VoIP Application Note:

AltiGen IP-PBX SIP Trunk Configuration Using Bandwidth.com

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Configure an AltiGen IP-PBX

Goal

Provide a reference to AltiGen-certified dealers for configuring an AltiGen IP PBX (AltiServ) with Bandwidth.com's SIP Trunking service.

Introduction

From an AltiGen administrator's point of view, a SIP trunk is very similar to a Triton analog trunk. Once it is up and running, administrators can assign a trunk access code to it. They also can mix SIP trunks with analog trunks, T1, or PRI trunks in the Out Call Routing table. Like analog trunks, each SIP trunk has a PSTN number. They can also receive caller name as well as caller ID.

Prerequisites

SIP trunking information provided by Bandwidth.com:

- Primary, SIP proxy server IP address **216.82.224.202** or DNS **sp-udp01.iad.bandwidth.com**
- Secondary, SIP proxy server IP address **216.82.225.202** or DNS **sp-udp01.sna.bandwidth.com**
- Or, you can use DNS SRV **ot.bandwidth.com** (the EdgeMarc router supports this)
- Trunking DID(s):
  - The DID(s) are forwarded to the Public WAN IP address(s), DNS or DNS SRV records of the EdgeMarc or InGate SIParator.

EdgeMarc 4500 series:

The EdgeMarc is used to provide NAT traversal for SIP messaging. It is also a SIP-aware firewall and VoIP traffic shaping device.

- Production release VOS software 7.9.9 or better
- Public IP address for WAN SIP Trunking
- Private LAN IP Address
- Private IP address of the Altigen

InGate SIParator series:

The InGate SIParator is used to provide NAT traversal for SIP messaging. It is optionally capable of providing a SIP-aware firewall and tagging packets for SIP QoS.

- Production release 4.5.2 (or better)
- InGate Startup Tool 1.1.7 (or better)
- Public IP address for WAN SIP trunking
- Private LAN IP Address
- Private IP address of the AltiGen IP-PBX

AltiGen Max 1000:

- AltiWare Version 6.0 Update 1 ACC or ACM (or better)
- Any phones that work with Altigen Max 1000

References:

- Ingate SIParator Getting Started Guide
- EdgeMarc 4500 Administrator's Guide
- AltiGen Administrator's Guide
- Bandwidth.com Connecting with SIP

Installation Worksheet

Use this form to aid in setting up your SIP Trunking service.

WAN Side:

Internet Access Type & Speed:	
WAN IP Address:	
WAN Subnet Mask:	
WAN Gateway IP Address:	

LAN Side:

LAN IP of SIParator/EdgeMarc:	
LAN Subnet Mask:	
LAN IP Address of AltiGen IP-PBX:	
VLAN ID:	

PBX Info:

Model:	
Firmware Version:	
Number of SIP Trunk Licenses:	
Add-on Software Applications:	
Number of Users:	
Number of Concurrent Calls:	

NAT Traversal:

Devise:	
Serial #:	
MAC:	
Firmware Version:	

Notes:

Call Quality Management

Internet Access Quality:

Traffic Shaping: The EdgeMarc appliance from EdgeWater Networks is the device we use to deploy traffic shaping. It shapes both inbound and outbound data/voice using standard TCP collision management. The benefit of this method is very competent voice quality on any Tier 1-provided Internet connection. For more information on this, check out the knowledgebase at www.edgewaternetworks.com .

Voice Optimized Internet Access: Voice Optimized Internet access is a specially delivered circuit with two separate subnets: one prioritized for voice, and one for data. The benefit of this method is that it is offered on DSL and T-1, and the voice quality is excellent. There is also a proprietary router provided with this solution. The InGate or EdgeMarc can still be used behind the router for NAT traversal.

Packet Prioritization: This method utilizes priority marked queues or packets to ensure quality across the carrier Internet backbones.

> Qwest Priority T-1 – This method provided by Qwest allows voice to reside on a priority queue for outbound traffic. The issues with this method are price and non-prioritization of inbound traffic. This is also only available on Internet T-1s.

LAN Quality:

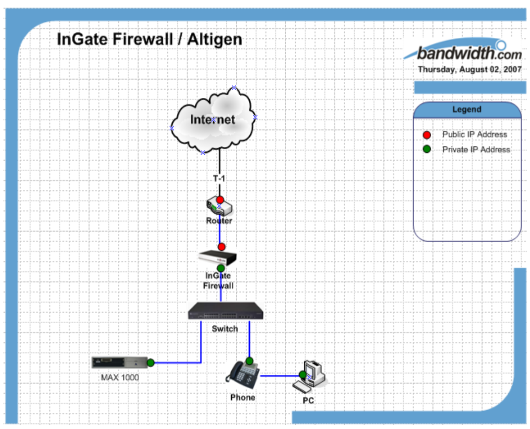
SIP Aware Firewall / ALG: Most people think of a firewall as purely a security device, but most one-way and no-way audio issues can be traced to an improperly configured or non-SIP ALG firewall devices. Bandwidth.com supports the InGate SIParator and the EdgeMarc from Edgewater networks and has tested and certified it to handle both the security needs of the customer network and the ALG needs of the SIP trunking service.

