AltiGen KB

Field Alert #211 Bandwidth.com certified as SIP-Trunk service provider

AltiGen has tested and certified Bandwidth.com as a SIP-Trunk service provider for AltiGen IP PBX systems. **Introduction** From an 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 4.79.212.236 or DNS sp-udp01.atl.bandwidth.com Secondary, SIP proxy server IP address 216.82.224.202 or DNS sp-udp01.iad.bandwidth.com Trunking DID(s): The DID(s) are forwarded to the Public WAN IP address(s), DNS or DNS SRV records of the PBX or InGate SIParator.

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 MAX1000

AltiWare 5.1 (ACC or ACM) version 5.1.0.1412 or above

Any phones that work with Altigen MAX1000

References

Ingate SIParator Getting Started Guide Bandwidth.com SIP Trunking Service Description AltiGen AltiWare Administrator's Guide

Installation Worksheet

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

WAN Side:

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

LAN Side:

LAN IP of SIParator:	
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:

Device:	
Serial #:	
MAC:	
Firmware Version:	

Call Quality Management

Internet Access Quality

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 ADSL, SDSL and T-1, and the voice quality is excellent. The main issue with this method is that the voice side of the network only allows voice traffic on specific ports. This can be problematic for Teleworker, since it is not standard SIP traffic. At this time, Teleworker must reside on the data side. There is also a proprietary router provided with this solution. The InGate can still be used behind the router for NAT traversal.

Packet Prioritization: This method uses priority marked queues or packets to ensure quality across the carrier

Internet backbones. Bamdwidth.com supports two of these services today.

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 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 stand-alone VoIP appliance. InGate also has a firewall product with QoS options for the Packet Prioritization Internet Access options Bandwidth.com supports from Qwest and Sprint. For more information, visit www.ingate.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.

If the SIParator was preconfigured by Bandwidth.com, skip to step 10. Go to www.ingate.com > Support > Register Account and register your company. Choose Login to the site and choose Register a new unit. Make sure you have your serial number in front of you. 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. Next you will need to download the latest InGate Startup Tool, a link is provided in the InGate portal. Once the InGate Startup Tool is downloaded, please install it on the computer that will be configuring the InGate. Connect the laptop or computer with the InGate Startup Tool to the Eth0 port on the SIParator using a standard CAT5 Ethernet cable. After connecting the SIParator, double-click the on your desktop. Insure the laptop or computer being used is on the same LAN subnet as the SIParator. Plug the SIParator into a power source.

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.

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.

5. Bandwidth.com SIP proxy uses a private IP address, so don’t configure AltiServ behind NAT, unless

you are using remote IP phones.
If you are using remote IP phones, configure AltiServ behind NAT. Make sure the Bandwidth.com SIP proxy IP address is configured as a private network under AltiWare Admin > VoIP > Enterprise Network Management > IP Networks > IP Network. Then, for information about configuring AltiServ behind a firewall / NAT router, see AltiWare Administrator's online Help: Index > AltiServ behind NAT > configuring.

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. In this example, the IP address is 72.16.223.36. The range we add is:

§ From: 72.16.223.36

§ To: 72.16.223.36

§ 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, Password, and SIP Register Period. This account information is obtained from Bandwidth.com.

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.

Numbering Info

Remember that Bandwidth.com wants to see a “+” and a “1” on all outgoing calls, so this must be planned for. Also, Bandwidth.com 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 below.

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

Troubleshooting Guide

https://know.altigen.com/questions/838/