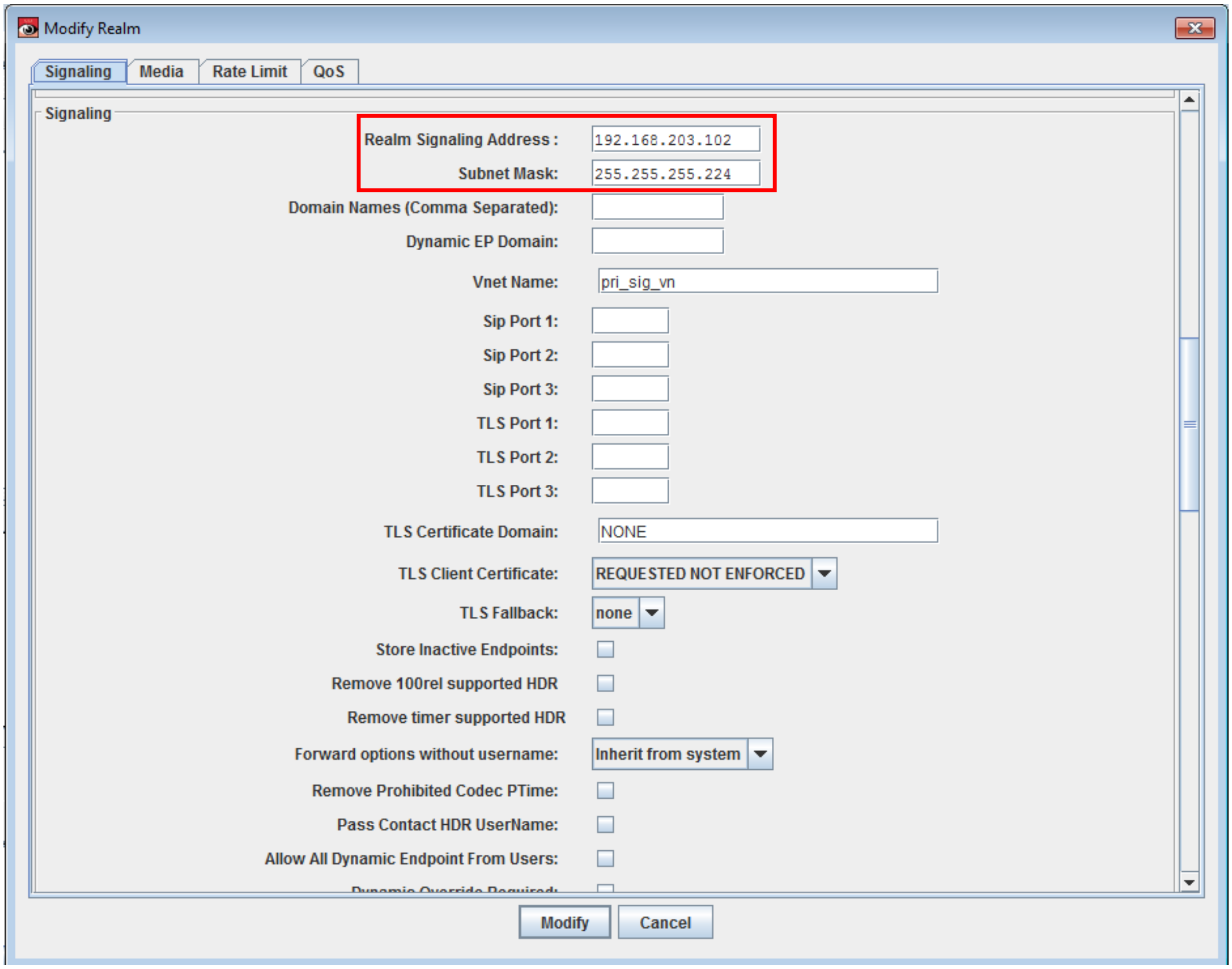


Enter a Realm Name.

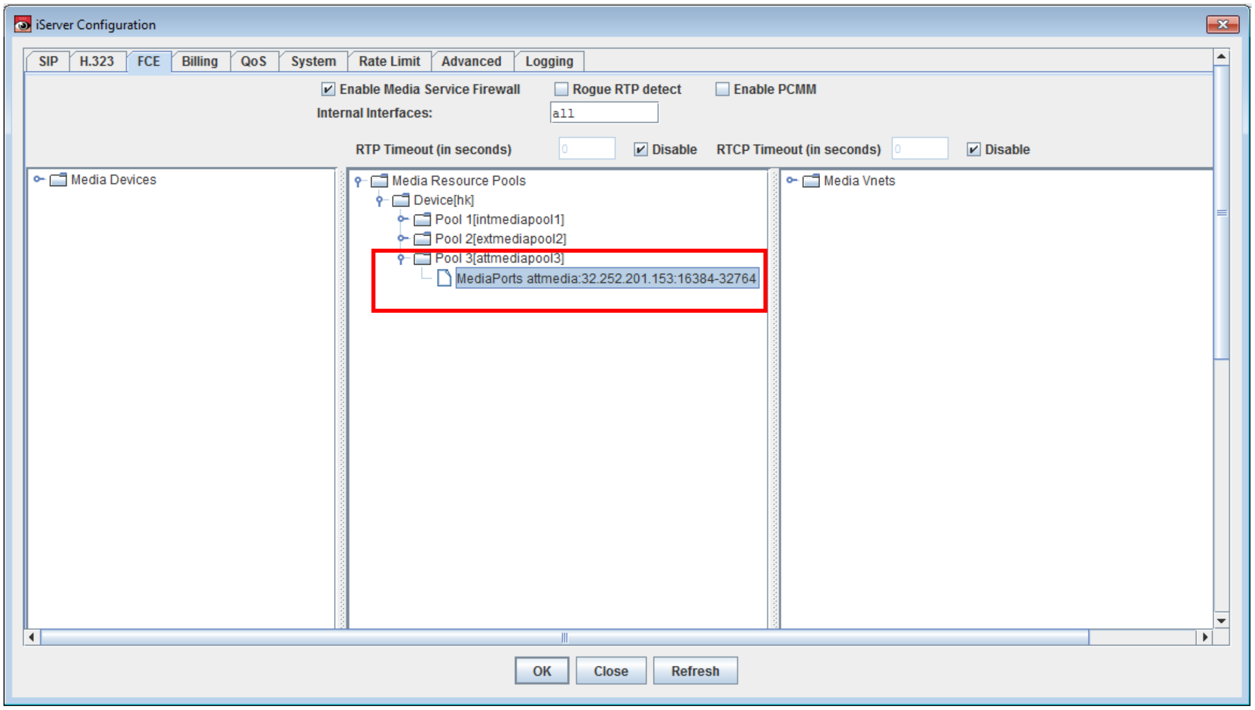


Configure the Realm Signaling Address and the Subnet Mask


Modify Realm
Signaling
Media
Rate Limit
QoS
Partition
admin
Enable Signaling
Realm Name:
attintpeer
SIP Authentication:
<none>
inv
SIP Blocked Methods:
<none>
inv
CID Block:
CID UnBlock:
Header Policy Profile:
NONE
Service Route:
Pass
Strict Route Check:
Disable
Handle SIP Path Header on Ingress:
Disable
Default Server RegId:
<none>
Default Server Port:
<none>
Codemap File Number:
SIP Timer T1(in microsecond):
0
SIP Invite Max Retransmit count:
0
Max SDP Session Id:
0
Max SDP Version Id:
0
Split SIP Header:
0
Modify
Cancel