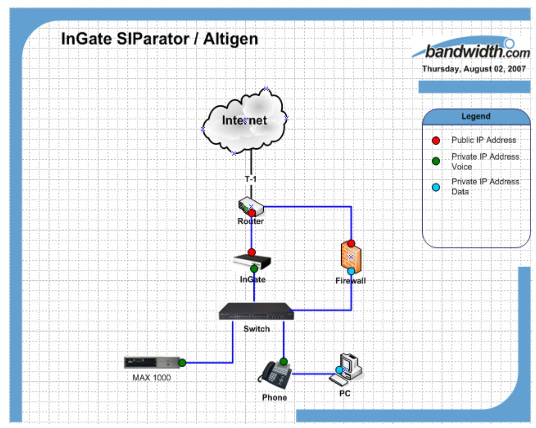
VLANs: VLANs are a standard way to separate virtually the voice and data traffic on the customer LAN. This allows for easy prioritization and limits the effect of viruses and other broadcast events on the customer's network from affecting voice quality.

PoE: While PoE provides power to the phones, it also cleans power and can reduce or eliminate power-related issues, such as hum and static, on IP phones.

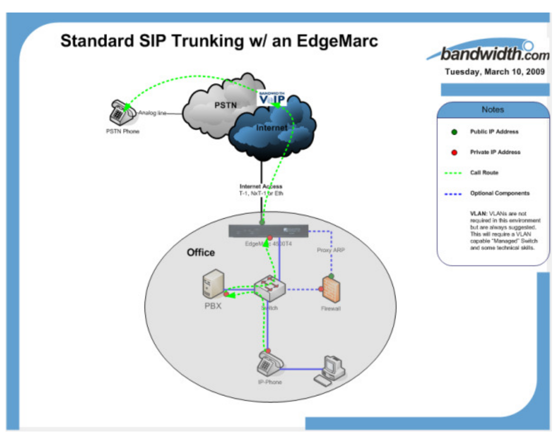
Approved Network Topologies

Ingate Sytems – SIParator/Firewall: The Ingate SIParator is an appliance that provides SIP ALG for NAT traversal. The SIParator is utilized when there is a complex network in place and the implementation requires a standalone VoIP appliance. InGate also has a firewall product with QoS options. For more information, visit www.ingate.com.



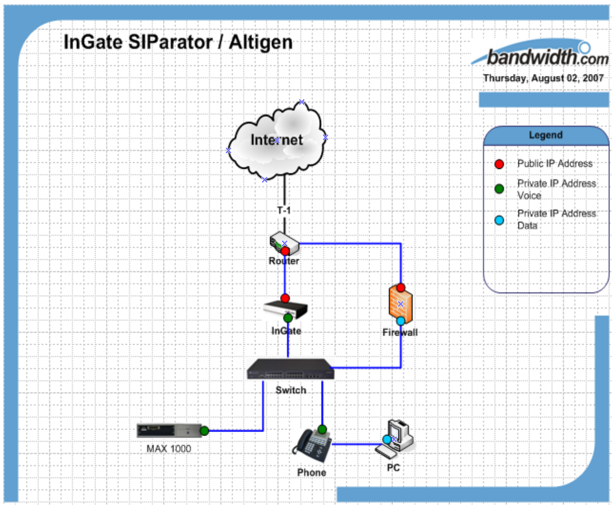


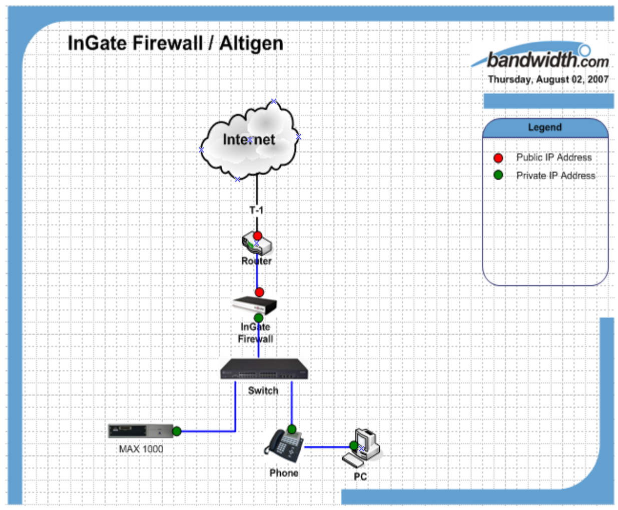
Edgewater Networks - EdgeMarc Appliance: The EdgeMarc Network Appliance is a Router with either T-1 or a 10/100 WAN interface. It also has four switched 10/100 LAN interfaces. The EdgeMarc will provide SIP ALG for NAT Traversal, Traffic Shaping and a fully functional firewall. The EdgeMarc will also resolve Bandwidth.com's DNS SRV record to allow for SIP survivability. Typically, the EdgeMarc is used when a T-1 router is needed with Traffic Shaping, but it can reside behind an existing router. For more information, visit www.edgewaternetworks.com.



InGate SIParator Configuration

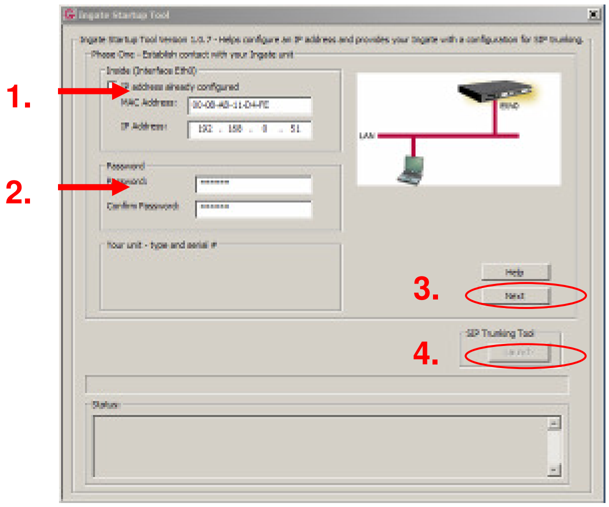
This configuration assumes that the SIParator is set at the factory default. If the SIParator is coming from Bandwidth.com, it will most likely be pre-configured for your network.



