Application Note: Enterprise PBX with SIP Trunking and Remote Users

Summary

This application note provides a description of how to deploy an enterprise PBX with SIP Trunking and remote users using EdgeMarc Network Services Gateway. This solution enables remote users to connect to the company PBX and make outbound calls through the SIP Trunk just like the locally connected users to the PBX.

Introduction

For some time now, enterprises with legacy as well as IP PBXs have been enjoying the benefit of less expensive SIP trunks for outbound communications. In addition to being less expensive as compared to PSTN, SIP trunks also provide a way for these enterprises to use Service Provider (SP) owned PSTN gateways instead of using their own, resulting in additional cost savings.

Enterprises with IP PBXs are now considering connecting their road warriors and remote users to the company PBX as a natural extension to the cost saving effort. This way the remote users can not only stay in touch with their colleagues at work from anywhere in the world, but they can also cost effectively stay in touch with the key people outside the company through the SIP trunk.

Deployment and Configuration

The solution can be deployed like any other SIP Trunking solution* using the EdgeMarc with the following exceptions needed to connect the remote users:

- An IP PBX with two (2) Ethernet interfaces must be used (for demonstration purposes Asterisk will be used). One interface will be connected to the LAN and the other with a public IP address will be connected to the Internet through the EdgeMarc.

Figure 1 below shows an example deployment of Asterisk with SIP Trunking through the EdgeMarc which provides the following functions among other things:

- Rate pacing of registrations to the Softswitch
- Failover to PSTN in case of the WAN link failure
- MOS scoring
- Prioritized voice packets

The figure shows a typical deployment of an IP PBX (Asterisk in this case) at the headquarters with some IP phones on the company's local LAN. It also shows a remote satellite office with two IP phones behind a NAT device (possibly a gateway router). In addition, it shows a road warrior using a softphone on a laptop with a public IP address shown as <Public IP address 3>.

All IP phones on the LAN will be using the local IP address (192.168.1.254) of Asterisk for registration and the remote phones will be using the public IP address (shown as <Public IP address 2> in Figure 1) of Asterisk.

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*SIP Trunking Application Note can be obtained from the following link:
http://www.edgewaternetworks.com/documentation.htm

Care must be taken to configure the Asterisk properly so that SIP traversal works for the remote phones which are behind a NAT device. In addition, Asterisk must also send periodic OPTIONS messages to the IP phones behind the NAT to keep the NAT mappings for those phones stable.

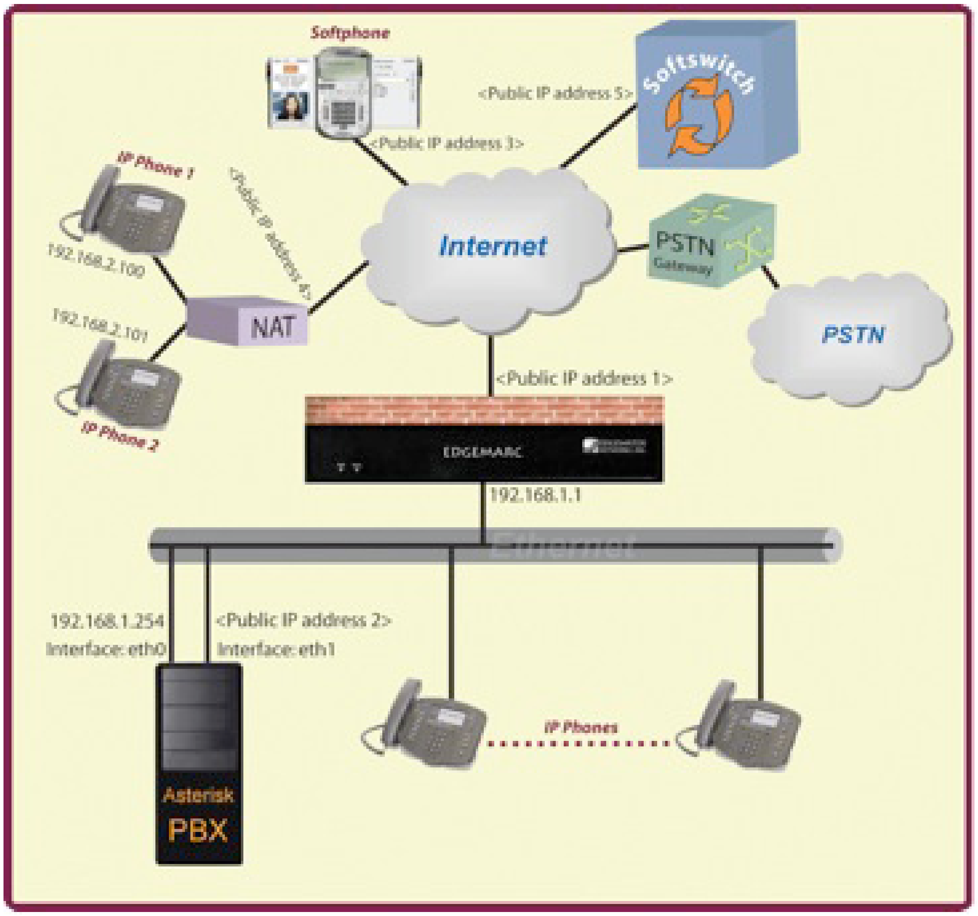

Figure 1: Enterprise PBX with SIP Trunking and remote users

Configuring the EdgeMarc

Configuring the VoIP ALG SIP settings for SIP Trunk

First set the "SIP Server Address" field by going to "Configuration Menu->VoIP ALG->SIP" and setting it to the Fully Qualified Domain Name (FQDN) or IP address of the SIP Trunk provider.

Next go to "Configuration Menu->VoIP ALG->SIP->Trunking" and create a SIP Trunking device with an IP address of 192.168.254 (LAN address of Asterisk as shown in **Figure 1**) and create an inbound "Default Rule" to send all inbound from the ALG to the newly created device unless otherwise specified by another trunking rule.

Note: Please make sure that "**Enable Transparent Proxy Mode**" is not selected.

Setting up Proxy ARP for Remote Phone Registrations

The following Proxy ARP entry must be configured in the EdgeMarc so that the traffic can be routed by the EdgeMarc to the public IP interface (shown as <Public IP Address 2> in Figure 1) of the computer running Asterisk:

Configured Proxy ARP Entries:

IP Address/Bitmask	On IF	Proxy on IF	Gateway
<Public IP Address 2>/32	LAN	WAN	12.48.202.1

In addition, the command below has to be added to the user commands. To achieve that, select "User Commands" from the "System" submenu of the "Configuration Menu" and enter the following commands into the command window and hit submit when done:

- iptables -I FORWARD -i eth1 -o eth0 -d <Public IP Address 2>/32 -p udp --dport 5060 -j ACCEPT

Configuring the IP Phones

All IP phones on the local LAN should be configured to register with Asterisk on 192.168.1.254 and all the remote phones should be configured to register with Asterisk on <Public IP Address 2>.

Configuring Asterisk

The general section must contain a register statement to register with the SIP Trunk provider. The syntax of register is as follows: register => user[:secret[:authuser]]@host[:port][/extension]. Make sure that the host in the register statement is the same as the LAN address of the EdgeMarc.

When creating the channel definition for the SIP Trunk, set host to the LAN address of the EdgeMarc like host=192.168.1.1. Furthermore add the following statement in the definition: fromuser=<user name of SIP Trunk>

Asterisk needs to be in the media path for each remote phone behind NAT. This is achieved by putting the following statement in the channel or extension definition for each phone:

canreinvite=no

Asterisk must also send periodic OPTIONS messages to the remote phone behind a NAT device, which can be achieved by putting the following command in the channel or extension definition of the phone in the sip.conf file:

qualify=yes

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So for a remote phone behind a NAT device, the following lines must be added to the channel or extension definition in the sip.conf file:

```
canreinvite=no
qualify=yes
nat=yes
```

Configuring the Firewall on the Computer running the Asterisk

Program the Linux firewall on the computer running Asterisk to only allow Asterisk related traffic along with any other traffic needed such as SSH, Telnet, and etc.

For more information please contact sales@edgewaternetworks.com or visit us at www.EdgewaterNetworks.com

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