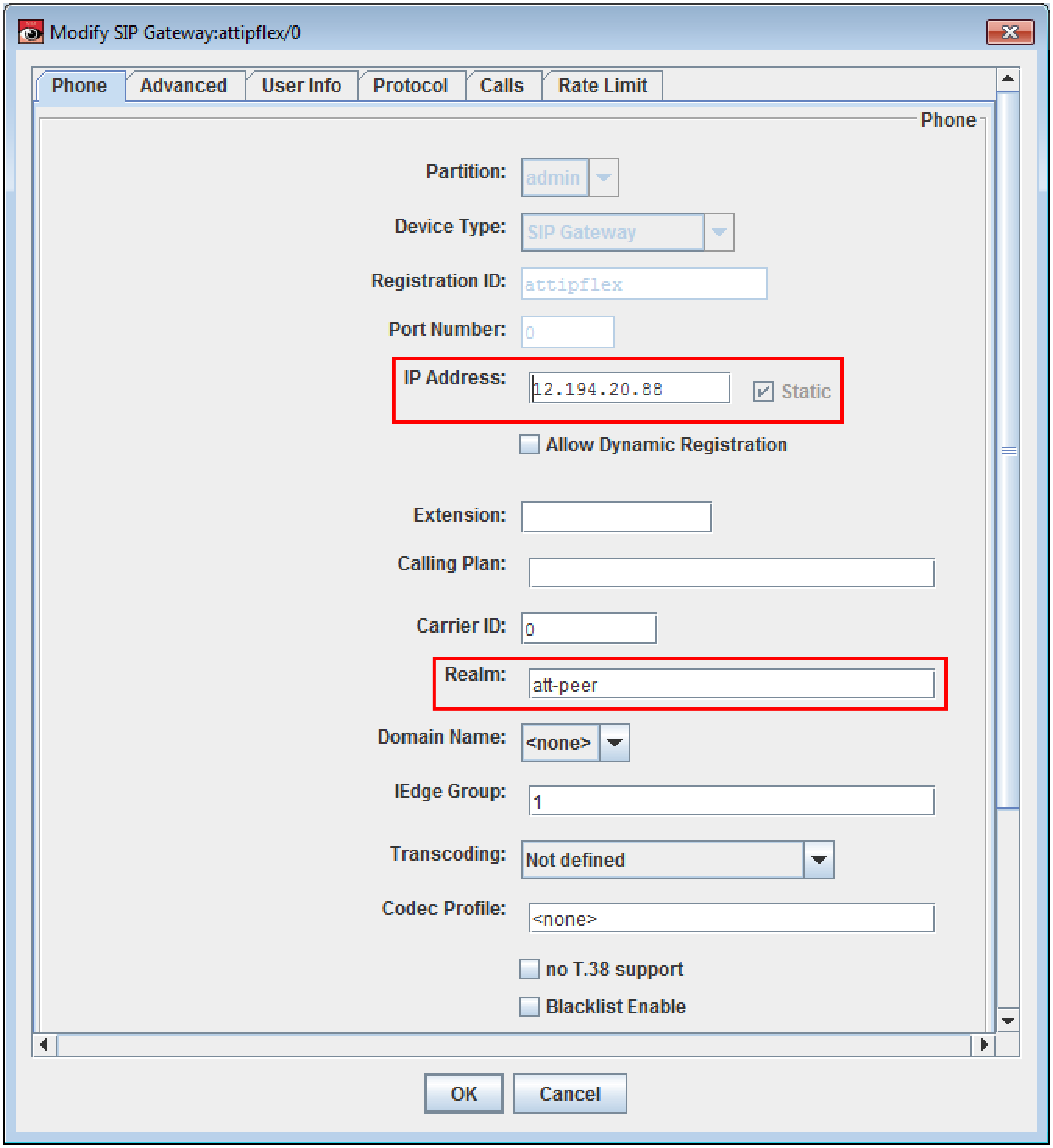

Modify Realm
Signaling
Media
Rate Limit
QoS
Signaling
Realm Signaling Address :
192.168.203.102
Subnet Mask:
255.255.255.224
Domain Names (Comma Separated):
Dynamic EP Domain:
Vnet Name:
pri_sig_vn
Sip Port 1:
Sip Port 2:
Sip Port 3:
TLS Port 1:
TLS Port 2:
TLS Port 3:
TLS Certificate Domain:
NONE
TLS Client Certificate:
REQUESTED NOT ENFORCED
TLS Fallback:
none
Store Inactive Endpoints:
Remove 100rel supported HDR
Remove timer supported HDR
Forward options without username:
Inherit from system
Remove Prohibited Codec PTime:
Pass Contact HDR UserName:
Allow All Dynamic Endpoint From Users:
Modify
Cancel

On the iServer FCE tab the pool for media to AT&T.



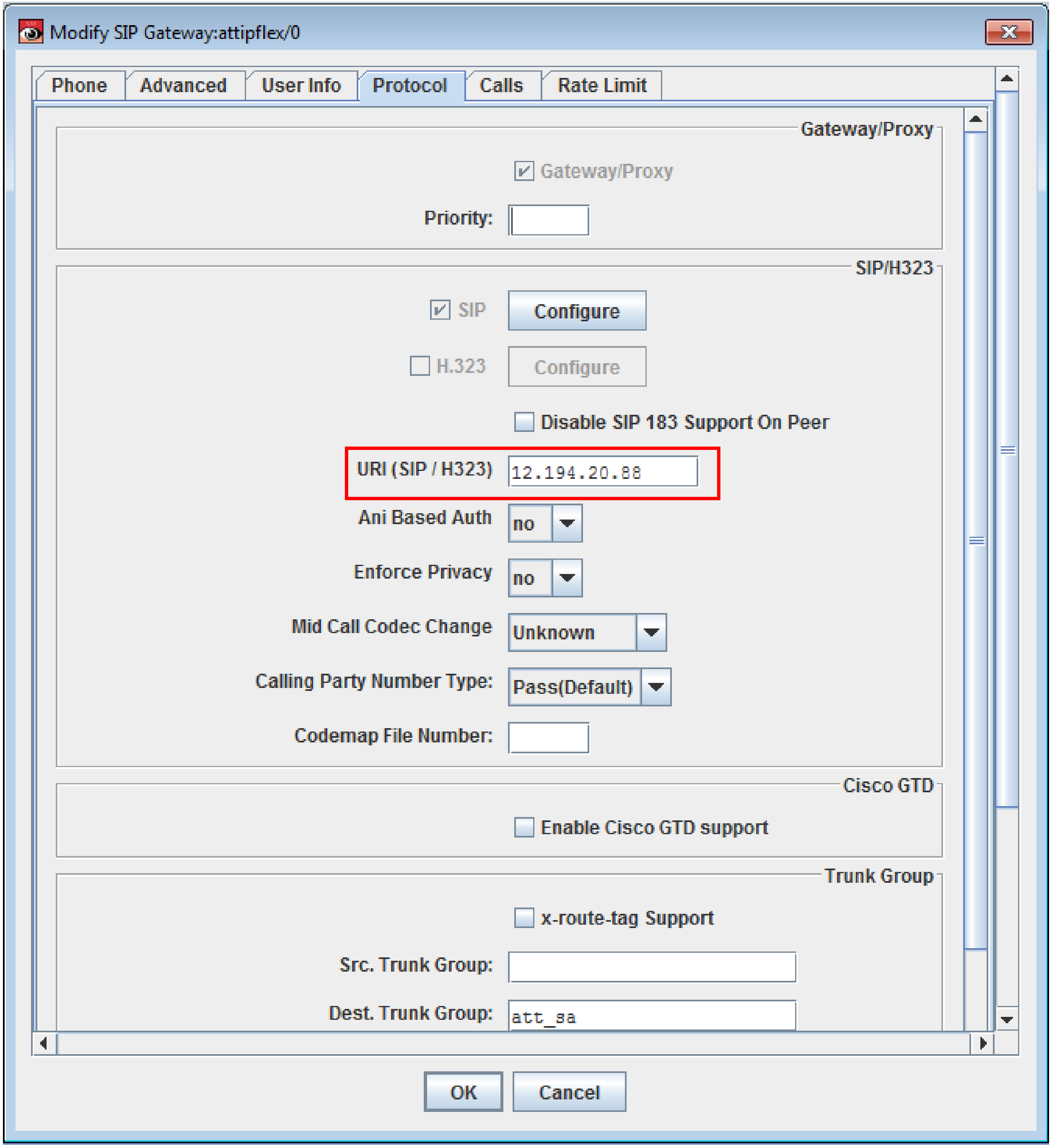
Be sure the media ports are in the range requested by AT&T.

Configure a SIP Gateway Endpoint to AT&T.

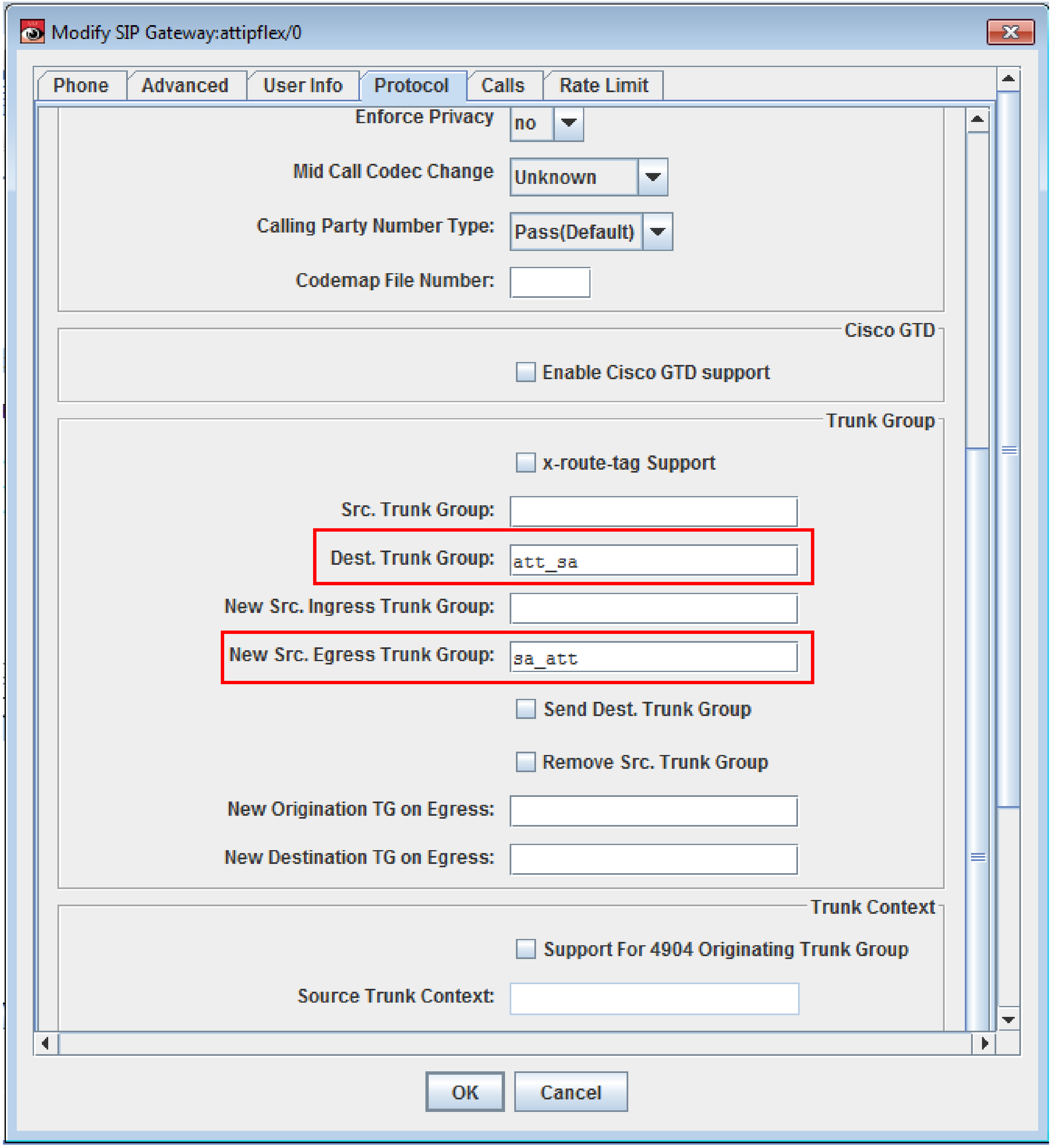


Enter the IP Address for the AT&T SIP Proxy and the Realm you created.

On the Protocol tab.