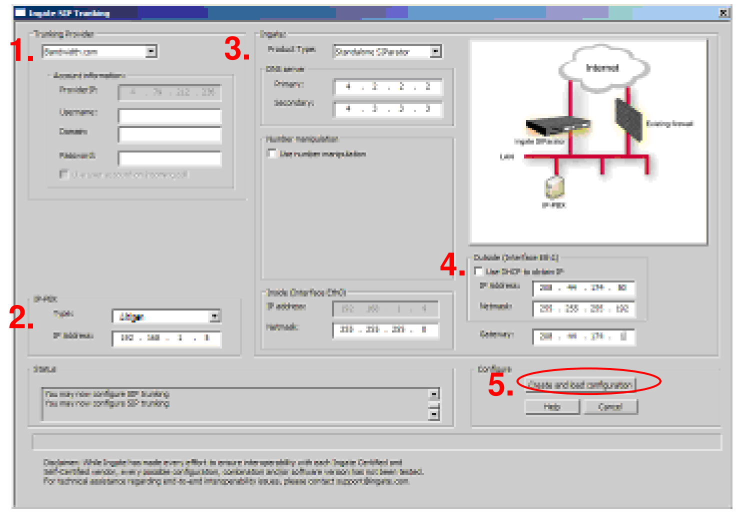


1. If the SIParator was preconfigured by Bandwidth.com, skip to step 10.
2. Go to **"www.ingate.com"** -> **"Support"** -> **"Register Account"** and register your company.
3. Choose Login to the site and choose **"Register a new unit."** Make sure you have your serial number in front of you.
4. Activate the necessary licenses using **"Activate License"** link. This will download a .lic file for each license file required. If you are downloading more than one license, you will need to rename the files so that they do not overwrite each other. Typical license examples are: SIP trunking (typically comes with SIParator when delivered by Bandwidth.com), additional call traversal licenses, QoS, and SIP-Aware Firewall.
5. Next you will need to download the latest **"InGate Startup Tool"**, a link is provided in the InGate portal.
6. Once the InGate Startup Tool is downloaded, please install it on the computer that will be configuring the InGate.
7. Connect the laptop or computer with the InGate Startup Tool to the Eth0 port on the SIParator using a standard CAT5 Ethernet cable.
8. After connecting the SIParator, double-click on the  on your desktop. Insure the laptop or computer being used is on the same LAN subnet as the SIParator.
9. Plug the SIParator into a power source.

