Configuring an EdgeMarc for SIP trunking with an IP PBX

This document describes the steps needed to configure an IP PBX behind the EdgeMarc which is pointing to a SIP Trunk provider on the WAN.

Please note that this solution documents the basic configuration needed in the PBX and that the requirements of your specific SIP trunking environment may require modifications to the configuration steps provided in this document.

There are three modes of operations that are dicussed in this article.

A) The IP PBX has a static IP address and *is able* to support REGISTERS. It has to register a DID (usually the main trunk DID). Upon a successful registration, the SIP trunk provider will then route all numbers in that trunk group to the EdgeMarc WAN. The EdgeMarc will then forward all calls (using the dial rules) to the PBX defined in the Trunking page.   - page 2 - 9

B) The IP PBX has a static IP address but *is NOT able* to send REGISTERS. The SIP trunk provider doesn't need any Registrations and the provider statically assigns the EdgeMarc's WAN IP as the trusted IP to route calls to and accept calls from.   - page 2 - 9

C) The IP PBX has a static IP address but *is NOT able* to send REGISTERS. However the SIP trunk provider requires a Registration and the EdgeMarc has to Register on behalf of the PBX.   - page 10 - 14

Prerequisites and Assumptions:

SIP trunking information provided by the VoIP service provider:
• SIP proxy server IP address or DNS name.
• Auth name and password for Pilot DID.

EdgeMarc VOS 9.12.4 or later (for Scenario A and B)
EdgeMarc VOS 10.2.4 or later (for Scenario C)

NOTE: The GUI in VOS 13.x is different from the screenshots
in this article but the concepts are the same.

The example in this article assumes that no SIP header manipulation is needed.
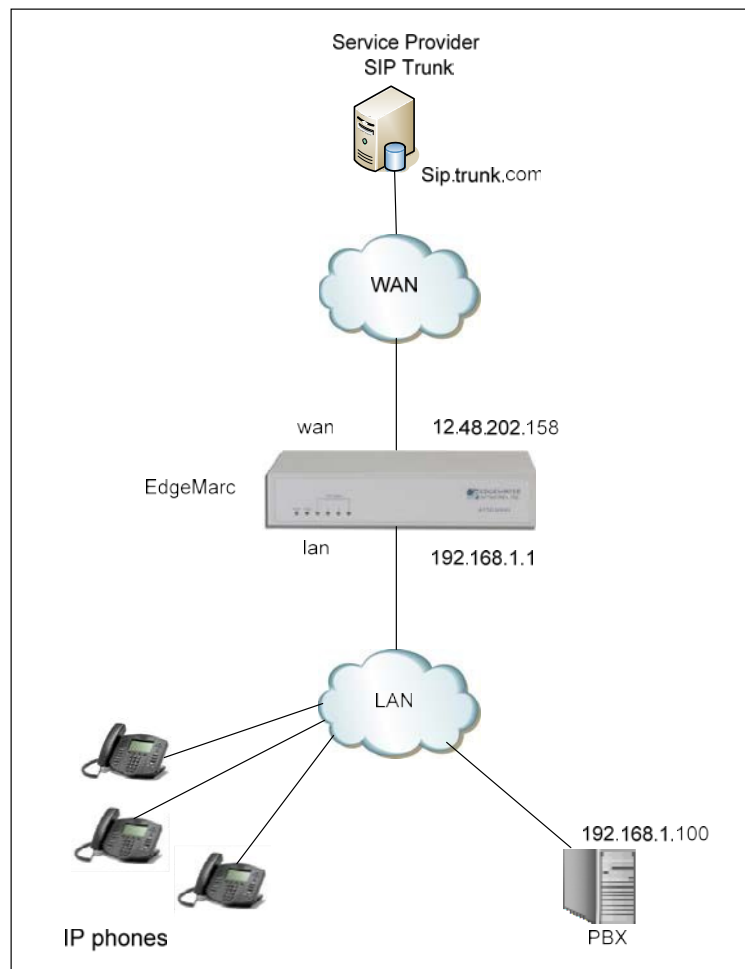
Sample Network Diagram

Fig1 : Network diagram

In this example, there is an IP PBX on the LAN that is configured with 192.168.1.100 and the EdgeMarc LAN is 192.168.1.1. The WAN interface of the EdgeMarc is 12.48.202.158 and the VoIP provider is providing a SIP trunk and it's address is sip.trunk.com.