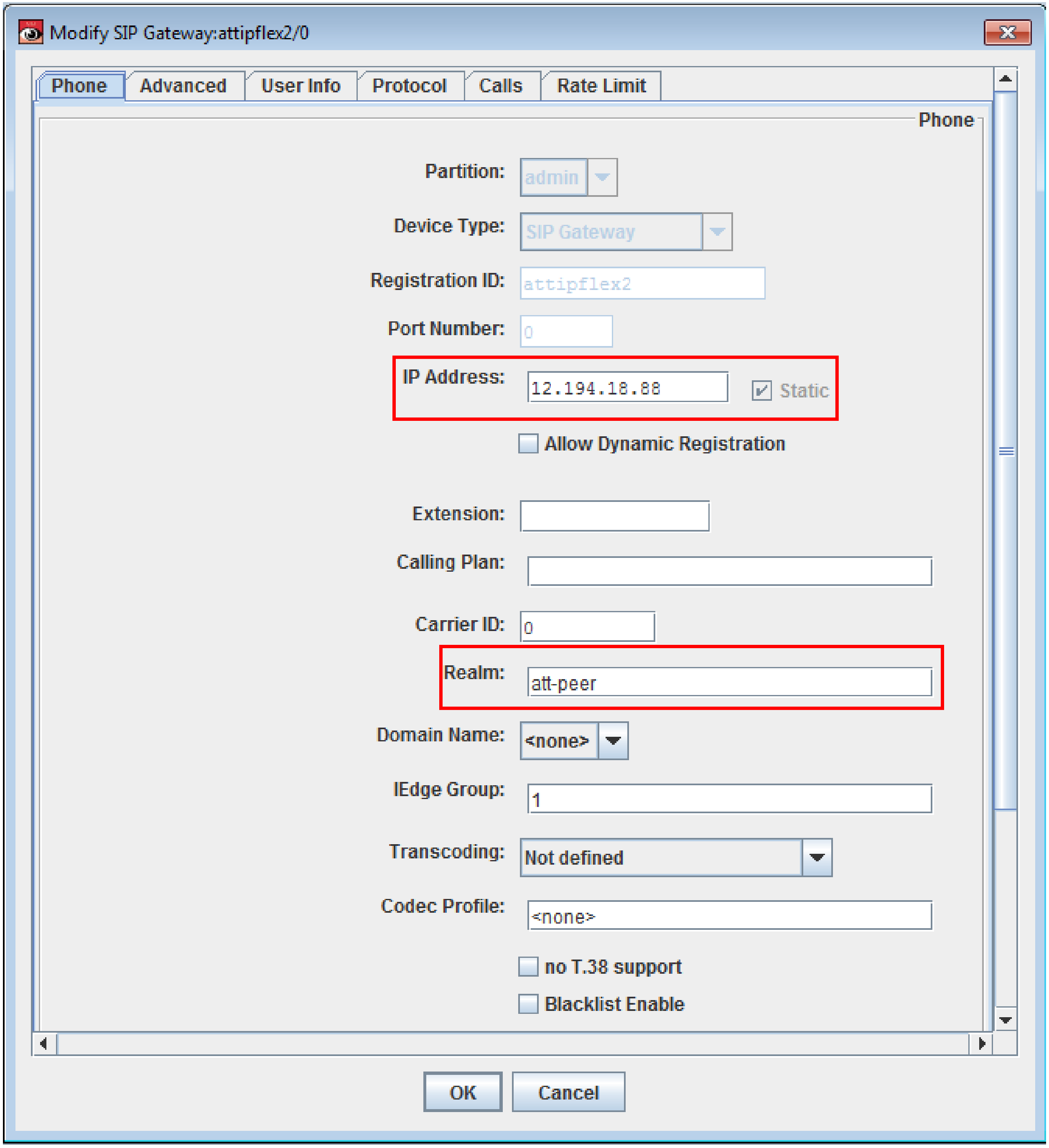


Add the URI.



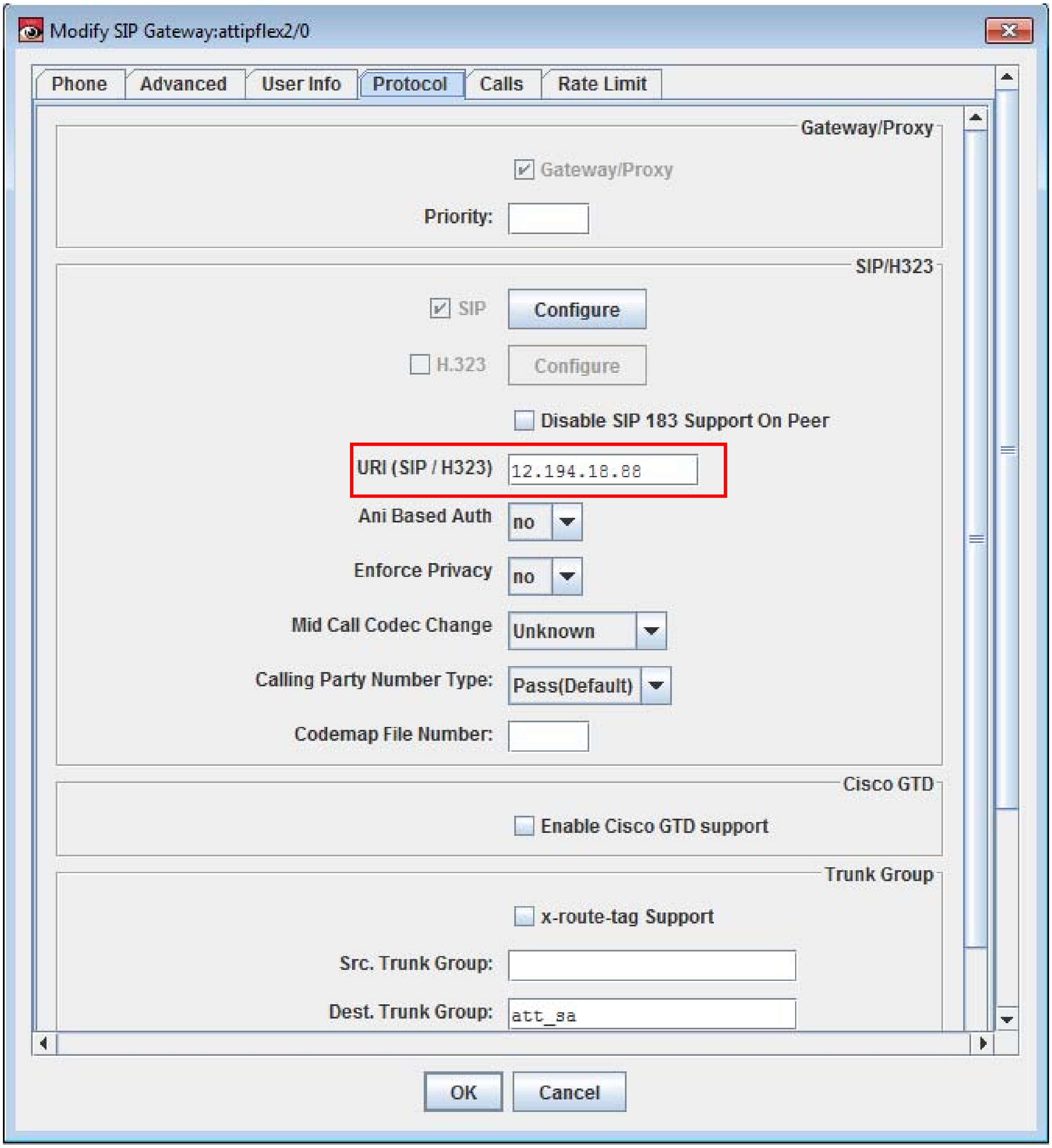
Set the Dest. Trunk Group and New Src. Egress Trunk Group.

Add another Endpoint for the backup proxy.

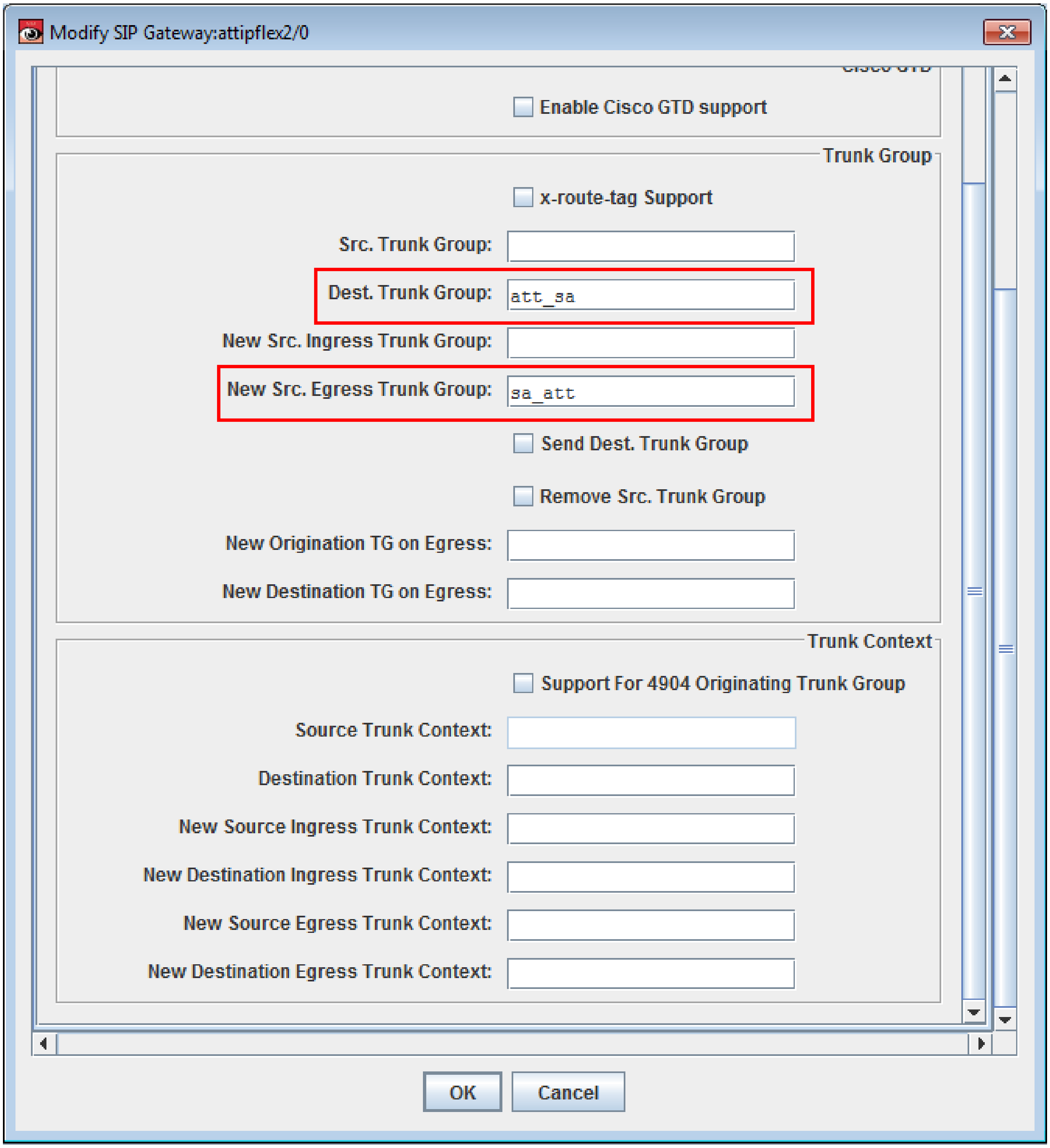


Enter the IP Address for the AT&T SIP Proxy and the Realm.