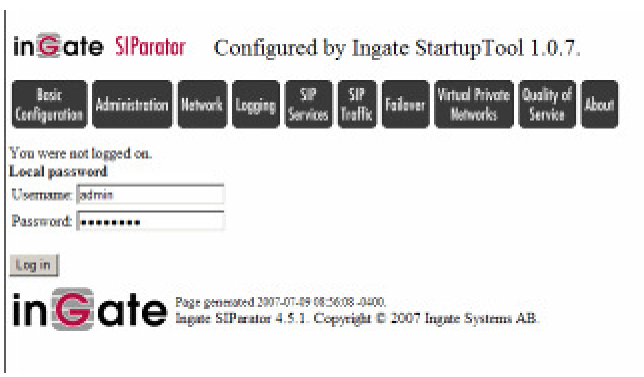
1st Screen



2nd Screen

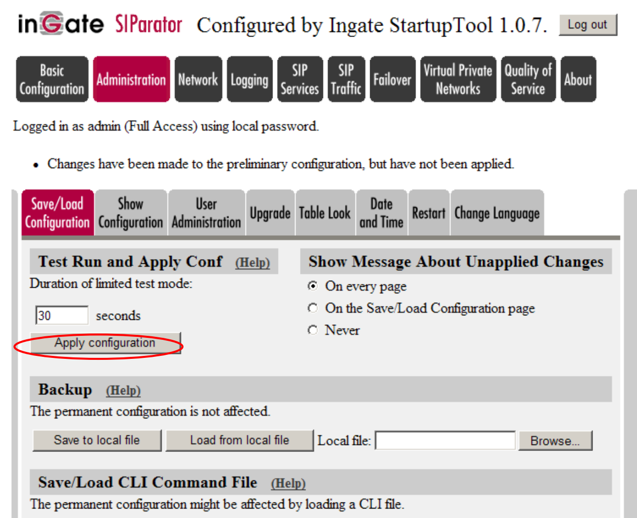


Web GUI Login



Apply Configuration

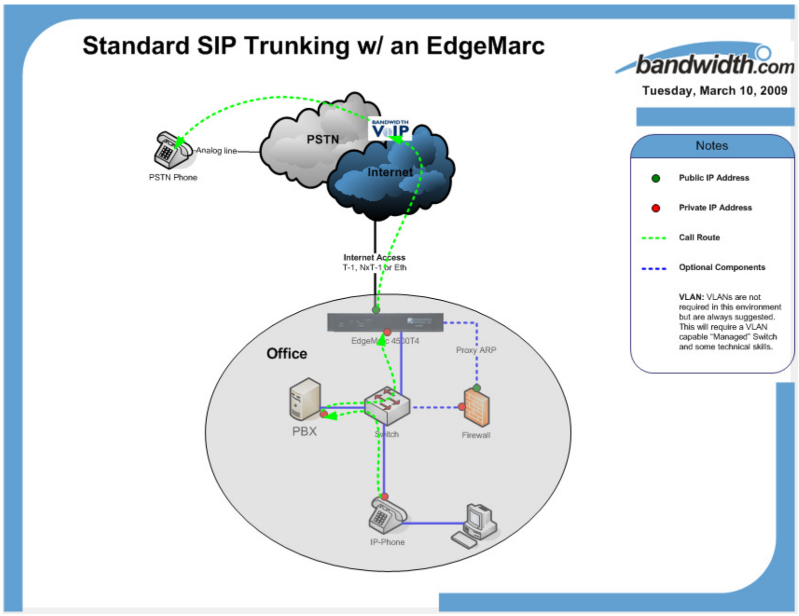
(this is a 2 step process; be sure to confirm!)



10. Once the setup is complete, connect the WAN network to the **Eth1** port and the LAN network (same as PBX) to the Eth0 port of the SIParator.

EdgeMarc Configuration

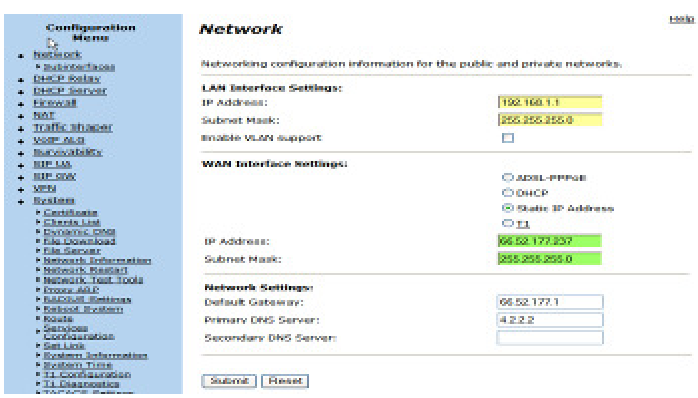
This configuration assumes that the EdgeMarc is set at the factory default. If the EdgeMarc is coming from Bandwidth.com, it will most likely be pre-configured for your network.



1. Connect and configure PC to the LAN subnet to which the Edgemarc is connected.
2. Log into the EdgeMarc (Factory Default) by opening a web browser and entering http://192.168.1.1
   Once you have clicked on the link, you will be prompted for a login:

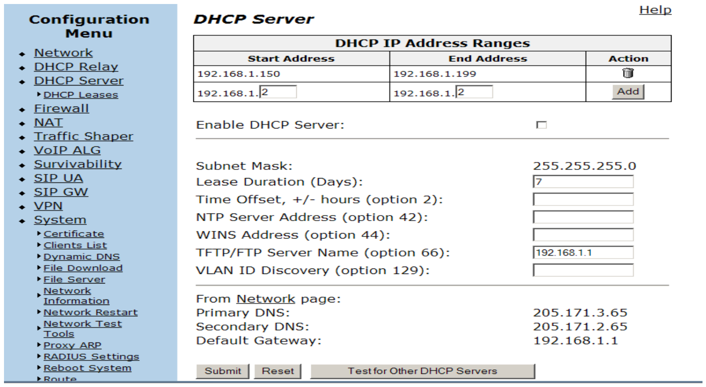
   Login: root
   Password: default

3. Click on the **"Network"** link.
4. Configure the **"WAN and LAN Interface Settings"**



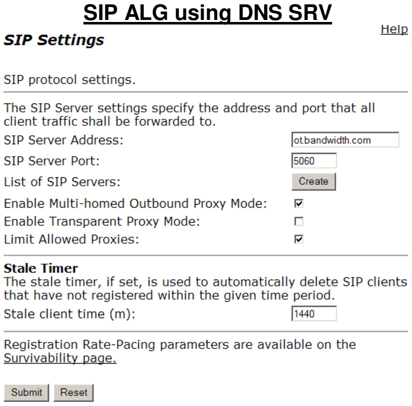
5. Select **"Submit"**

6. Click on the **"DHCP Server"** link and disable the DHCP server;

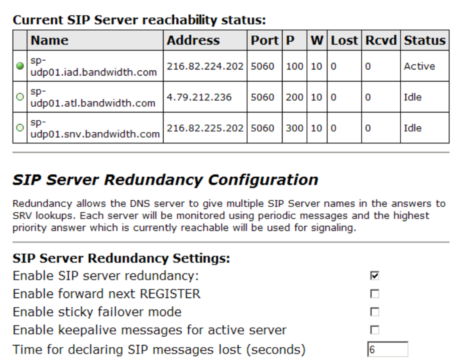


7. Select **"Submit"**
8. Click on the **"VoIP ALG"** link
9. Select **"SIP"**
10. Configure **"SIP Server Address"** and port (typically port 5060 for SIP). Add the DNS SRV Record: **ot.bandwidth.com** under **"SIP Server Domain Name"** or click **"Add Row"** and enter the following two SIP Server Addresses: **216.82.224.202** and **216.82.225.202** using port 5060 for both. The EdgeMarc will direct all Outbound calls to the primary SIP server address and failover to the secondary should a service outage go into effect. Make sure to check the box marked, **"Enable Multi-homed Outbound Proxy Mode"**.

