

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

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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

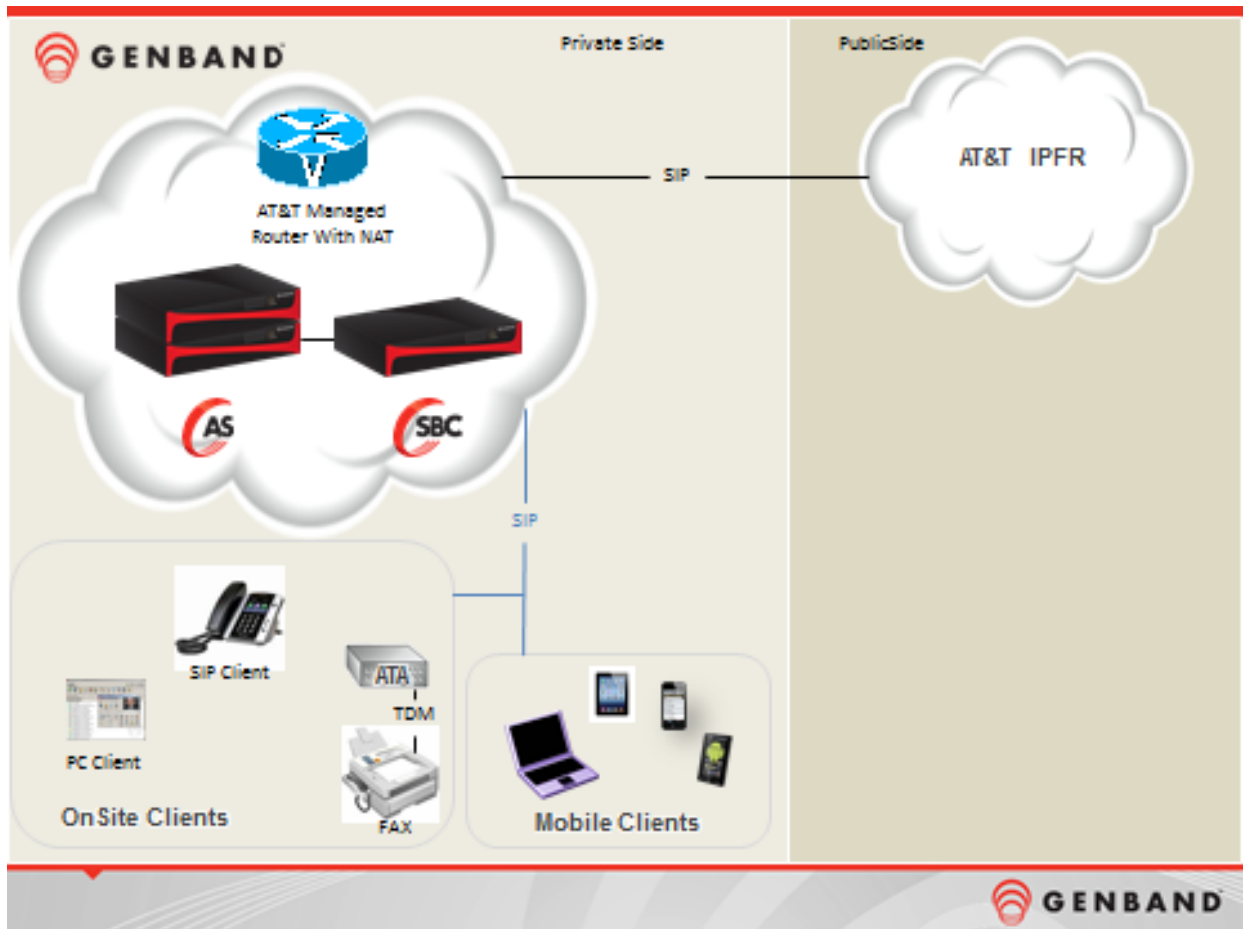
- AT&T requires RTP ports 16384-32767 be used for media (these are configured in the GENBAND SBC used for AT&T trunk connectivity)
- Preferred configuration is to handle SIP Refer messages on the GENBAND AS. This is accomplished by setting the SIP Profile for the Node associated to the AT&T trunk.
- "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