On the Protocol tab.

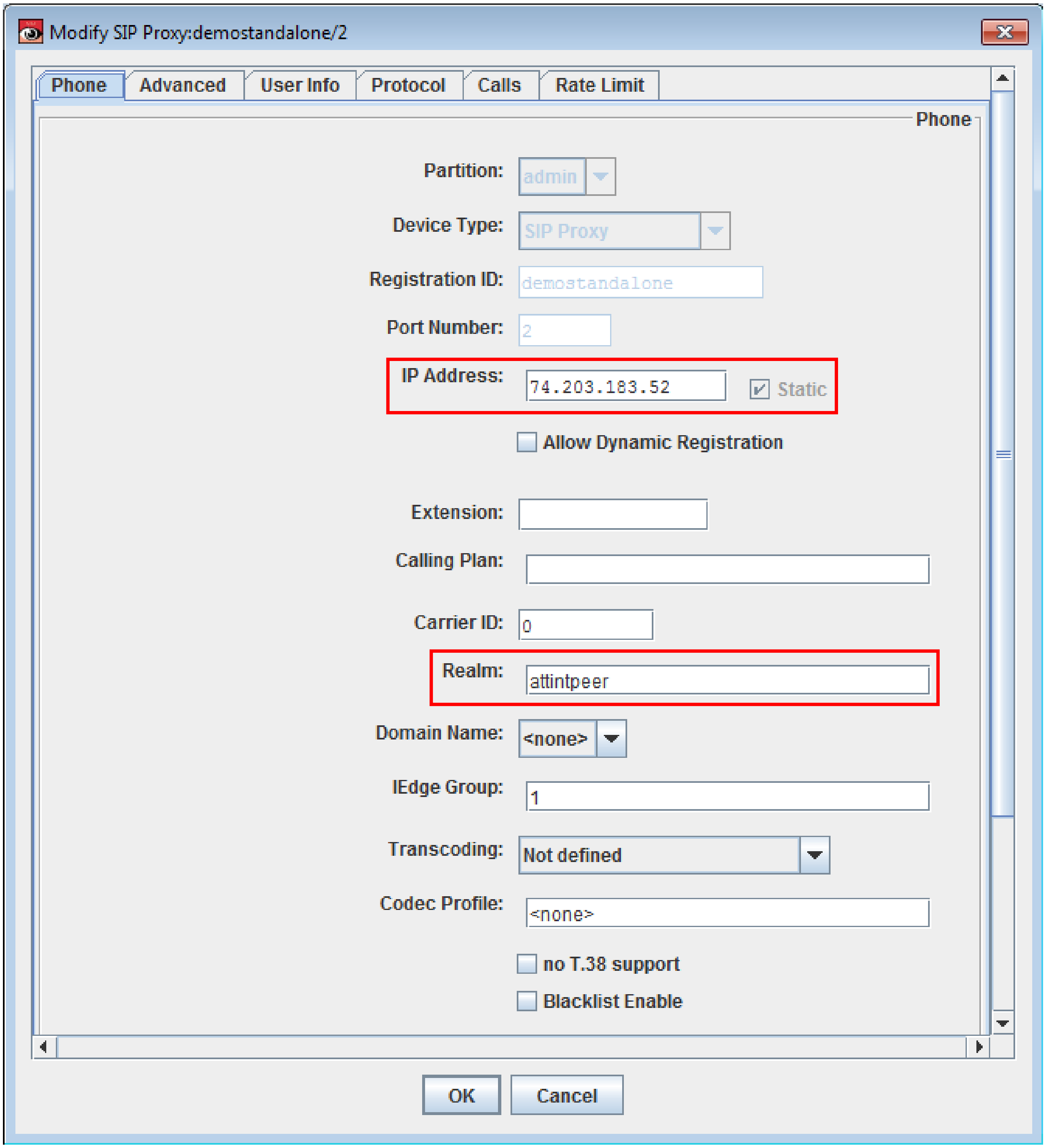


Add the URI.



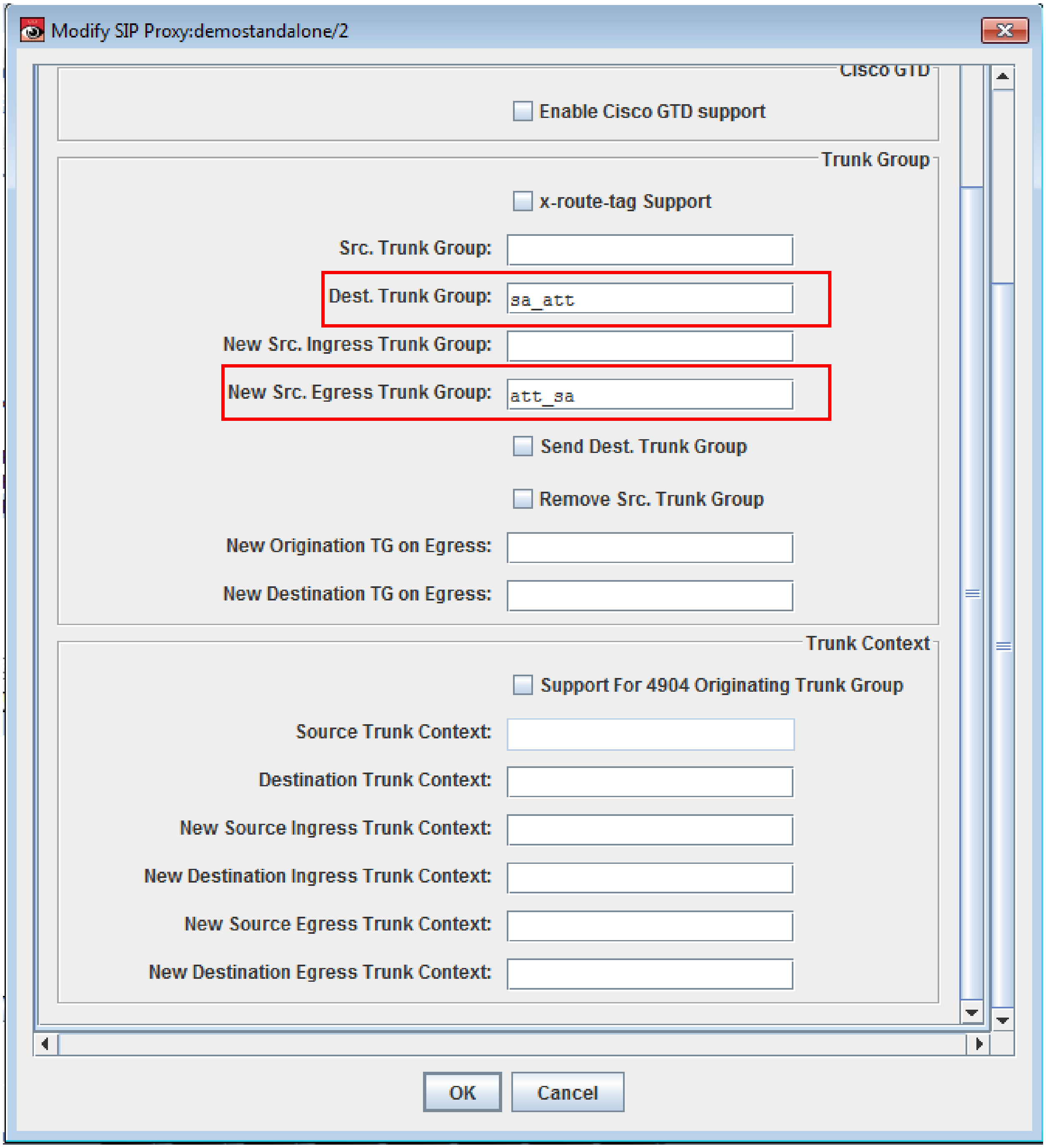
Set the Dest. Trunk Group and New Src. Egress Trunk Group.

Now add the Endpoint to the GENBAD AS



Enter the IP Address for the AS (SESM1ServiceAddr) and the Realm.