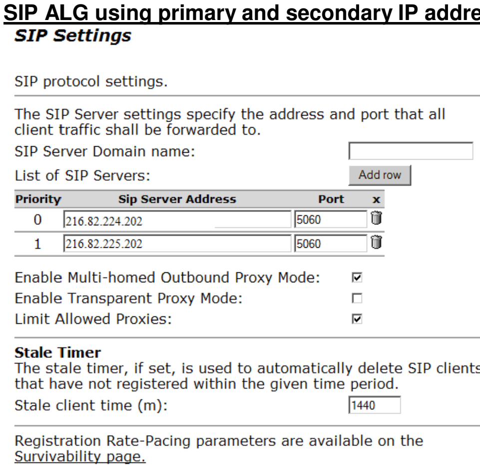
SIP ALG using DNS SRV



Check that DNS SRV works in the Survivability section

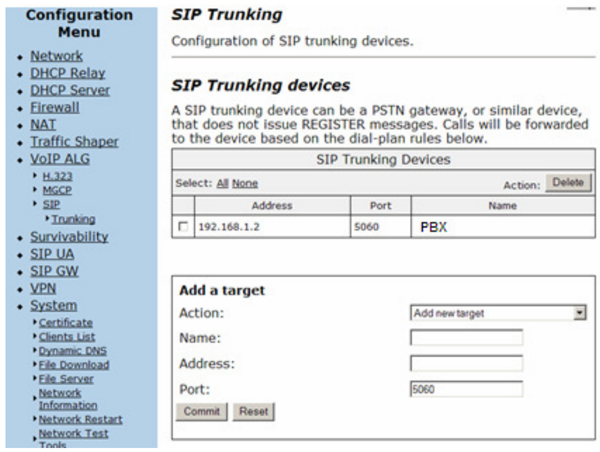


SIP ALG using primary and secondary IP addresses

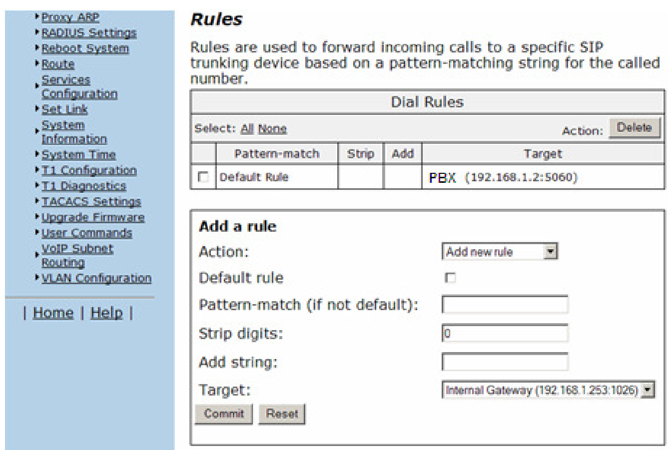


11. Select **"Submit"**

12. Select **"SIP Trunking Page"**
    The SIP Trunking device needs to point to the IP-PBX's IP address.



13. Select **"Commit"**
14. Select **"SIP Trunking Page – Rules"**
    Add a default call routing rule, which will route any inbound call to the PBX



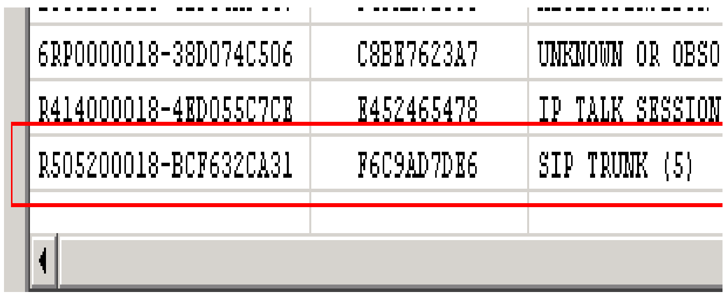
15. Select **"Commit"**
16. To allow for **Remote Phone Registration** and Calling make the following changes
    a. Configuration Changes → NAT – enter the following NAT statements
        i. tcp;WAN_IP / WAN_SUBNET - 10032>ALTIGEN_IP-10032
        ii. tcp;WAN_IP / WAN_SUBNET - 10064>ALTIGEN_IP-10064
        iii. udp;WAN_IP / WAN_SUBNET - 10060>ALTIGEN_IP-10060
    b. Configuration Changes → System → User Commands – enter the IP Tables commands
        i. iptables -I FORWARD 1 -i eth1 -d ALTIGEN_IP -p udp --dport 49152:49211 -j ACCEPT
        ii. iptables -A PREROUTING -t nat -p udp --dport 49152:49211 -d WAN_IP -j DNAT --to ALTIGEN_IP:49152-49211
        iii. iptables -A POSTROUTING -t nat -p udp -s ALTIGEN_IP --dport 49152:49211 -j SNAT --to WAN_IP
        iv. iptables -t nat -A POSTROUTING -p udp -d ALTIGEN_IP --dport 49152:49211 -s LAN_BLOCK 192.168.1.0/24 -j SNAT --to LAN_IP

AltiGen Setup

Turn on SIP Trunking

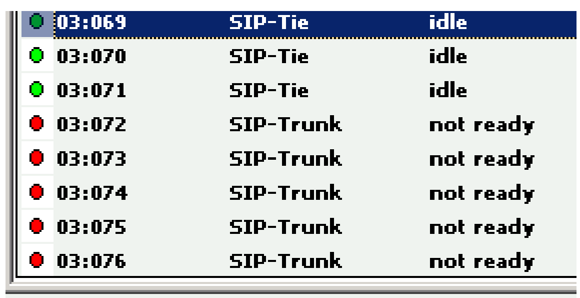
1. Obtain a SIP trunk license from AltiGen.