The screenshot shows a window titled "Modify Signaling Vnet" with the following configuration fields:

- Partition: admin
- Name: attvnet
- Interface Name: eth2
- VLAN ID: 902
- IP Version: IPv4
- Firewall Zone Name: def_sig_zone
- Primary Gateway: 32.252.201.155

The "Gateway Monitoring" section includes:

- Gateway Monitoring: None
- Secondary Gateway: 0.0.0.0
- Ping Interval (in milliseconds): 500
- Success Ping Count: 5
- Failed Ping Count: 3
- SNMP Trap Reporting
- Switch to StandBy

Buttons at the bottom: Modify, Cancel

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>

Modify Realm

Signaling Media Rate Limit QoS

Partition: admin

Enable Signaling

Realm Name: att-peer

SIP Authentication: <none>
inv
reg

SIP Blocked Methods: <none>
inv
reg

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):

SIP Invite Max Retransmit count:

Max SDP Session Id:

Max SDP Version Id:

Split SIP Header:

Accent Large SIP Header

Modify Cancel

Enter a Realm Name.

The image shows a 'Modify Realm' dialog box with the 'Signaling' tab selected. The 'Realm Signaling Address' and 'Subnet Mask' fields are highlighted with a red box. The address is 32.252.201.154 and the subnet mask is 255.255.255.248. Other fields include Domain Names (Comma Separated), Dynamic EP Domain, Vnet Name (attvnet), Sip Port 1-3, TLS Port 1-3, 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, and Dynamic Override Required.

Realm Signaling Address :	32.252.201.154
Subnet Mask:	255.255.255.248
Domain Names (Comma Separated):	
Dynamic EP Domain:	
Vnet Name:	attvnet
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:	<input type="checkbox"/>
Remove 100rel supported HDR	<input type="checkbox"/>
Remove timer supported HDR	<input type="checkbox"/>
Forward options without username:	Inherit from system
Remove Prohibited Codec PTime:	<input type="checkbox"/>
Pass Contact HDR UserName:	<input type="checkbox"/>
Allow All Dynamic Endpoint From Users:	<input type="checkbox"/>
Dynamic Override Required:	<input type="checkbox"/>

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 ren

SIP Blocked Methods: <none> inv ren

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):

SIP Invite Max Retransmit count:

Max SDP Session Id:

Max SDP Version Id:

Split SIP Header:

Accent Large SIP Header

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:

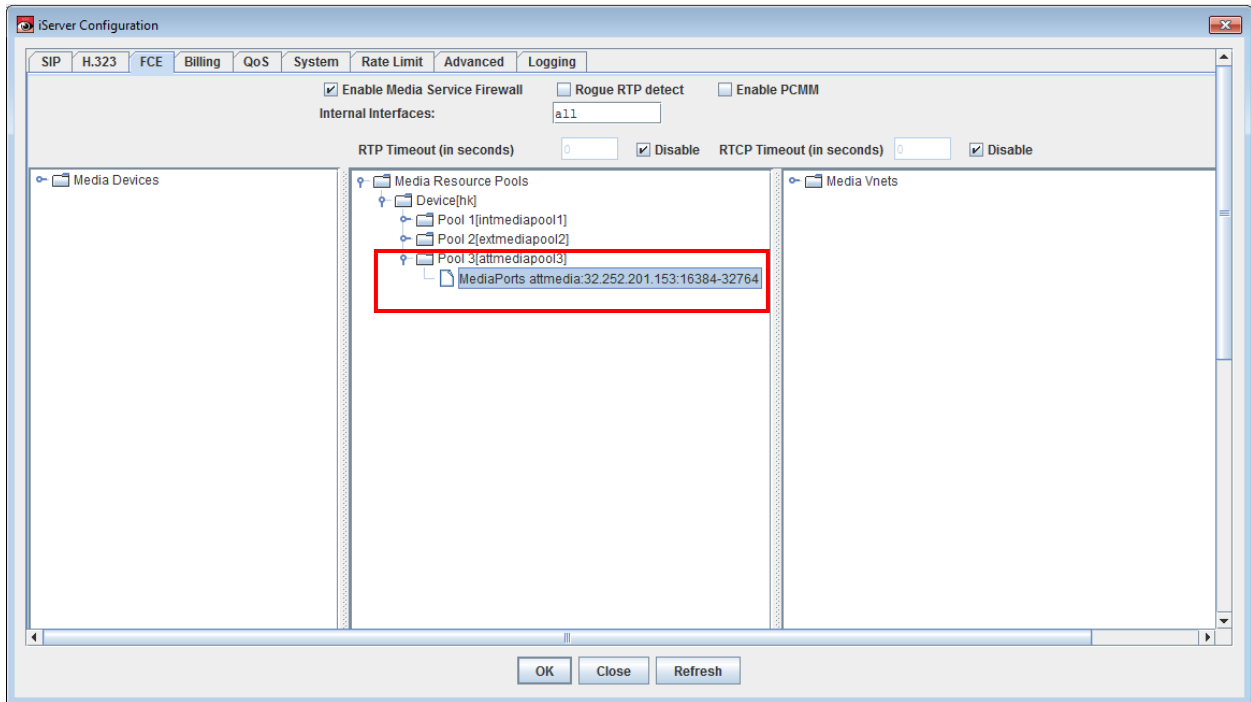
Pass Contact HDR UserName:

Allow All Dynamic Endpoint From Users:

Dynamic Override Required:

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.

The screenshot shows a configuration window titled "Modify SIP Gateway:attipflex/0". The window has several tabs: "Phone", "Advanced", "User Info", "Protocol", "Calls", and "Rate Limit". The "Phone" tab is selected. The configuration fields are as follows:

- Partition: admin (dropdown)
- Device Type: SIP Gateway (dropdown)
- Registration ID: attipflex (text field)
- Port Number: 0 (text field)
- IP Address: 12.194.20.88 (text field) with a checked "Static" checkbox.
- Allow Dynamic Registration: unchecked checkbox.
- Extension: (empty text field)
- Calling Plan: (empty text field)
- Carrier ID: 0 (text field)
- Realm: att-peer (text field)
- Domain Name: <none> (dropdown)
- IEdge Group: 1 (text field)
- Transcoding: Not defined (dropdown)
- Codec Profile: <none> (text field)
- no T.38 support: unchecked checkbox.
- Blacklist Enable: unchecked checkbox.

At the bottom of the window are "OK" and "Cancel" buttons.

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

On the Protocol tab.

The screenshot shows a configuration window titled "Modify SIP Gateway:attipflex/0" with several tabs: "Phone", "Advanced", "User Info", "Protocol", "Calls", and "Rate Limit". The "Protocol" tab is selected. The window is divided into several sections:

- Gateway/Proxy:** Contains a checked checkbox for "Gateway/Proxy" and a "Priority:" field.
- SIP/H323:** Contains a checked checkbox for "SIP" with a "Configure" button, an unchecked checkbox for "H.323" with a "Configure" button, and an unchecked checkbox for "Disable SIP 183 Support On Peer". The "URI (SIP / H323)" field is highlighted with a red box and contains the value "12.194.20.88". Below this are dropdown menus for "Ani Based Auth" (set to "no"), "Enforce Privacy" (set to "no"), "Mid Call Codec Change" (set to "Unknown"), and "Calling Party Number Type" (set to "Pass(Default)"). There is also a "Codemap File Number:" field.
- Cisco GTD:** Contains an unchecked checkbox for "Enable Cisco GTD support".
- Trunk Group:** Contains an unchecked checkbox for "x-route-tag Support", a "Src. Trunk Group:" field, and a "Dest. Trunk Group:" field containing the value "att_sa".

At the bottom of the window are "OK" and "Cancel" buttons.

Add the URI.

Modify SIP Gateway:attipflex/0

Phone Advanced User Info Protocol Calls Rate Limit

Enforce Privacy: no

Mid Call Codec Change: Unknown

Calling Party Number Type: Pass(Default)

Codemap File Number:

Cisco GTD

Enable Cisco GTD support

Trunk Group

x-route-tag Support

Src. Trunk Group:

Dest. Trunk Group: att_sa

New Src. Ingress Trunk Group:

New Src. Egress Trunk Group: sa_att

Send Dest. Trunk Group

Remove Src. Trunk Group

New Origination TG on Egress:

New Destination TG on Egress:

Trunk Context

Support For 4904 Originating Trunk Group

Source Trunk Context:

OK Cancel

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

Add another Endpoint for the backup proxy.

Modify SIP Gateway:attipflex2/0

Phone | Advanced | User Info | Protocol | Calls | Rate Limit

Phone

Partition: admin

Device Type: SIP Gateway

Registration ID: attipflex2

Port Number: 0

IP Address: 12.194.18.88 Static

Allow Dynamic Registration

Extension:

Calling Plan:

Carrier ID: 0

Realm: att-peer

Domain Name: <none>

IEdge Group: 1

Transcoding: Not defined

Codec Profile: <none>

no T.38 support

Blacklist Enable

OK Cancel

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

On the Protocol tab.

The screenshot shows a configuration window titled "Modify SIP Gateway:attiplflex2/0" with several tabs: "Phone", "Advanced", "User Info", "Protocol" (selected), "Calls", and "Rate Limit". The "Protocol" tab is active and contains the following settings:

- Gateway/Proxy:** Gateway/Proxy, Priority:
- SIP/H323:** SIP (with "Configure" button), H.323 (with "Configure" button), Disable SIP 183 Support On Peer
- URI (SIP / H323):** (highlighted with a red box)
- Ani Based Auth:** (dropdown)
- Enforce Privacy:** (dropdown)
- Mid Call Codec Change:** (dropdown)
- Calling Party Number Type:** (dropdown)
- Codemap File Number:**
- Cisco GTD:** Enable Cisco GTD support
- Trunk Group:** x-route-tag Support
- Src. Trunk Group:**
- Dest. Trunk Group:**

At the bottom of the window are "OK" and "Cancel" buttons.

Add the URI.

Modify SIP Gateway:attipflex2/0

Enable Cisco GTD support

Trunk Group

x-route-tag Support

Src. Trunk Group:

Dest. Trunk Group:

New Src. Ingress Trunk Group:

New Src. Egress Trunk Group:

Send Dest. Trunk Group

Remove Src. Trunk Group

New Origination TG on Egress:

New Destination TG on Egress:

Trunk Context

Support For 4904 Originating Trunk Group

Source Trunk Context:

Destination Trunk Context:

New Source Ingress Trunk Context:

New Destination Ingress Trunk Context:

New Source Egress Trunk Context:

New Destination Egress Trunk Context:

OK Cancel

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

Now add the Endpoint to the GENBAD AS

Modify SIP Proxy:demostandalone/2

Phone | Advanced | User Info | Protocol | Calls | Rate Limit

Phone

Partition: admin

Device Type: SIP Proxy

Registration ID: demostandalone

Port Number: 2

IP Address: 74.203.183.52 Static

Allow Dynamic Registration

Extension:

Calling Plan:

Carrier ID: 0

Realm: attintpeer

Domain Name: <none>

IEdge Group: 1

Transcoding: Not defined

Codec Profile: <none>

no T.38 support

Blacklist Enable

OK Cancel

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

On the Protocol tab.

Modify SIP Proxy: demostandalone/2

Enable Cisco GTD support

Cisco GTD

x-route-tag Support

Trunk Group

Src. Trunk Group:

Dest. Trunk Group:

New Src. Ingress Trunk Group:

New Src. Egress Trunk Group:

Send Dest. Trunk Group

Remove Src. Trunk Group

New Origination TG on Egress:

New Destination TG on Egress:

Trunk Context

Support For 4904 Originating Trunk Group

Source Trunk Context:

Destination Trunk Context:

New Source Ingress Trunk Context:

New Destination Ingress Trunk Context:

New Source Egress Trunk Context:

New Destination Egress Trunk Context:

OK Cancel

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.

The screenshot displays the MCP System Management Console interface. The top status bar shows "Total Alarms: 0 Critical: 0 Major: 0 Minor: 0 Warning: 0". The left navigation pane lists various system components, with "Informational Elements" selected. The main workspace contains three panels:

- Addresses:** A table listing logical names and their corresponding IPv4 and IPv6 addresses.
- External Nodes:** A table listing node names and their addresses.
- Informational Elements:** A table listing elements with columns for Short Name, Long Name, Node, Identification Port, Trusted status, Exempt DoS Protection, and Type. Below the table is an "Edit Informational Element" dialog box.

The "Edit Informational Element" dialog box shows the following configuration for the selected element:

- ShortName: attiflex
- LongName: attiflex
- Trusted:
- ExemptDoSProtection:
- Identification Port: 5060
- Type: Gateway
- CTI Support: [Configure CTI Support](#)
- Transport Information:
 - Node: attiflex
 - Enable SIP UDP Port: SIP UDP Port: 5060
 - Enable SIP TCP Port: SIP TCP Port: 5060
 - Enable SIP TLS Port: SIP TLS Port: 5061

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.

The screenshot displays the MCP System Management Console interface. The main window is titled "MCP System Management Console : MCP_17.0.22.6_2015-05-07-1216 : jason : 74.203.183.51". The interface includes a menu bar (File, Views, Administration, Tools, Help) and a status bar showing "Total Alarms: 0 Critical: 0 Major: 0 Minor: 0 Warning: 0".

The left sidebar shows a tree view of system components, with "SIP Profiles" selected. The main area is divided into two panes:

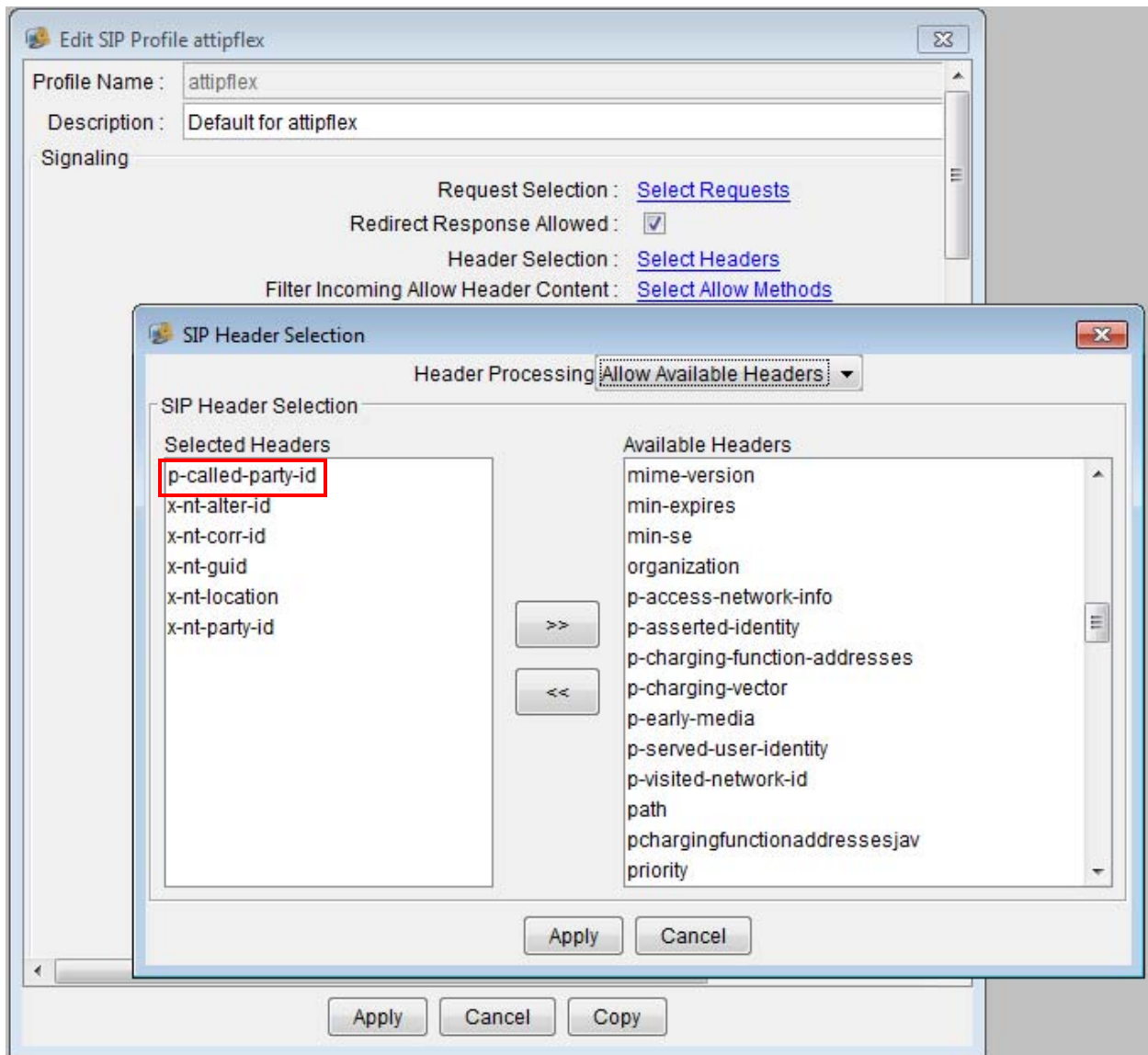
- SIP Profiles:** A table listing various profiles. The "attipflex" profile is highlighted.
- Edit SIP Profile attipflex:** A configuration window for the selected profile.

Profile Name	Description
A2PC	A2 PC or GENCom for Windows
accas	accas
accord	accord
acpri	acpri
AlcateLucentISAM	Alcate_Lucent ISAM
asteriskpbx	asteriskpbx
attipflex	Default for attipflex
Audiocodes	audiocodes
AudiocodesSipGatewayMP	AudioCodes MP IAD PROFILE
AvayaIPPhone11	AvayaIPPhone11
AX120	LG AX120 IAD

The "Edit SIP Profile attipflex" window shows the following configuration details:

- Profile Name: attipflex
- Description: Default for attipflex
- 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-m-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 Mode 1:
- Require Priority RingBack:
- Play Announcements:
- Unique Call IDs:
- Ephemeral Source port:
- Allow P-Asserted Identity Header:

Buttons at the bottom of the "Edit SIP Profile attipflex" window include "Apply", "Cancel", and "Copy".



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 :

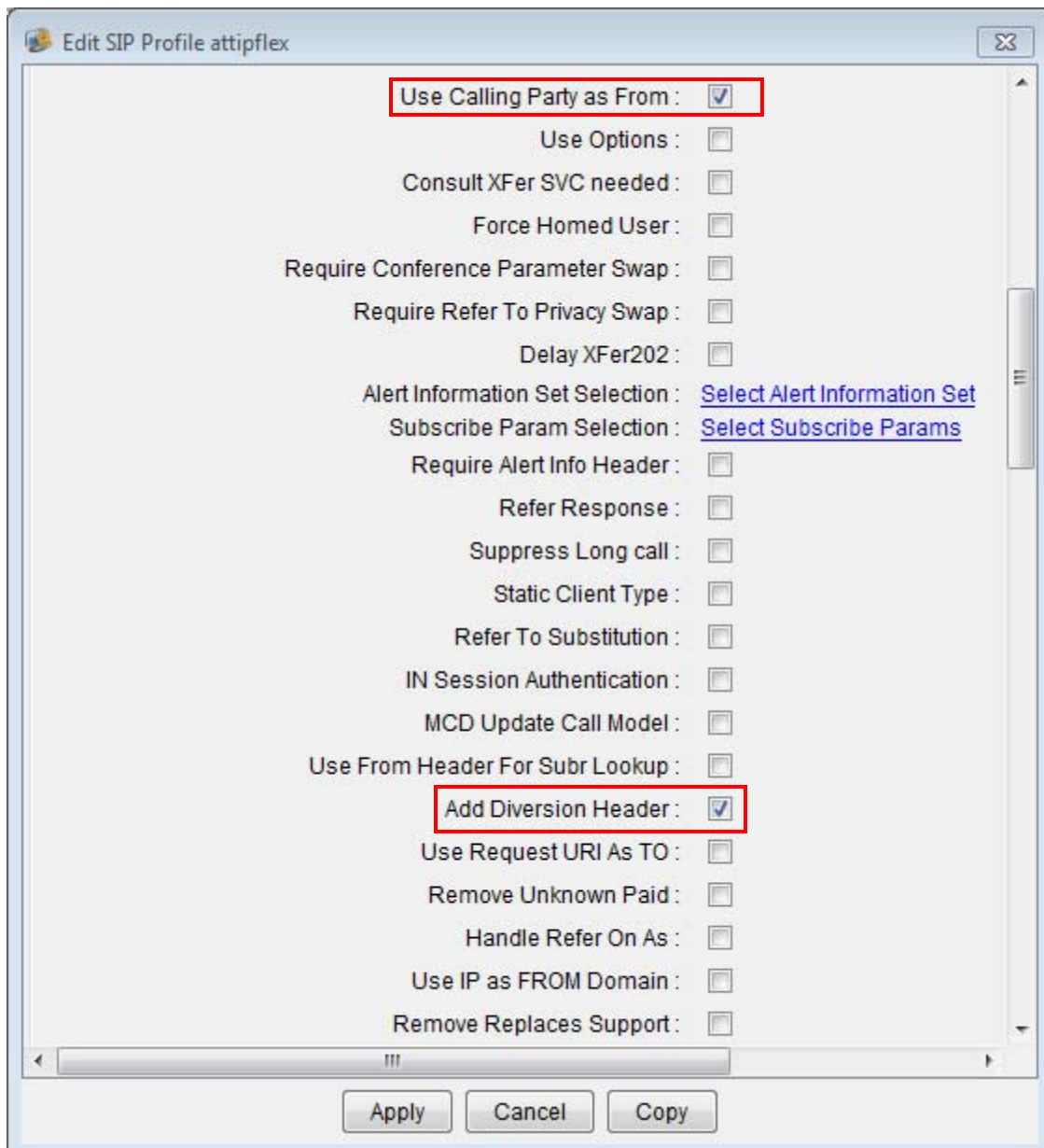
Play Announcements :

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 attiplflex

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. :

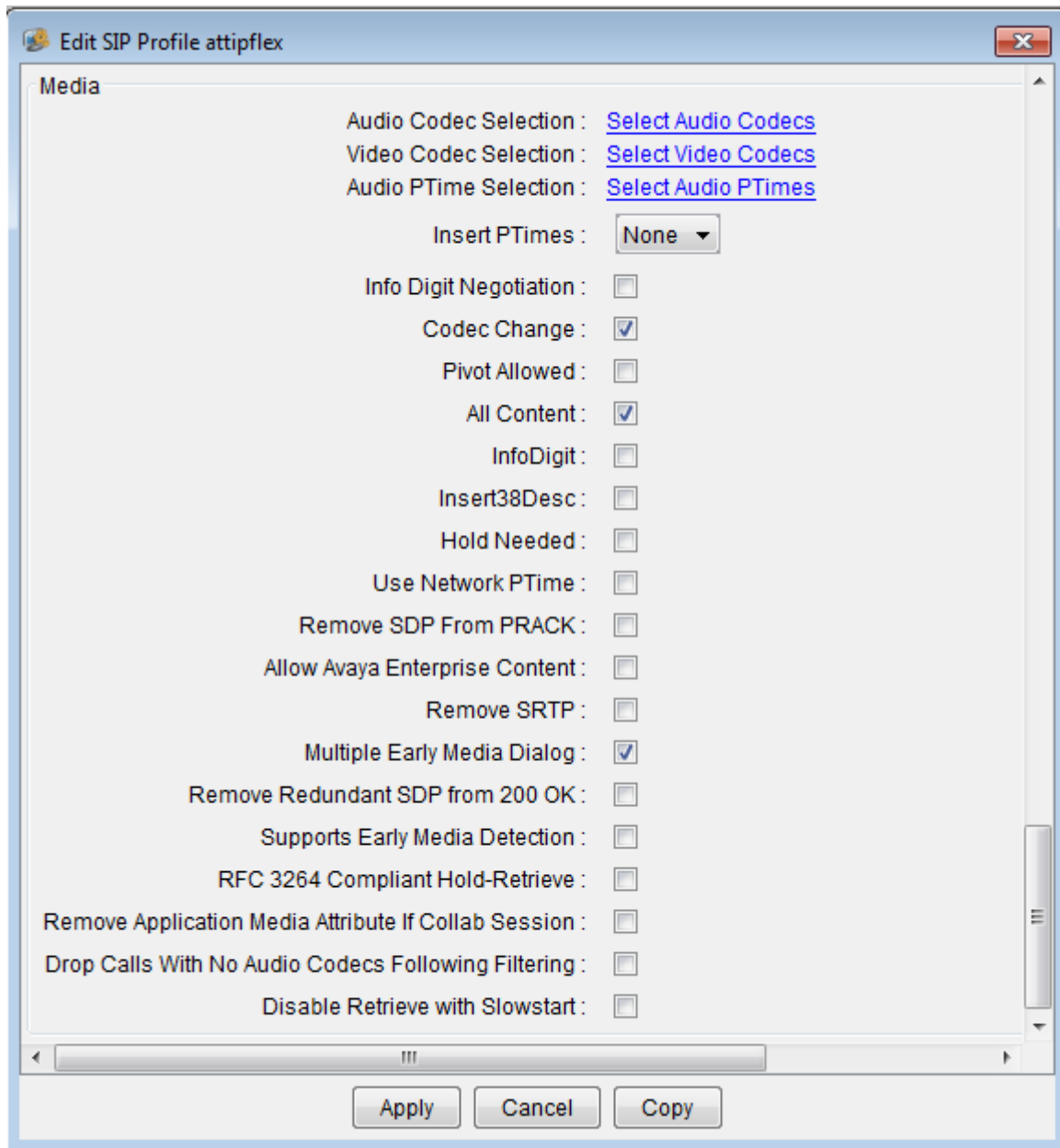
Send "486 Busy Here" for GCP busy tone. :

Media

Audio Codec Selection : [Select Audio Codecs](#)

Video Codec Selection : [Select Video Codecs](#)

Apply Cancel Copy



4.3 AS Configuration – Prov

Access the AS Prov WEB GUI.

The screenshot shows the 'Service Node' configuration page in the GENBAND Provisioning Client. The page title is 'Service Node' and the breadcrumb is 'Node'. The main form is titled 'Modify - attipflex'. The form fields are: Node name (text input: attipflex), Node address (radio buttons for 'External domain' and 'Address Name', with 'Address Name' selected; a text input field contains 'attipflex' and a 'Select Address Name' button), Node type (dropdown menu: attipflex), Location (dropdown menu: Other), and Is trusted (checkbox: checked). Below these are three unchecked checkboxes: Behind 1-to-1 NAT, Enhanced IM, and Dual CLI. A 'Save' button is at the bottom of the form.

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.

The screenshot shows the 'Service Node' configuration page with the 'Domains' tab selected in the table below the form. The table has columns for Name, Type, Address, Domains, and Delete. The 'attipflex' node is listed, and the 'Domains' column for this node is highlighted with a red box.

Name	Type	Address	Domains	Delete
attipflex	attipflex	attipflex	Domains	Delete

Click on Domains.

Service Node

Assign Domain

Assign domains to node - attioflex

Available domains: demogenband.co Add

Assigned domains: Remove

Save

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.

User

Search Advanced Search Add Add User With Defaults Domain Defaults

Select user (user@domain) Search

User Routing Service

Static Routes Charge ID User Identification

Select user (user@domain) 2028680@demogenbar >>

Private charge ID 9722028680

Public charge ID 9722028680

Save

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 where 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