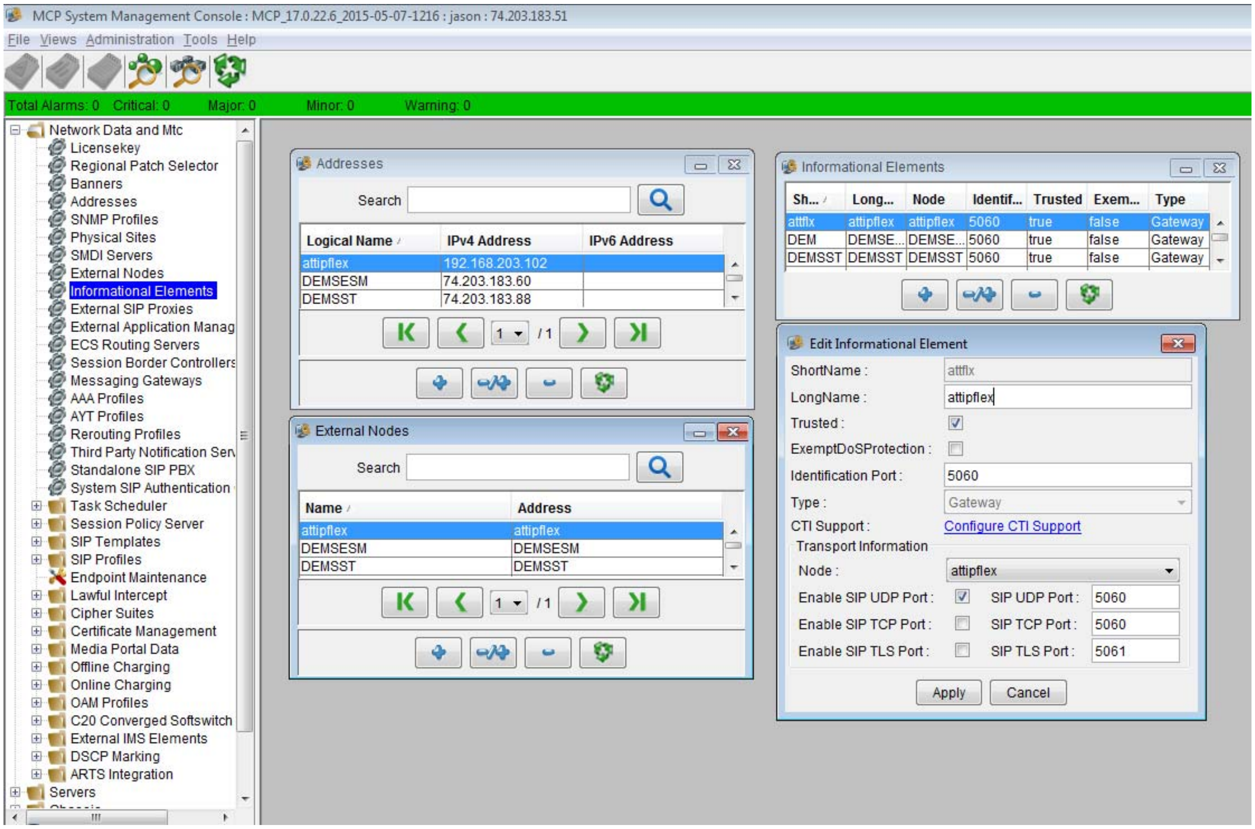
On the Protocol tab.



Set the Dest. Trunk Group and New Src. Egress Trunk Group.

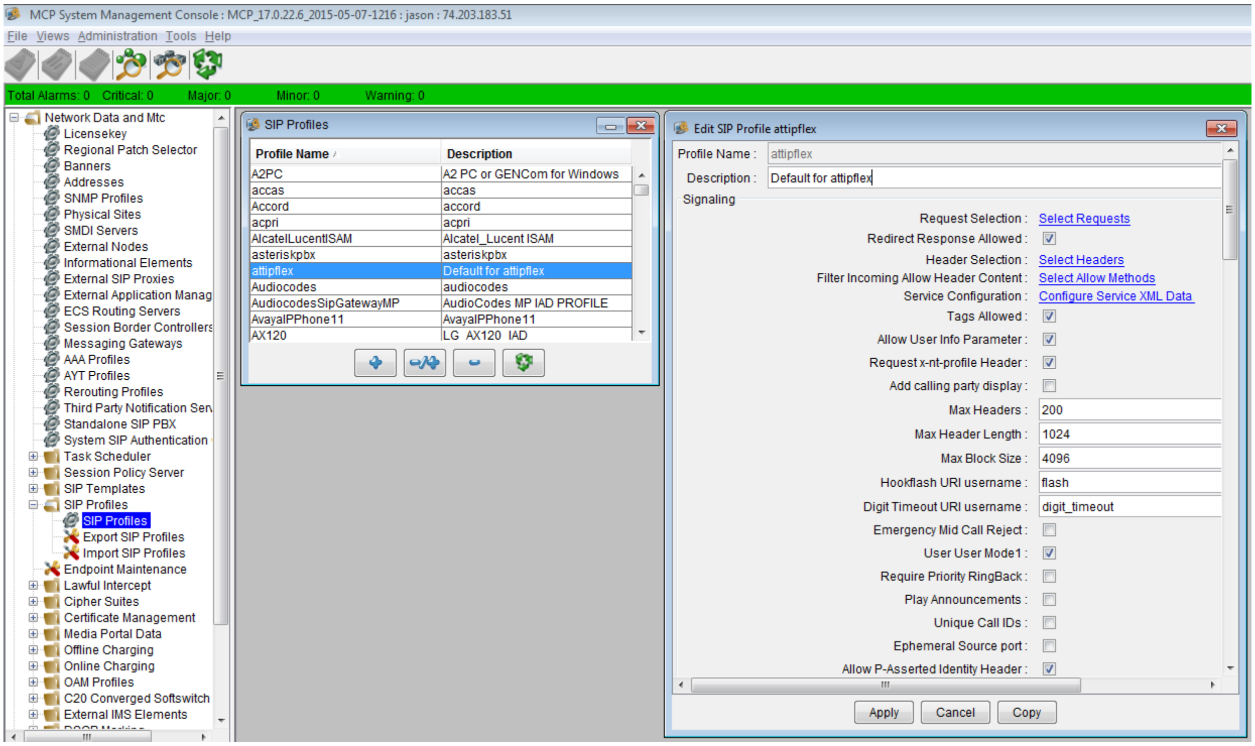
4.2 AS Configuration – MCP

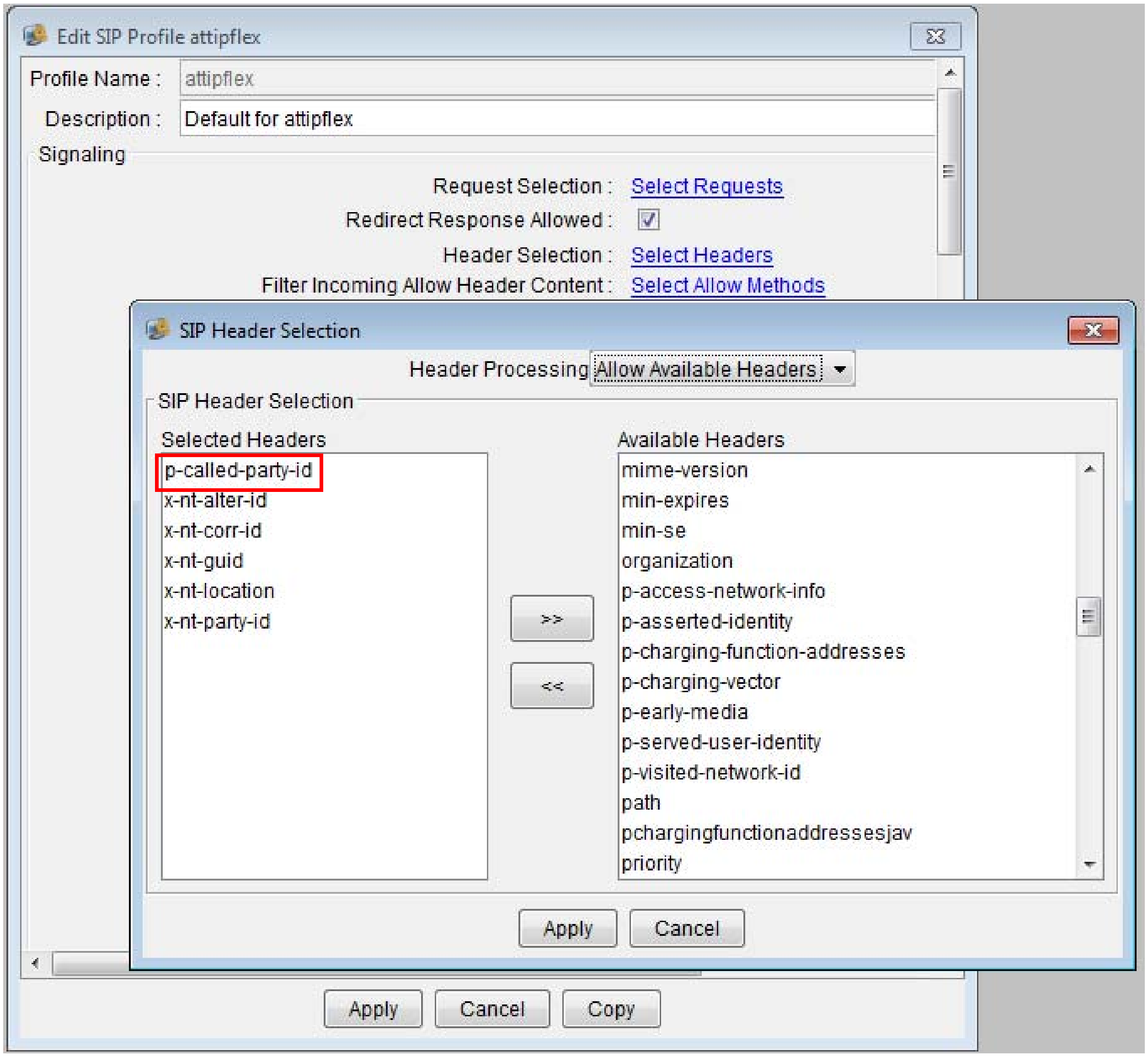
Access the AS MCP GUI and configure the SIP link to the SBC.