Configuring the EdgeMarc for Scenario A and B

The steps below assume you are starting from the factory default settings. Please refer to the EdgeMarc VOS user guide for more detailed information on the different configuration steps.

A1. Connect and configure a PC to the LAN port 4 on the EdgeMarc.
   Configure your PC NIC settings
   - IP Address: 192.168.1.2
   - Subnet Mask: 255.255.255.0
   - Default gateway: 192.168.1.1

A2. Log into the EdgeMarc (factory default) by opening a web browser and entering http://192.168.1.1
   - Username: root
   - Password: default

A3. Click on the "**Network**" link

A4. Configure the "**WAN and LAN Interface settings**".
   In this example, the LAN IP is 192.168.1.1 and the WAN IP is 12.48.202.158

   Please note: The IP address used for the LAN interface settings must be used later when configuring the Avaya PBX as the service provider SIP trunk address. The PBX will forward all SIP trunking traffic to the LAN interface of the EdgeMarc.

A5. Select "**Submit"** when done.
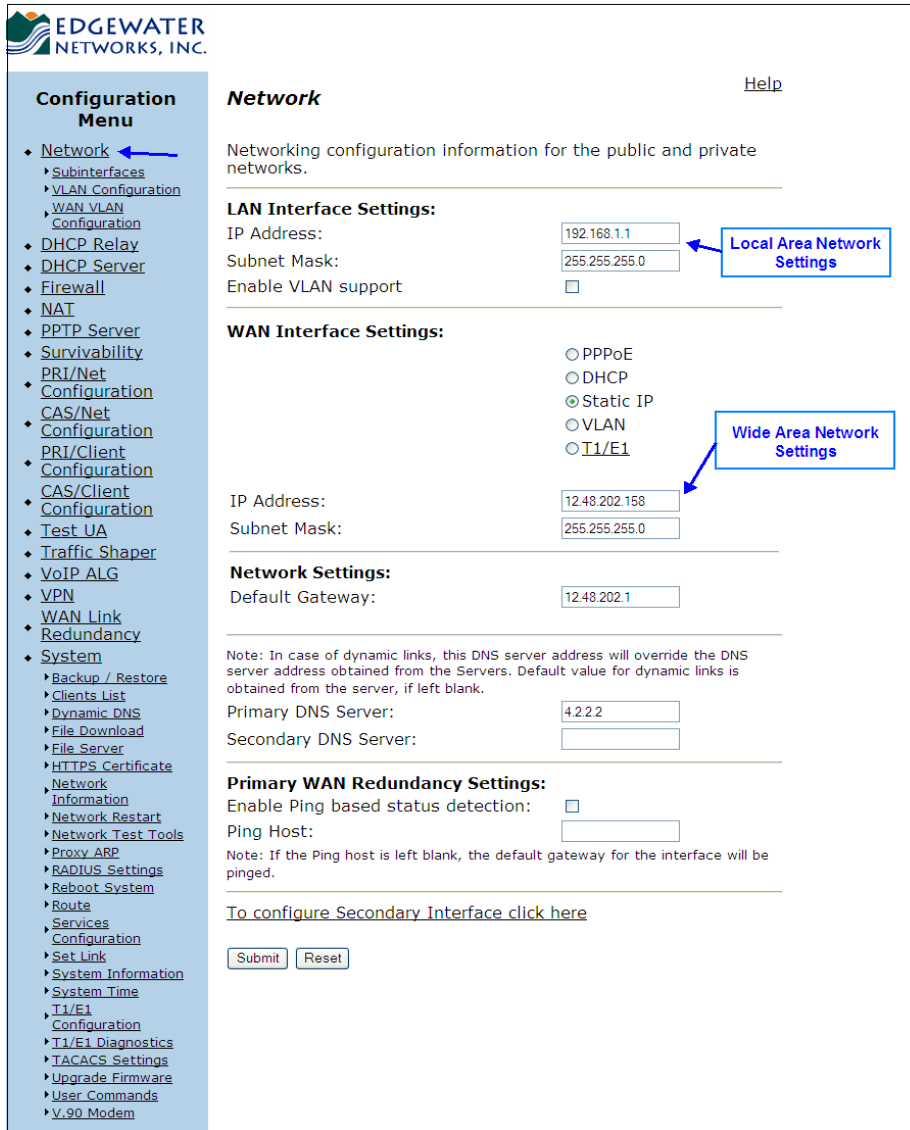
Fig 2: Network page