2. Register the license using AltiGen's online license registration procedure.



3. Reboot AltiServ.

4. After the system is rebooted, **SIP-Trunk** should show as "not ready" in Trunk View.

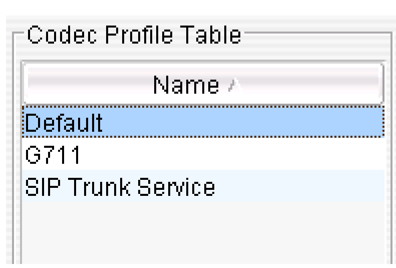


Configuring in AltiEnterprise Manager

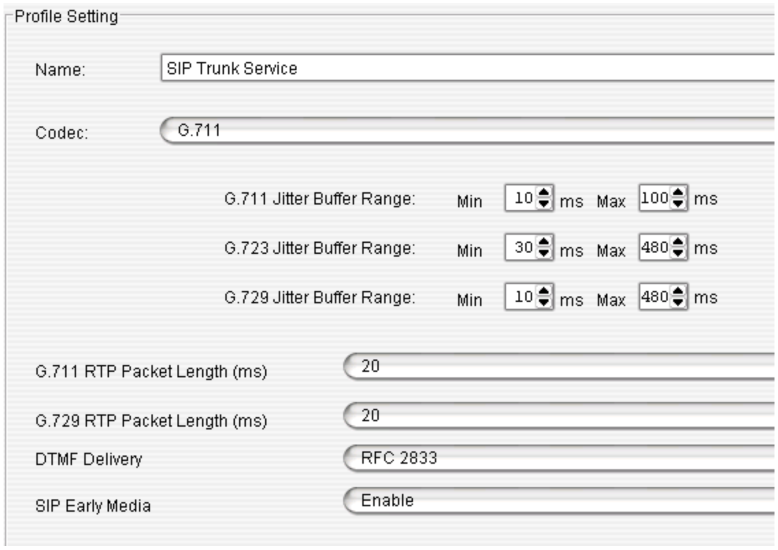
1. Choose **AltiWare Administrator > VoIP > Enterprise Network Management.** This opens AltiEnterprise Manager.

2. In AltiEnterprise Manager, click the **Codec** button.

3. Add a codec profile. In this example, we name the new profile "SIP Trunk Service".

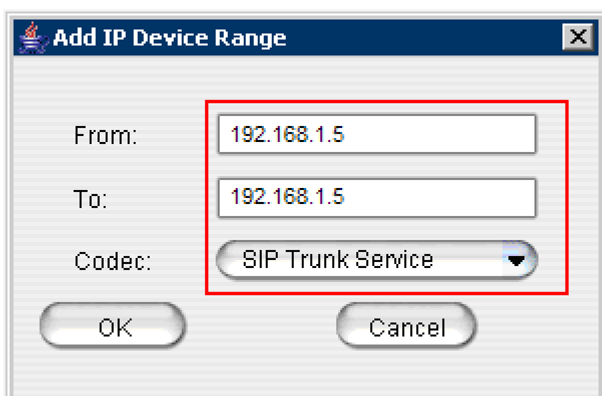


4. Set the following:
   - **Codec** = G.711
   - **DTMF Delivery** = RFC 2833
   - **SIP Early Media** = Enable



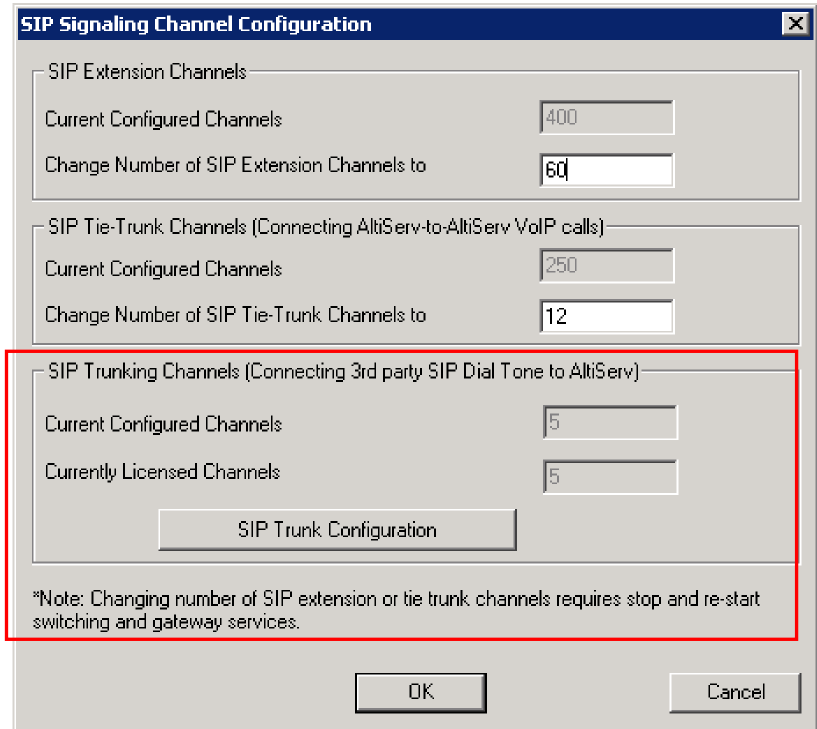
5. In AltiEnterprise Manager, go to **Servers** button > **IP Codec** tab > **IP Device Range** panel. Add a range, using the SIP Proxy IP address that you got from your service provider. Use the LAN side address of the InGate or EdgeMarc. In this example, the IP address is 192.168.1.5. The range we add is:

   - **From**: 192.168.1.5
   - **To**: 192.168.1.5
   - Set **Codec** as “SIP Trunk Service”.