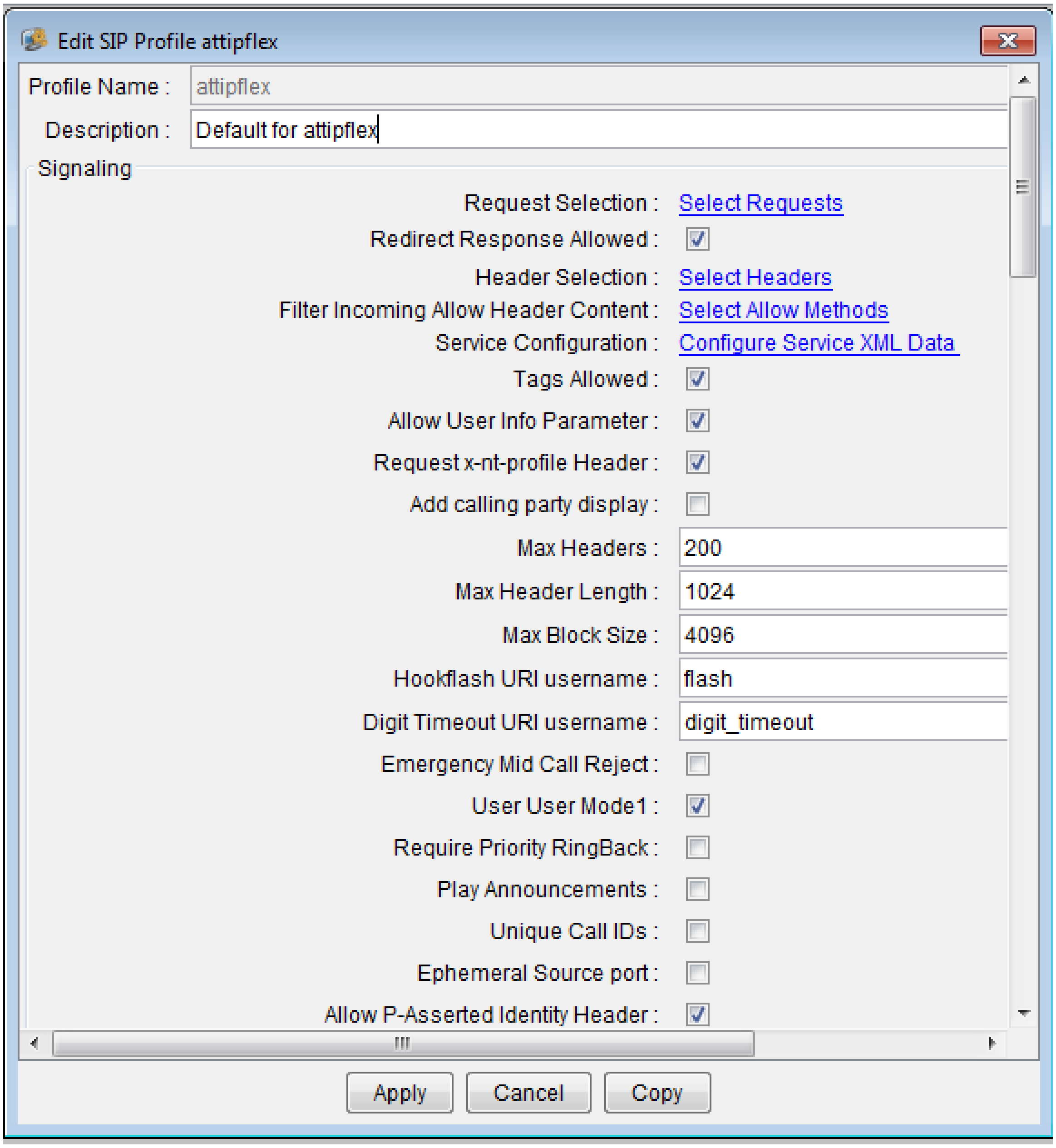
Add the Address for the SBC Private interface. Configure an External Node and a Informational Element.

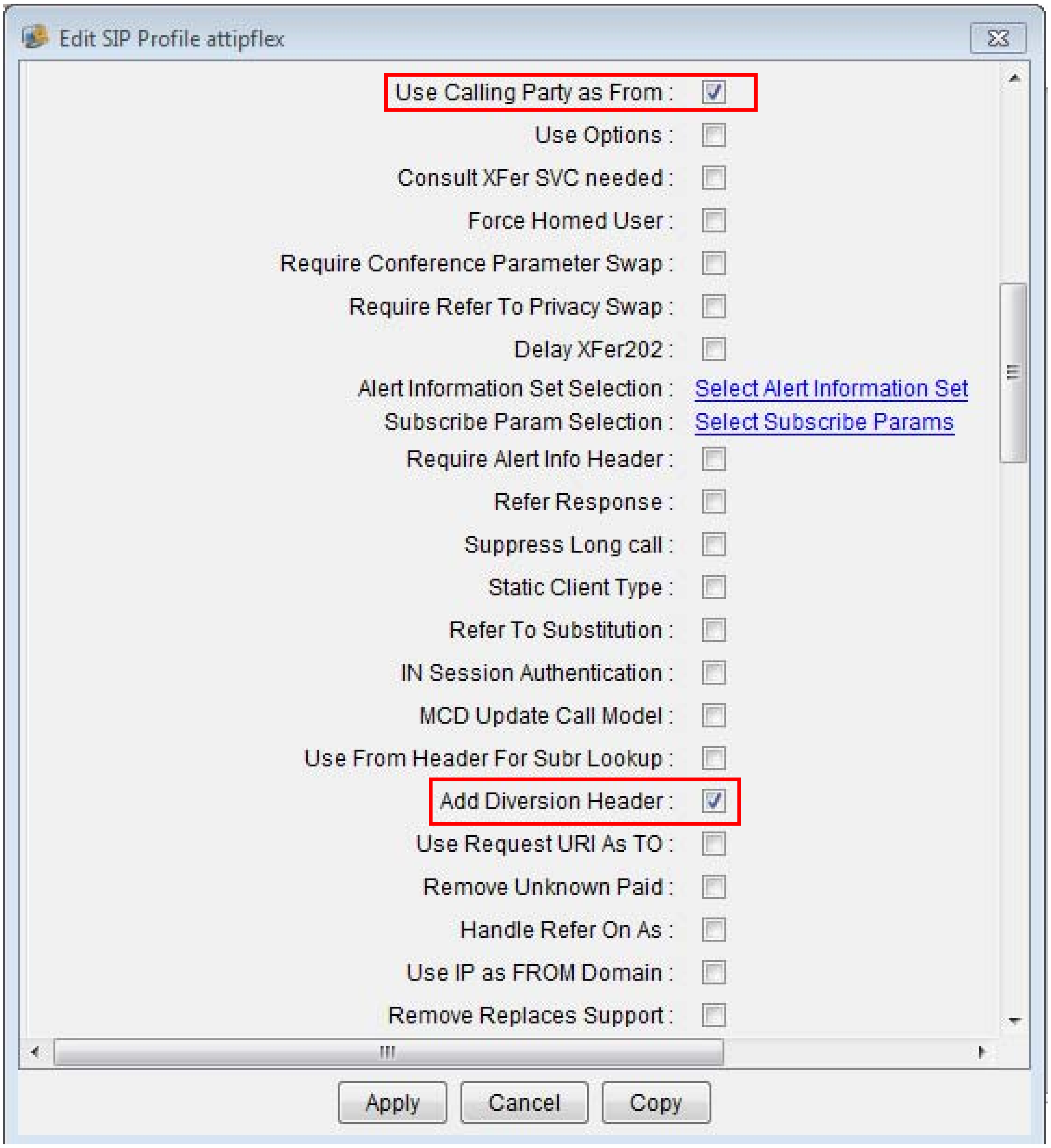
Next define a SIP Profile for use with the AT&T link.