A6. Next, click on the "**VoIP ALG**" link

A7. Click on the "**SIP**" link

A8. Configure "**SIP Server Address**" and "**SIP Server Port**". The "SIP Server Address" is provided by the SIP trunking service provider.

In this example, the SIP trunk provider is 'sip.trunk.com'

The SIP Server Port is usually port 5060 however you may receive different instructions from your SIP trunking service provider. The EdgeMarc will forward all outbound calls to the SIP Server Address and Port. It will also expect to receive all inbound calls using the SIP Server Address and Port.

A9. Enable the "**Limit Outbound to listed Proxies / SIP Servers**" and "**Limit Inbound to listed Proxies / SIP Servers**" checkbox (Optional)

Note: This option will ensure that the EdgeMarc only forwards outbound / inbound SIP messages received from the IP PBX to the IP address configured in the SIP Server Address field (ie. SIP trunking service provider). This filtering feature is only one part of a multi-tiered security plan that should be implemented when using SIP trunking services. Other common techniques include configuring the IP PBX to restrict international dialing to authorized users, configuring passcodes for international dialing, disabling the ability to "zero" out of IVR or voicemail systems to place phone calls and restricting the use of the SIP trunk to only LAN side phones connected and registered to the IP PBX.

A10. Select "**Submit"**

EDGEWATER
NETWORKS, INC.

Configuration Menu

- Network
- DHCP Relay
- DHCP Server
- Firewall
- NAT
- PPTP Server
- Survivability
- PRI/Net Configuration
- CAS/Net Configuration
- PRI/Client Configuration
- CAS/Client Configuration
- Test UA
- Traffic Shaper
- VoIP ALG ←
  - ▸ MGCP
  - ▸ SIP ←
    - ▸ Trunking
    - ▸ Adv. Trunking
- VPN
- WAN Link Redundancy
- System
  - ▸ Backup / Restore
  - ▸ Clients List
  - ▸ Dynamic DNS
  - ▸ File Download
  - ▸ File Server
  - ▸ HTTPS Certificate
  - ▸ Network Information
  - ▸ Network Restart
  - ▸ Network Test Tools
  - ▸ Proxy ARP
  - ▸ RADIUS Settings
  - ▸ Reboot System
  - ▸ Route
  - ▸ Services Configuration
  - ▸ Set Link
  - ▸ System Information
  - ▸ System Time
  - ▸ T1/E1 Configuration
  - ▸ T1/E1 Diagnostics
  - ▸ TACACS Settings
  - ▸ Upgrade Firmware
  - ▸ User Commands
  - ▸ V.90 Modem

Help

SIP Settings

SIP protocol settings.

The SIP Server settings specify the address and port that all client traffic shall be forwarded to.

SIP Server Address: `sip.trunk.com`

SIP Server Port: `5060` → **SIP Trunk Provider info**

Use Custom Domain: ☐

SIP Server Domain: [   ]

List of SIP Servers: [Create]

Enable Multi-homed Outbound Proxy Mode: ☐

Enable Transparent Proxy Mode: ☐

Limit Outbound to listed Proxies / SIP Servers: ☑ ←

Limit Inbound to listed Proxies / SIP Servers: ☑ ←

Stale Timer
The stale timer, if set, is used to automatically delete SIP clients that have not registered within the given time period.

Stale client time (m): `1440`

Registration Rate-Pacing parameters are available on the Survivability page.

[Submit] [Reset]

Fig 3: SIP Settings page

A11. Select "**VoIP ALG**" link.
A12. Select "**VoIP ALG -> SIP -> Trunking**" link.
A13. Add a trunking device for the IP PBX by configuring the "**Name**", "**Address**", and "**Port**".
A14. The Address and Port need to match the IP address and SIP port configured in the IP PBX.
A15. Select "**Commit**"
A16. The IP PBX should appear in the SIP Trunking Devices table (as shown below)

Figure 4: SIP Trunking devices table

A17. On the same page, scroll down to the Dial Rules Section.
A18. Select "**Action - Add new rule**"
A19. Set "**Type:**" to "**Inbound**"

A20. Select the "**Default Rule**" checkbox
A21. Set the "**Trunking Device**" to be the name and IP address of the PBX.
A22. Select "**Commit**"

The Dial Rules page should look like Figure 5.
This will create a routing rule that will direct all inbound calls received on the WAN interface of the EdgeMarc to the PBX.

Figure 5: Dial Rules table

A23. Configure the rest of the EdgeMarc as per the manual.

Other settings on the EdgeMarc are optional for basic SIP trunking to operate, but we recommend configuring at least the following:

- Traffic Shaping and CAC: Adjust these values according to your WAN bandwidth.
- MOS monitoring and capturing it to EdgeView
  e.g.: Point the EdgeMarc to EdgeView to capture all the MOS scoring on the SIP trunking calls on the LAN and WAN side. This will allow you to "demarc" VoIP quality issue between LAN and WAN.

Configuring the PBX

A 24. On the PBX, configure it's SIP server to point to the LAN IP of the EdgeMarc (192.168.1.1)

A25. Make sure that the PBX doesn't have a 'behind a NAT' setting. If it does, disable it.

Making and Receiving calls

- **Registration mode**

  If the SIP trunk provider requires a registration from the PBX and the PBX is able to register, the registration sent by the PBX should be received on the lan side of the EdgeMarc, processed by the ALG and will be forwarded out the WAN to the SIP trunk provider. As soon as it's registered, the PBX should be able to make calls. All inbound calls from the SIP trunk provider will be forwarded (according to the Dial Rules – Fig 5) to the PBX.

- **Static mode**

  If the PBX doesn't register, make sure that the provider has statically assigned the WAN IP address of the EdgeMarc in his SIP Server as the trusted IP address to send and receive calls from. All outbound calls from the PBX will be forwarded to the SIP Server from the WAN IP of the EdgeMarc. All inbound calls from the SIP trunk provider will be sent to the WAN IP of the EdgeMarc and will be forwarded (according to the Dial Rules – Fig 5) to the PBX.

Configuring the EdgeMarc for Scenario C

Configure the EdgeMarc using the procedure listed in Step A1 – A10 above.

C1. Select "**VoIP ALG**" link.
C2. Select "**VoIP ALG -> SIP -> Adv. Trunking**" link.
C3. Under the Trunking Devices section, add a trunking device.
- Name : PBX (as an example)
- IP :   192.168.1.100
- Port: 5060
- Click on the "Update" button.

Figure 6: Defining the non-registering PBX in the Adv Trunking page

C4. Next, you would need to define the DID / Authentication name and password that the SIP trunk provider is expecting. (*This is where you define what the EdgeMarc will register on behalf of the non-registering PBX*)

- Add the DID or Pilot DID that needs to register with te SIP trunk provider (4074017663)
- Enable the Register Pilot checkbox.
- Enter the Authentication-name and Password for the Registration.

Figure 7: Information needed to register on behalf of the non-registering PBX in the Trunk page

C5. Next, you need to define the "Actions" for the Trunking Rules. Go to the Trunk section and add a "Action". In this example, an action was defined to route all incoming calls to the PBX.

- Name: incomingcalls (as an example)
- Send to: Select Trunking device and PBX from the pull down menu (this is the device defined in Trunking Device above – Fig 6)
- Click on the Update button.

Figure 8: Defining an Action item for routing calls

C6. Next, you need to define a matching pattern / rule for the calls.
In this example, we are trying to define a default inbound rule that will route all incoming calls to the PBX
- Select a call direction: Inbound
- Select "default"
- For Action: select "incomingcalls" (which was the Action defined in Fig 8)
- Click on the "Update" button.

C7. Finally, click on the Submit button at the bottom of the page to submit all changes made on the Adv. Trunking page.

Figure 9: Defining a pattern match for the dialling rules.

Summary: Here is a summary of the Adv. Trunking page and the various sections.