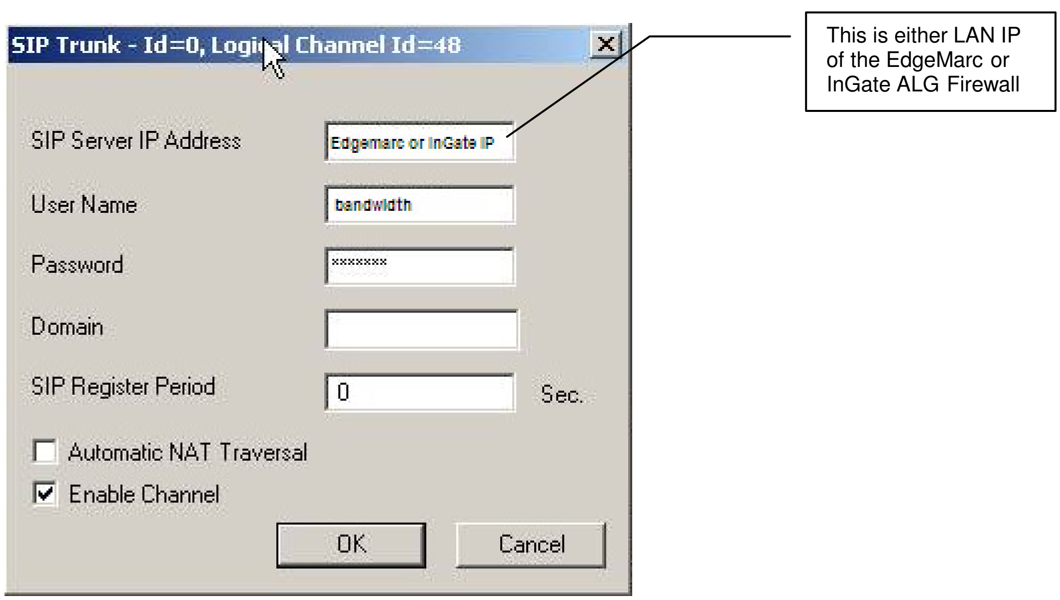
Configuring in AltiWare Administrator

1. Under “SIPSP” board configuration, the number of configured channels and licensed channels are displayed. In this example, there are 5 SIP trunk channels.



2. Click the SIP **Trunk Configuration** button. The SIP Trunk Configuration dialog box opens.

3. Highlight an entry in the dialog box, and then click **Edit**. The following dialog box opens:

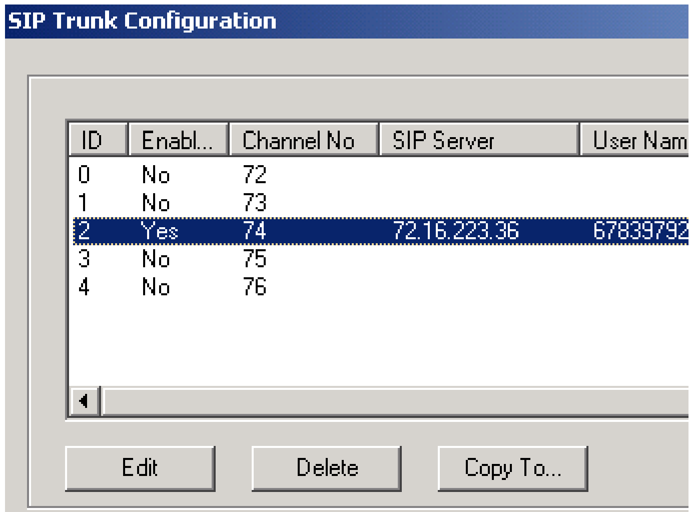


4. Enter the SIP Server IP Address, User Name <bandwidth>, Password <bandwidth>, and SIP Register Period <0>.

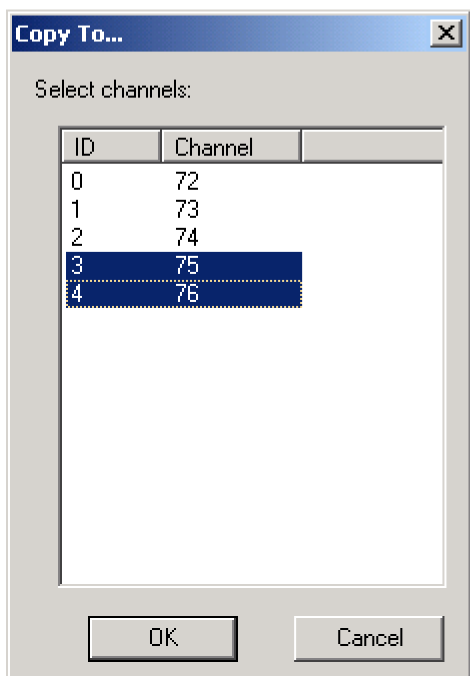
5. Uncheck **Automatic NAT Traversal** and check **Enable Channel**.