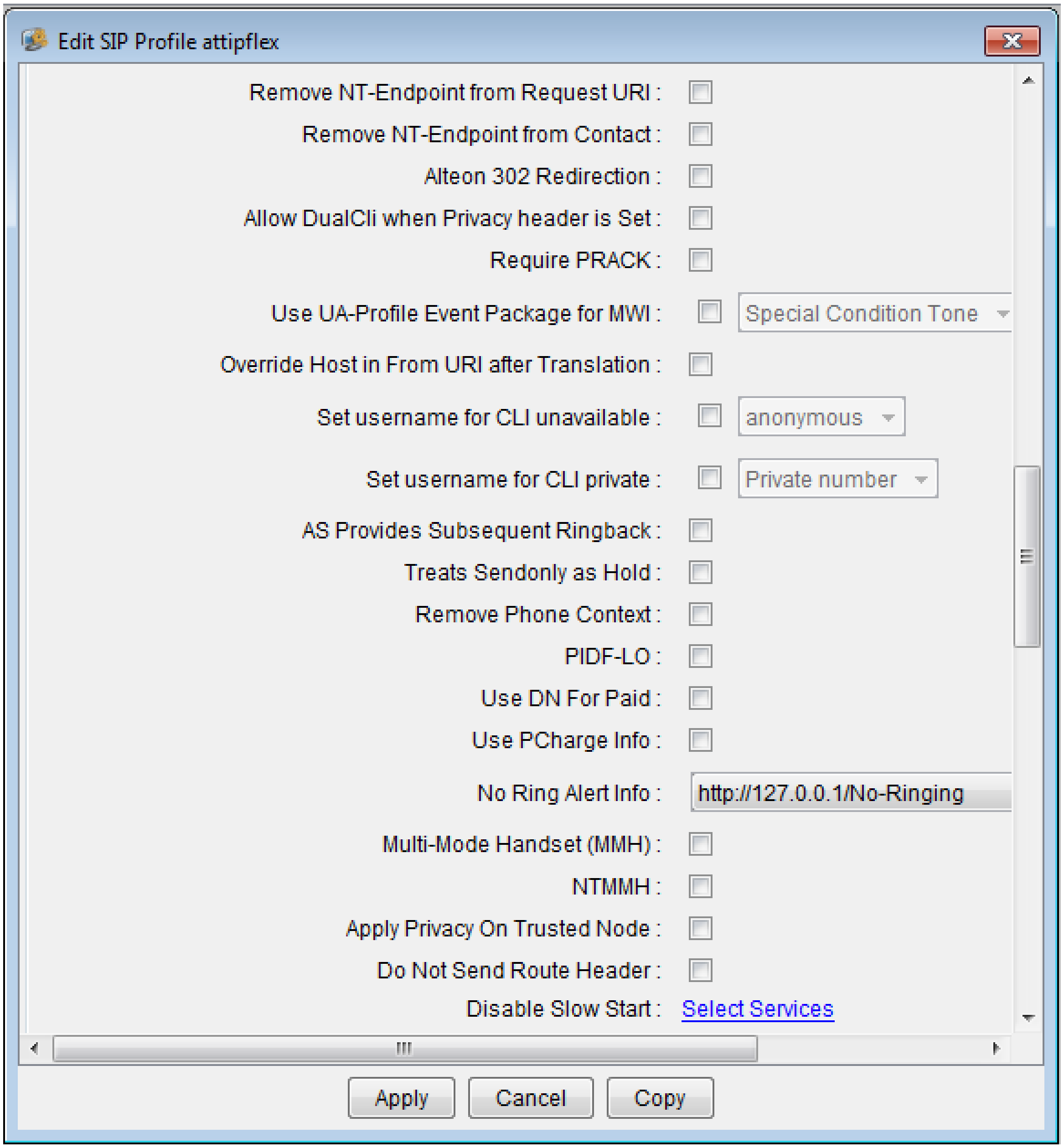


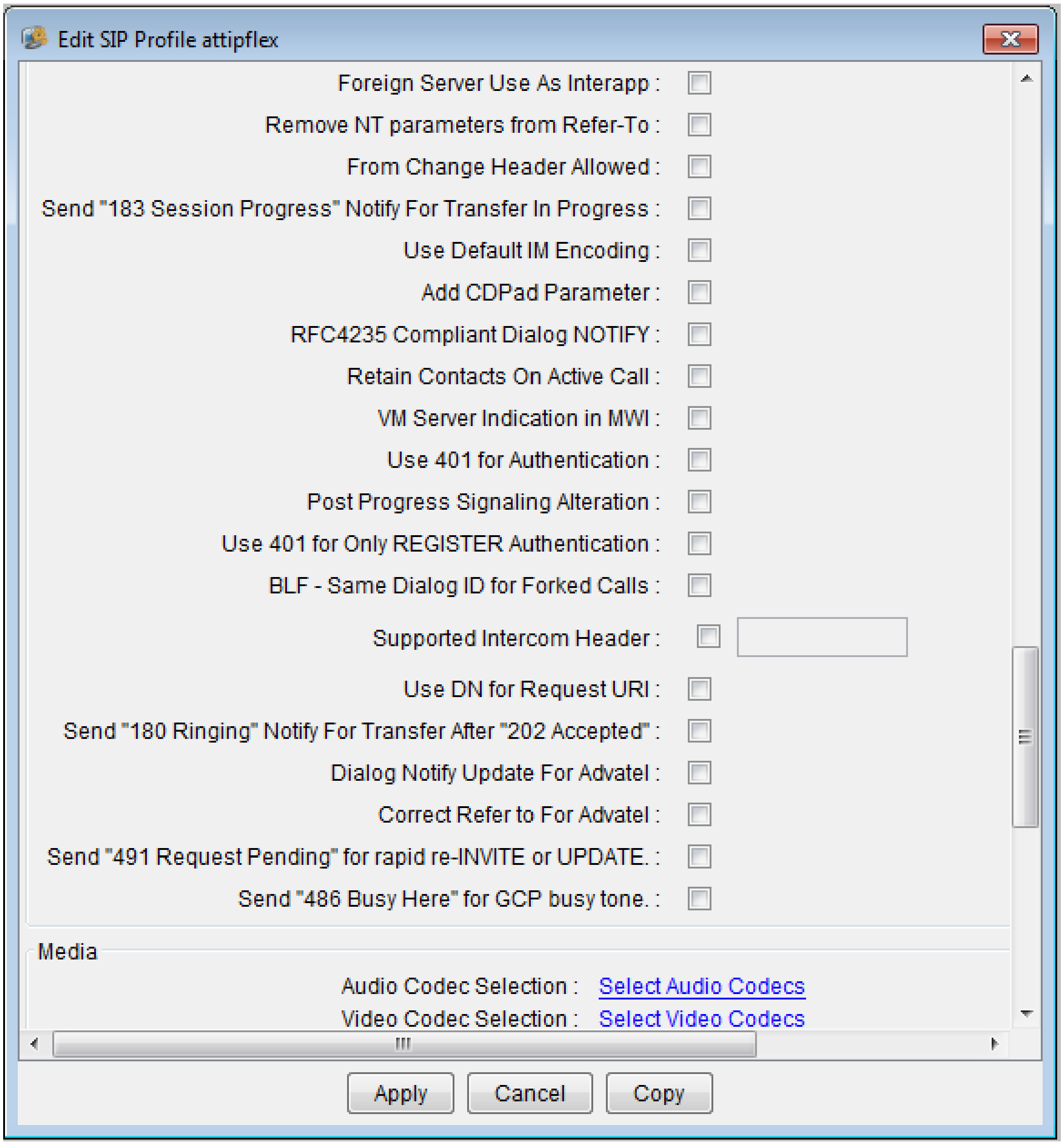
Under Header Selection verify that p-called-party-id is not in the Available Headers selection.

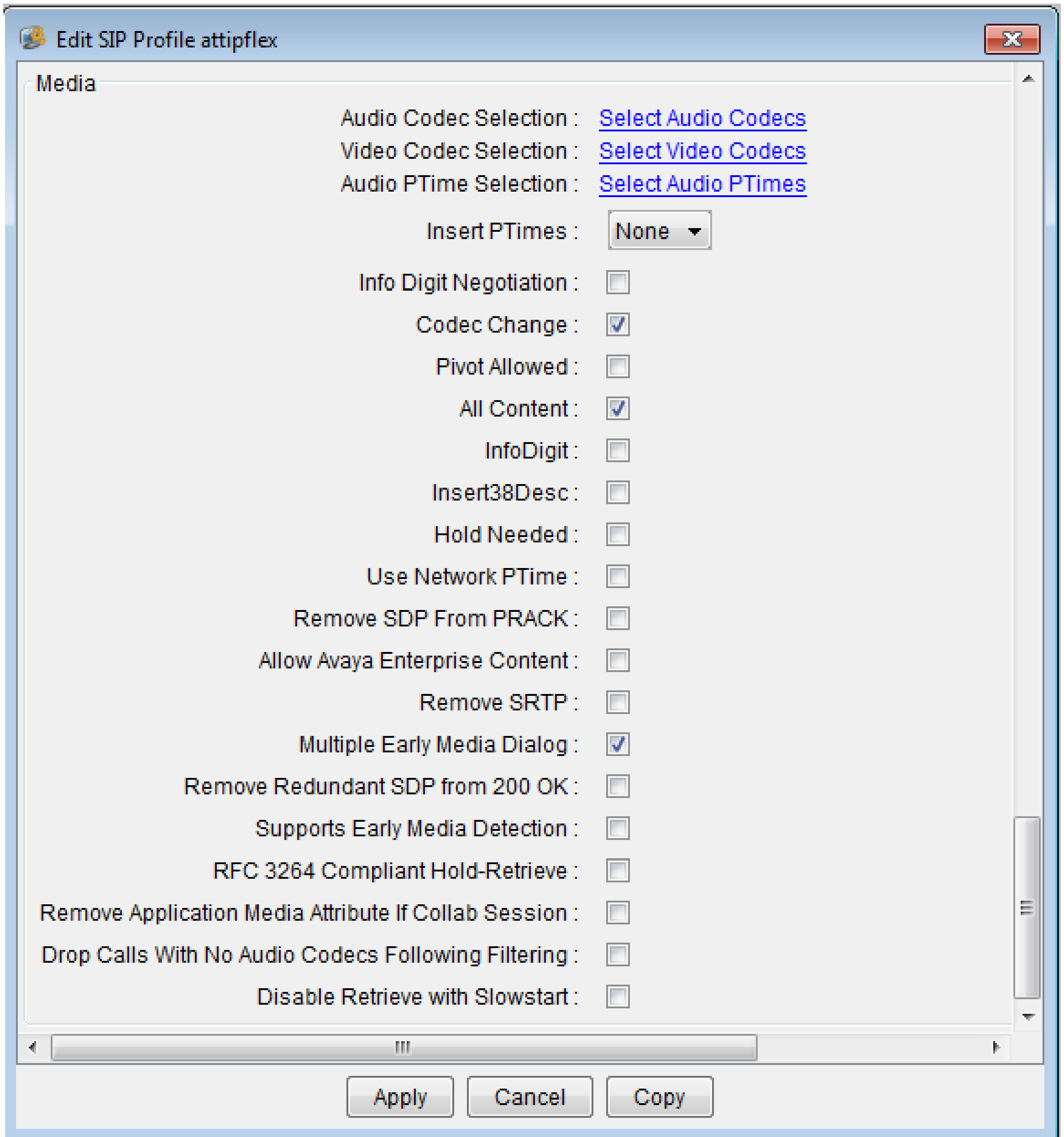

Edit SIP Profile attipflex
Profile Name : attipflex
Description : Default for attipflex
Signaling
Request Selection : Select Requests
Redirect Response Allowed :
Header Selection : Select Headers
Filter Incoming Allow Header Content : Select Allow Methods
Service Configuration : Configure Service XML Data
Tags Allowed :
Allow User Info Parameter :
Request x-nt-profile Header :
Add calling party display :
Max Headers : 200
Max Header Length : 1024
Max Block Size : 4096
Hookflash URI username : flash
Digit Timeout URI username : digit_timeout
Emergency Mid Call Reject :
User User Mode1 :
Require Priority RingBack :
Play Announcements :
Unique Call IDs :
Ephemeral Source port :
Allow P-Asserted Identity Header :
Apply
Cancel
Copy



Check Use Calling Party as From and Add Diversion Header.

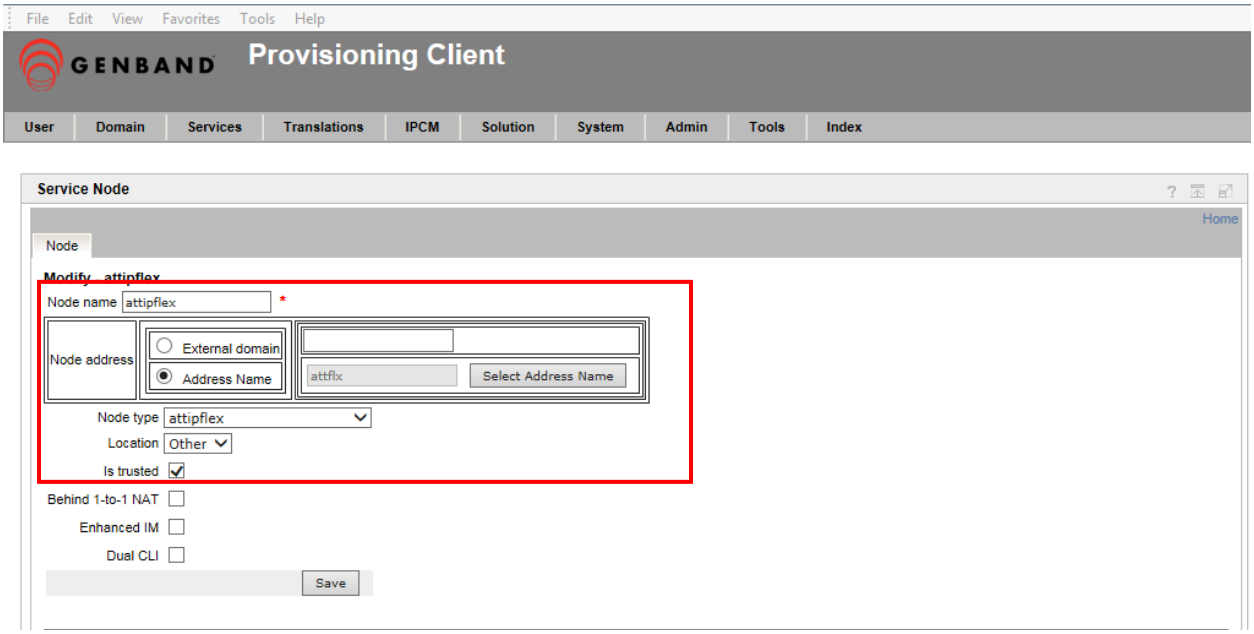

Edit SIP Profile attipflex
Remove NT-Endpoint from Request URI :
Remove NT-Endpoint from Contact :
Alteon 302 Redirection :
Allow DualCli when Privacy header is Set :
Require PRACK :
Use UA-Profile Event Package for MWI :
Special Condition Tone
Override Host in From URI after Translation :
Set username for CLI unavailable :
anonymous
Set username for CLI private :
Private number
AS Provides Subsequent Ringback :
Treats Sendonly as Hold :
Remove Phone Context :
PIDF-LO :
Use DN For Paid :
Use PCharge Info :
No Ring Alert Info :
http://127.0.0.1/No-Ringing
Multi-Mode Handset (MMH) :
NTMMH :
Apply Privacy On Trusted Node :
Do Not Send Route Header :
Disable Slow Start :
Select Services
Apply
Cancel
Copy


Edit SIP Profile attipflex
Foreign Server Use As Interapp :
Remove NT parameters from Refer-To :
From Change Header Allowed :
Send "183 Session Progress" Notify For Transfer In Progress :
Use Default IM Encoding :
Add CDPad Parameter :
RFC4235 Compliant Dialog NOTIFY :
Retain Contacts On Active Call :
VM Server Indication in MWI :
Use 401 for Authentication :
Post Progress Signaling Alteration :
Use 401 for Only REGISTER Authentication :
BLF - Same Dialog ID for Forked Calls :
Supported Intercom Header :
Use DN for Request URI :
Send "180 Ringing" Notify For Transfer After "202 Accepted" :
Dialog Notify Update For Advatel :
Correct Refer to For Advatel :
Send "491 Request Pending" for rapid re-INVITE or UPDATE. :
Send "486 Busy Here" for GCP busy tone. :
Media
Audio Codec Selection :
Select Audio Codecs
Video Codec Selection :
Select Video Codecs
Apply
Cancel
Copy

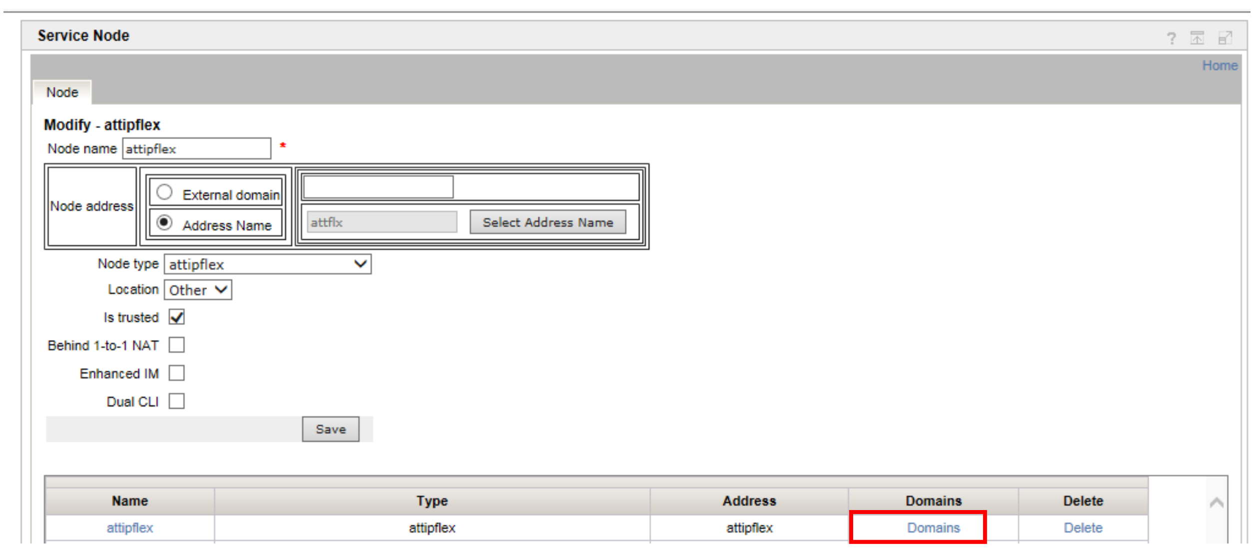

Edit SIP Profile attipflex
Media
Audio Codec Selection : Select Audio Codecs
Video Codec Selection : Select Video Codecs
Audio PTime Selection : Select Audio PTimes
Insert PTimes : None
Info Digit Negotiation :
Codec Change :
Pivot Allowed :
All Content :
InfoDigit :
Insert38Desc :
Hold Needed :
Use Network PTime :
Remove SDP From PRACK :
Allow Avaya Enterprise Content :
Remove SRTP :
Multiple Early Media Dialog :
Remove Redundant SDP from 200 OK :
Supports Early Media Detection :
RFC 3264 Compliant Hold-Retrieve :
Remove Application Media Attribute If Collab Session :
Drop Calls With No Audio Codecs Following Filtering :
Disable Retrieve with Slowstart :
Apply
Cancel
Copy

4.3 AS Configuration – Prov

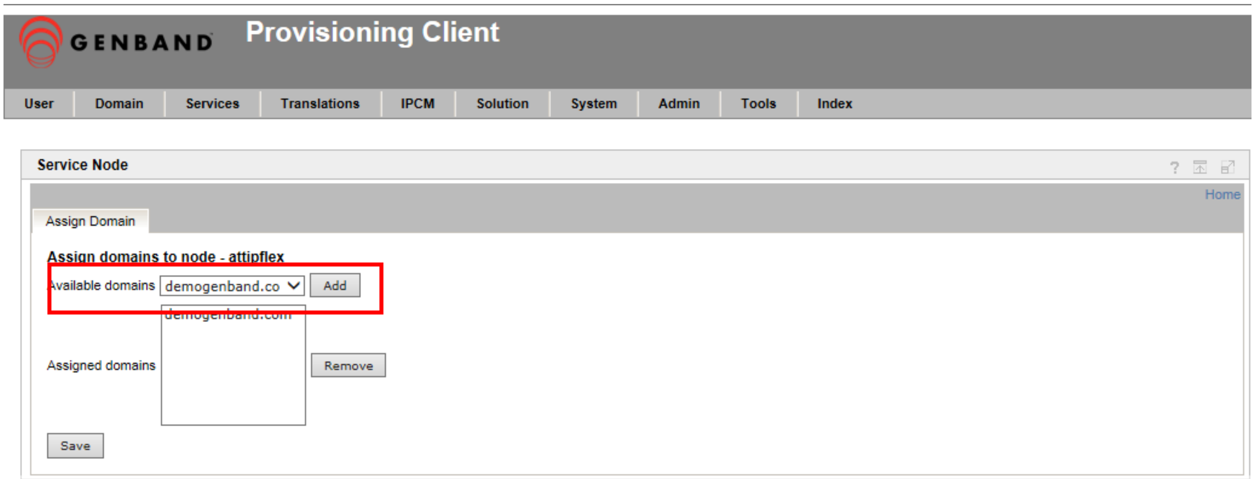
Access the AS Prov WEB GUI.



Under Translations, Service Nodes, add a Node for the AT&T link to the SBC. Enter a Node name. For the Node address choose the one built prior, the Informaiton Element name. Node type will be the SIP Profile name you built. Check the Is trusted box.

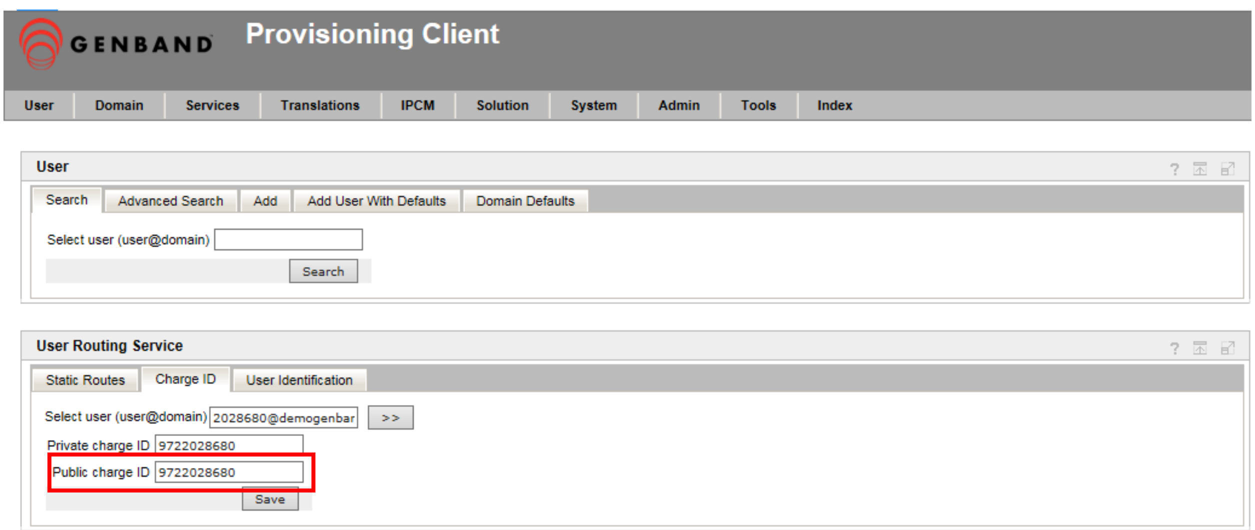


Click on Domains.



Add your SIP Domain and hit save. This finishes the build of the link for the SIP trunk to the SBC.

Verify for the SIP users that will utilize the AT&T IPFLR service that the following is set. Select the user/s and go to User Routing.



Verify the Public charge ID is set to a valid DID for the AT&T IPFLR trunk.

5 Troubleshooting

Issue isolation should be the first step. Perform traffic captures at various points in the call path to isolate were the fault begins. If the issue is isolated to a GENBAND product then contact GENBAND customer support.
Call: 1-866-GENBAND
WEB: www.genband.com