If multiple SIP trunks are subscribed, follow the steps below to copy the same settings to other SIP channels.

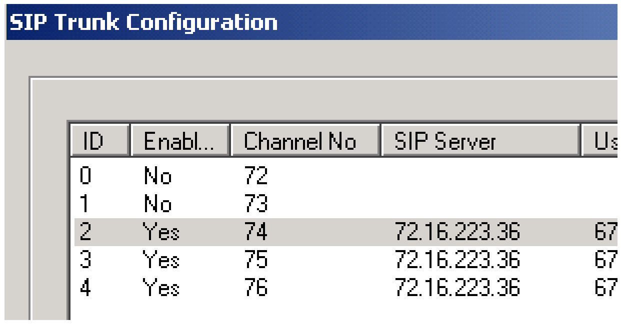
1. Highlight the source entry.



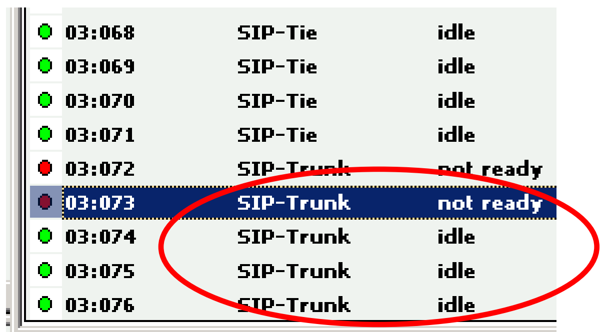
2. Click the **Copy To** button.



3. Highlight the destination channels and click **OK**.



In Trunk View, the SIP trunks will be displayed as **idle**. The SIP trunks are ready to use.



Configuring SIP trunks (trunk access code, in call routing, outcall routing, and so on) is similar to configuring analog trunks or T1/PRI trunks. See AltiWare Administrator online Help under “Trunk Configuration,” or see the *AltiWare Administration Manual*, “Trunk Configuration” chapter.

To work behind the EdgeMarc you will also need to make the following change to your Windows Registry:

HKEY_LOCAL_MACHINE\SOFTWARE\AltiGen Communications, Inc.\AltiWare\Service Providers\SIPSP\EnableStdRouteSet

Needs to be changed from 0 to 1 when using the EdgeMarc. It will need to stay 0 when behind an Ingate.

Numbering Info

Remember that Bandwidth.com wants to see a "+" and a "1" on all outgoing calls, so this must be planned for. Also, we will be sending you a "+" and a "1" on every call, so make sure to plan for this when setting up your PBX. The InGate can add the + for E.164 numbering for you; see this will be done in Number Manipulation in the Setup Tool.

Operator must be sent as a plain 0 (no 1)
911 must be sent as +1911
411 must be sent as +1411
International calls are to be sent without 011. Example: +442151245
Local and Long Distance Calls; Example: +19192971100

Initial Testing and Troubleshooting

Initial Test Plans:

If you run into an issue with any of these tests, consult the troubleshooting chart bellow:

1. Test an Outbound call to a Local Number. Check for Ringback, 2-way Audio, and Quality.
2. Test an Outbound call to a Long Distance Number. Check for Ringback, 2-way Audio, and Quality.
3. Test an Outbound call to an International Number. Check for Ringback, 2-way Audio, and Quality.
4. Test a Long Outbound call past 15 minutes.
5. Test multiple call concurrencies on Outbound calls. Setup multiple calls to PSTN
6. Test an Outbound Call to Operator "0"
7. Test an Outbound Call to Directory Assistance "411"
8. Test a 911 Call (PLEASE IDENTIFY TO THE OPERATOR THAT THIS IS A TEST!!!!!)
9. Test an Inbound call to an internal DID. Check for Ringback, 2-way Audio, and Quality.
10. Test an Inbound call to Auto-Attendant. Check DTMF and Audio Quality.
11. Test Transferring calls off-site.
12. Test an Outbound call to an Auto-Attendant and verify DTMF.

Troubleshooting Guide:

Issue	Cause	Remedy
No Calls IN/Out	InGate/EdgeMarc Configuration	Check InGate/EdgeMarc Configuration
	PBX Configuration	Check PBX Configuration
	Unqualified IP Address	Note WAN IP Address and Contact Bandwidth.com
No Calls Out	PBX Configuration	Check PBX Configuration
	Unqualified IP Address	Note WAN IP Address and Contact Bandwidth.com
No Calls In	PBX Configuration	Check PBX Configuration
	Unqualified IP Address	Note WAN IP Address and Contact Bandwidth.com
One-Way Audio	InGate/EdgeMarc Configuration	Check InGate Configuration
Echo	Excessive Delay	Check LAN and WAN for high latency
	Echo Cancellation Issue on PBX	Check Echo settings and/or consult PBX
	Echo Cancellation Issue on Bandwidth.com	Call Bandwidth.com and report
Call Dropping	Internet Access Issues	Call Internet Access Provider
	Extreme Latency on LAN	Check Latency on LAN
	SIP issue	Call Bandwidth.com
Static or HUM on Phones	Power issue	Check power if using AC should not be issue in PoE
Missing Parts of Words	Packet Loss or Latency on LAN	Check LAN
	Packet Loss or Latency on WAN	Check with Internet Access Provider
	Jitter Buffer Configuration	Check with PBX

Notes: