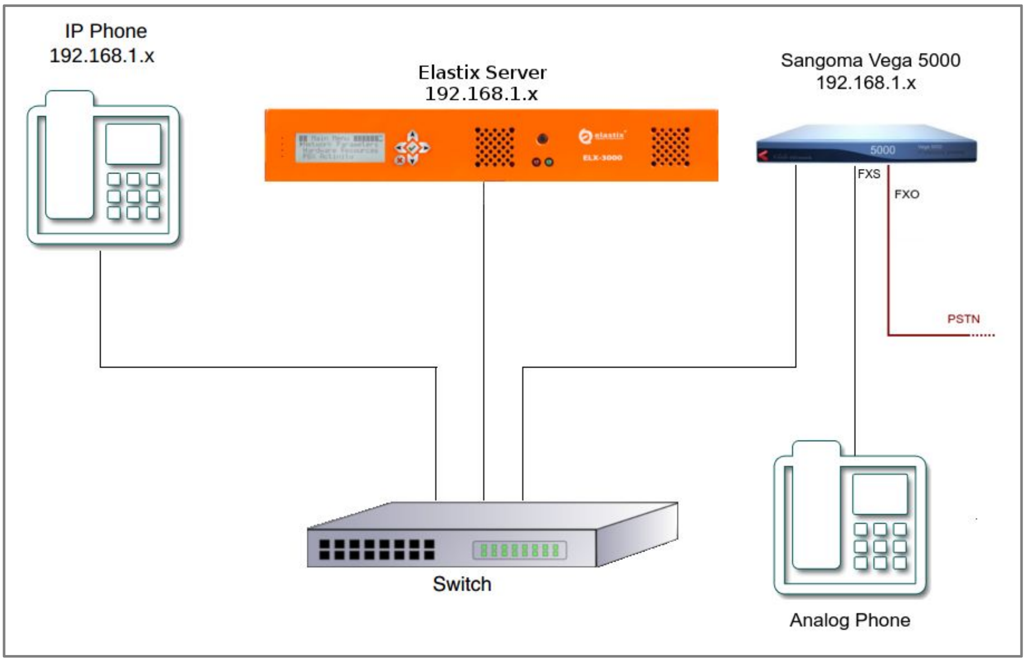
Sangoma Vega 5000 24FXS/2FXO
and Elastix Server
Setup Guide

1.0 Setup Diagram

Figure 1-1 is a setup diagram for a single Vega 5000 analog gateway configuration. We're going to configure a SIP Trunk for communication between the IP Phone and PSTN.

Figure 1-1. Setup Diagram



2.0 Host PC Environment

Table 2-1. Host Server Environment Details

	Description
Hardware Type	Elastix Appliance ELX-Series
Hardware Version	ELX-3000
Software Type	Elastix
Software Version	2.3

3.0 Test Setup Equipment

Table 3-1. Test Setup Equipment

Equipment	Model	Version
IP (SIP) Phone	N/A	N/A
Sangoma	Vega 5000	FW: R088S020
Switch	N/A	N/A

4.0 Setup Procedure

To set up the Elastix Server for the Vega 5000

1. Go to the web address of the Elastix Server Login page. The web address is determined by the customer, for this guide we have used the IP address 192.168.1.75

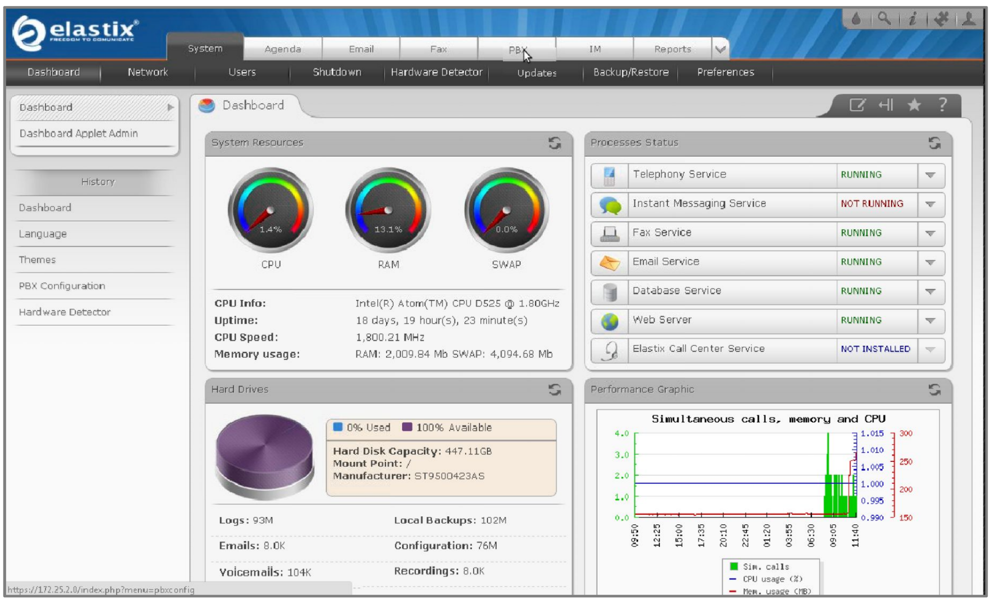
2. On the Login page, type the username and password for an administrative user into the Username and Password fields, see Figure 4-1. The username and password are determined by the customer.

Figure 4-1. Login

3. Press Enter or click on the Submit button to go to Elastix's Dashboard

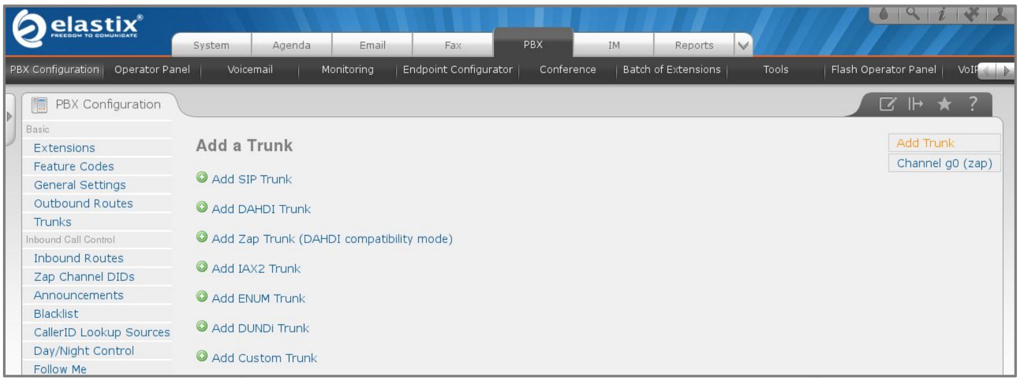
4. Once inside, click on the PBX tab on the menu at the top of the screen

Figure 4-2. Dashboard



5. We need to create a SIP Trunk to register the Vega with Elastix. Go to "PBX Configuration => Trunks => Add SIP Trunk", see Figure 4-3. This will take you to configure a SIP Trunk.

Figure 4-3. Add a SIP Trunk



6. On the "Add SIP Trunk" page (Figure 4-4), fill in the following information:

General Settings

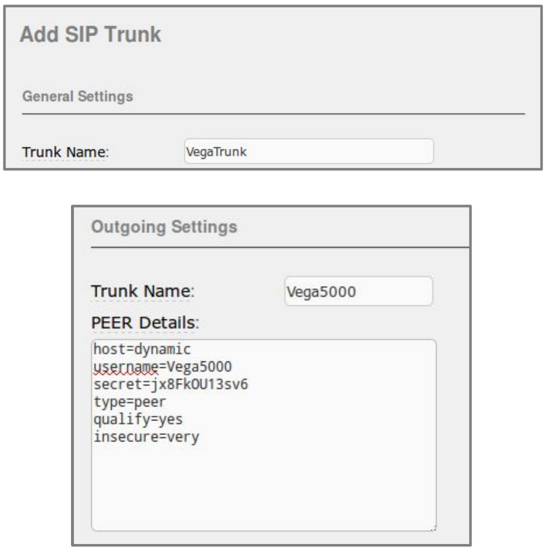
• **Trunk Name:** (VegaTrunk in this example)

Outgoing Settings

- **Trunk Name:** (Vega5000 in this example)
- **Peer Details:**

  host=dynamic
  username=(Vega5000 in this example)
  secret=(jx8FkOU13sv6 in this example)
  qualify=yes
  type=peer
  insecure=very

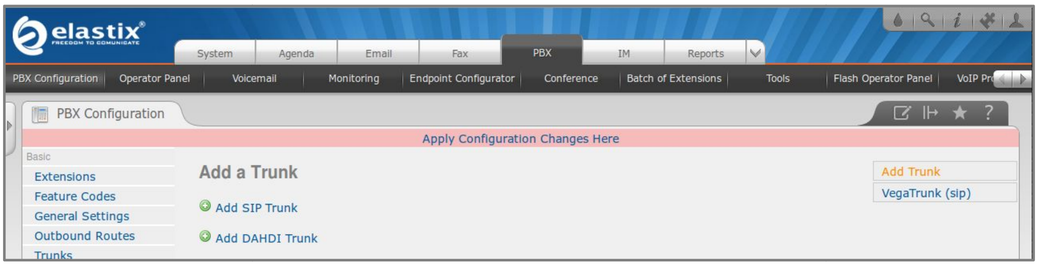
Figure 4-4. Add SIP Trunk



7. Click on the ‘Submit’ button at the end of the page. The SIP Trunk will be created and you will see the page on Figure 4-5 displaying the “Apply Configuration Changes Here” pink ribbon on top of the screen.

8. Click on the “Apply Configuration Changes Here” link

Figure 4-5. Apply Configuration Changes Here



9. With this you have finished creating a SIP Trunk that will be used by the Vega 5000 to register with the Elastix Server. Now, go to “PBX => PBX Configuration => Outbound Routes” to configure the outbound route to the Vega 50 Gateway. Fill in the following information: (Figure 4-6)

Route Settings

- **Route Name:** (“8_Vega” in this example)

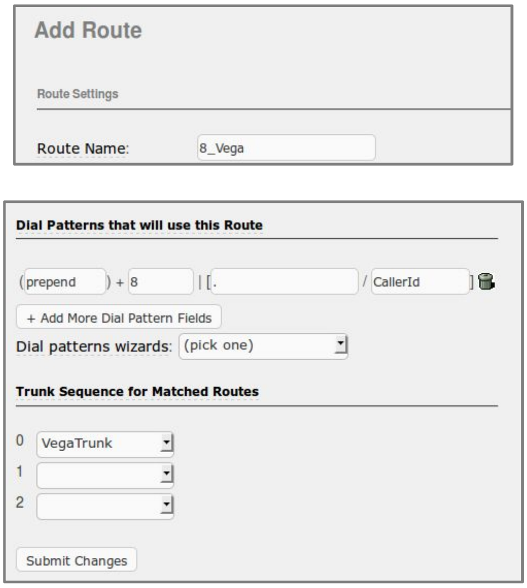
Dial patterns

- **Prefix:** (“8” in this example) | **Match pattern:** (“.” in this example)

Trunk Sequence for Matched Routes

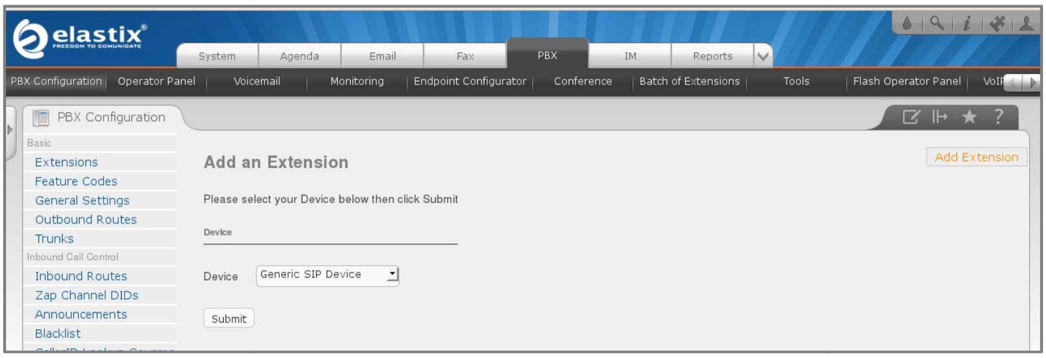
- **0:** (“VegaTrunk” in this example)

Figure 4-6. Add Route



10. Click on “Submit” at the end of the page and Apply changes. Now, we’ll create an extension for an IP Phone. Go to “PBX => PBX Configuration => Extensions” and click on “Submit” having the “Generic SIP Device” option selected. (Figure 4-7)

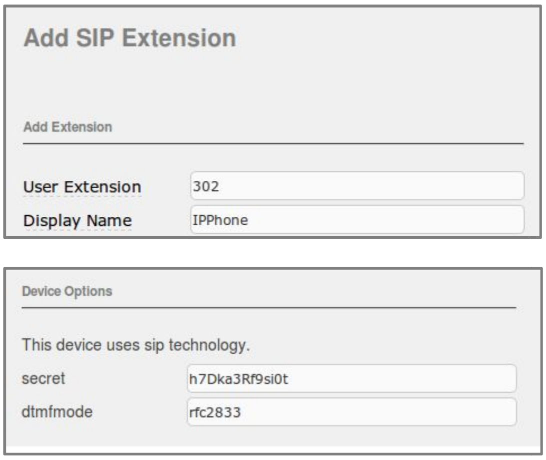
Figure 4-7. Add SIP Extension



11. Fill in the following information on the Add SIP Extension page (Figure 4-8):

- **User Extension** (302 in this example)
- **Display Name** (‘IPPhone’ in this example)
- **Secret** (‘h7Dka3Rf9si0t’ in this example)

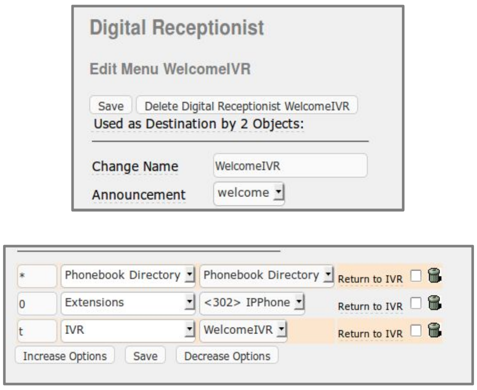
Figure 4-8. Add SIP Extension



12. Click on “Submit” at the end of the page and Apply changes. Now for incoming calls we will create an IVR. To do this, go to “PBX => PBX Configuration => IVR”. Click on “Add IVR” link (Figure 4.9). Set the following:

- **Name:** Name of IVR (WelcomeIVR in this example)
- **Announcement:** Voice prompt which will be played for incoming calls.
- **Options:**
    - o * - Phone book.
    - o 0 - 302 Extension
    - o t - Repeat the options of IVR (Add this option by modifying the IVR after creation)

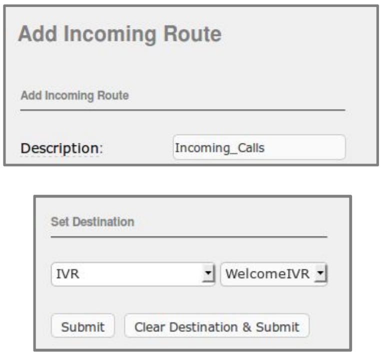
Figure 4-9. IVR



13. Click on “Save” and Apply changes by clicking on the pink ribbon that appears at the top of the page. Now go to “PBX => PBX Configuration => Inbound Routes”. Click on “Add Incoming Route” link (Figure 4.10). Set the following:

- **Description:** Name of inbound route (“Incoming_Calls” in this example)
- **Set destination:** Where the call will be routed. (“WelcomeIVR” IVR in this example)

Figure 4-10. Incoming Route



14. Click on “Submit” and apply changes. Now when we receive calls the “WelcomeIVR” IVR will play to the caller giving him choices to interact with Elastix Server.

15. To configure the gateway, you will need to enter the information from the trunk created on the Elastix Server into the Vega 5000 and set other parameters by logging into the WebUI.

    For the initial configuration, refer to the Vega 5000 Admin Guide found at: http://wiki.sangoma.com/Vega-5000-Technical-Documentation

    Note: Reset the Vega 5000 to factory default settings before continue.

Factory default settings

LAN1 IP Address	DHCP
LAN1 IP Address (If DHCP no available)	169.254.xxx.yyy
Web Access Administrator User	admin
Web Access Administrator Password	admin

Use Vega default IP address

If the Vega is powered up and no DHCP server is available, then the Vega will set its IP address to 169.254.xxx.yyy where xxx and yyy are defined by the MAC address of the Vega LAN interface. xxx and yyy are both one to three digit decimal values.

The MAC address of the Vega LAN interface will be 00:50:58:WW:XX:YY (found on the rear of the Vega – for details see the 'Use IP ping and Arp cache' section above) where WW, XX and YY are each 2 hexadecimal digits.

- The xxx value in the IP address is the decimal value of the XX hex value from the MAC address, and
- The yyy value in the IP address is the decimal value of the YY hex value from the MAC address.

Go to the Vega 5000's WebUI by pointing your browser to the Vega's IP address (Figure 4-11).

Figure 4-11. Vega 5000's WebUI

16. When the WebUI is loaded, go to "Quick Configuration" located on the left side of the page and click on "Continue" button (Figure 4-12).

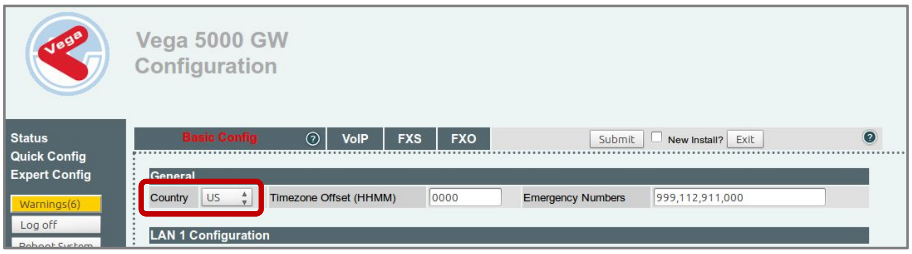
Figure 4-12. Quick Config

17. Once there, go to "Basic Config" tab and set the following (Figure 4-13):

General

- **Country**: US

Figure 4-13. Quick Config – Basic Config

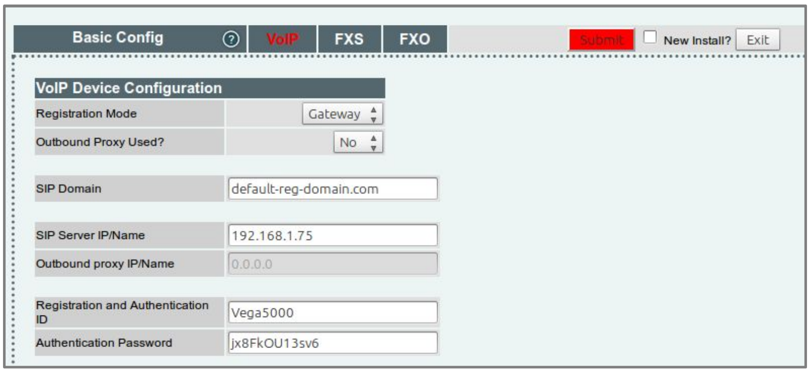


18. Go to “VoIP” tab and set the following information with the same data as step 6 (Figure 4-14):

VoIP Device Configuration

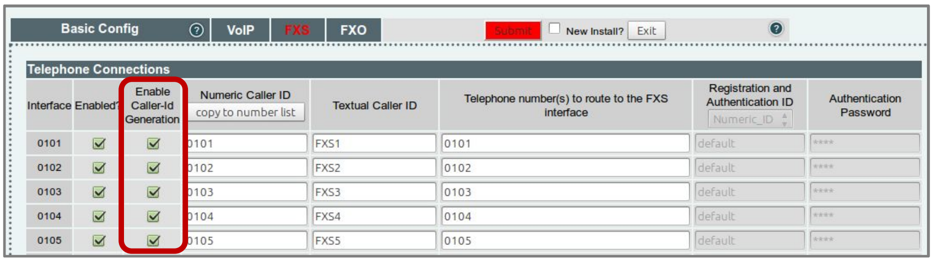
- **Registration Mode:** Gateway
- **SIP Server IP/Name:** Elastix Server’s IP Address (192.168.1.75)
- **Registration and Authentication ID:** (Vega5000 in this example)
- **Authentication Password:** (jx8FkOU13sv6 in this example)

Figure 4-14. Quick Config – VOIP



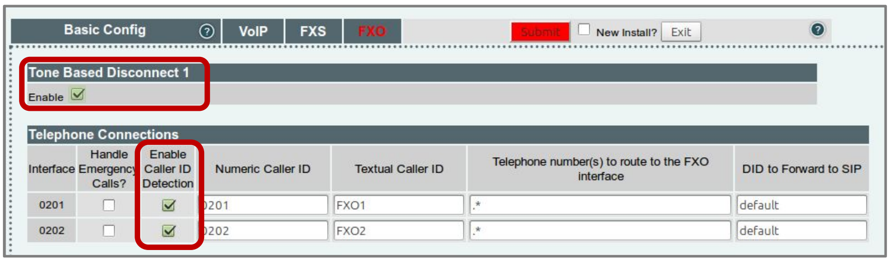
19. Go to “FXS” tab and check the “Enable Caller ID Generation” option. (Figure 4-15).

Figure 4-15. Quick Config – FXS



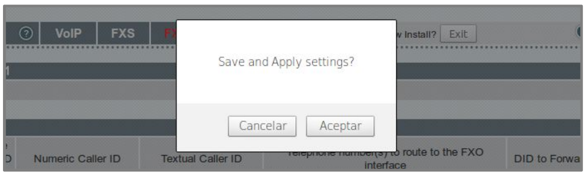
20. Go to “FXO” tab and check the “Tone Based Disconnect” and “Enable Caller ID Detection” option. (Figure 4-16).

Figure 4-16. Quick Config - FXO



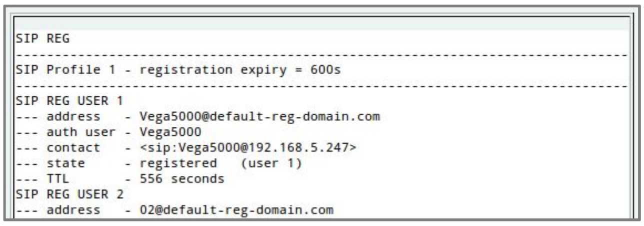
21. To apply the changes click on “Submit” button in red next to the tabs menu. (Figure 4-17):

Figure 4-17.Applying changes



22. Click on “Accept”. Now, go to “Status” located on the left side of the page to check whether the registration was successful (Figure 4-18).

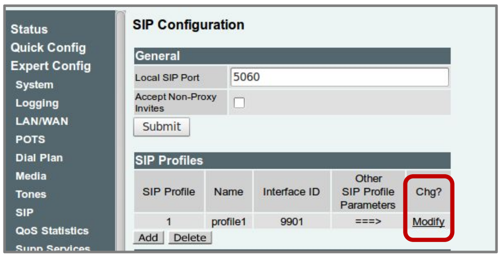
Figure 4-18. Status



```
SIP REG
----------------------------------------------------------------------
SIP Profile 1 - registration expiry = 600s
----------------------------------------------------------------------
SIP REG USER 1
--- address   - Vega5000@default-reg-domain.com
--- auth user - Vega5000
--- contact   - <sip:Vega5000@192.168.5.247>
--- state     - registered   (user 1)
--- TTL       - 556 seconds
SIP REG USER 2
--- address   - 02@default-reg-domain.com
```

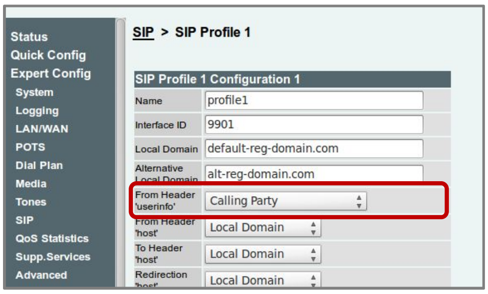
23. If the gateway is not registered check you have entered the correct information. Now go to “Expert Config” menu on the left side, and click on “SIP” option. (Figure 4-19).

Figure 4-19. Expert Config - SIP



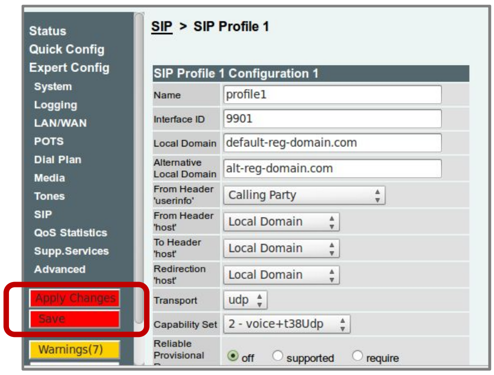
24. In the “SIP Profiles” section click on “Modify”, and set the ***From header user info*** parameter to ***Calling party.*** This setting will allow us to see the caller ID in a call. (Figure 4-20).

Figure 4-20. SIP Profile



25. Click on “Submit” button. To apply and save all the changes we have made, click on the red buttons located on the left. They are “Apply Changes” and “Save”, in that order (Figure 4-21).

Figure 4-21. SIP Profile

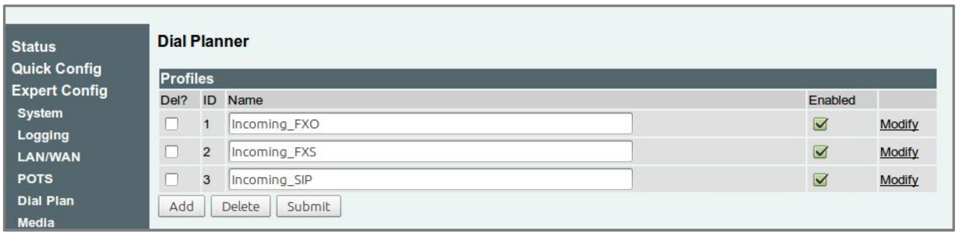


26. Once you have saved the changes, go to "Dial Plan" located on the "Expert Config" section (Figure 4.22)

Figure 4-22. Dial Plan

27. Here we're going to create a dial plan to route the calls. Delete all the entries and create 3 profiles that will be used for incoming calls to FXS, FXO and SIP interface. Below is the configuration of each profile (Figure 4-23).

Figure 4-23. Dial Plan



<u>For a better understanding:</u>

9901 : SIP Interface ID

0101 : FXS 1 port ID (0102 FXS 2, 0103 FXS 3, etc)

0201 : FXO 1 port ID (0202 FXO 2)

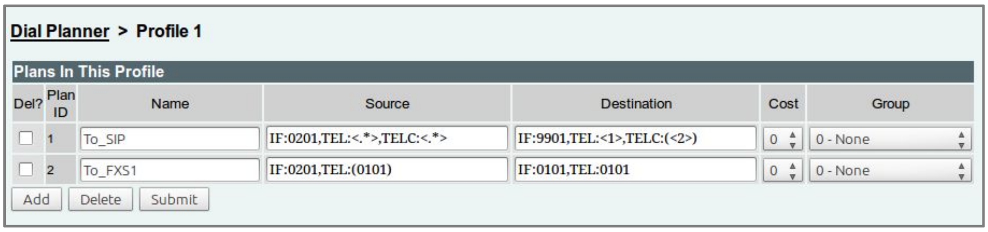
Incoming_FXO

ID	Name	Source	Destination
1	To_SIP	IF:0201,TEL:<.*>,TELC:<.*>	IF:9901,TEL:<1>,TELC:(<2>)
2	To_FXS1	IF:0201,TEL:(0101)	IF:0101,TEL:0101

The first rule will route the calls to the SIP Trunk. The dialed number can be any except 0101.

The second rule will route the call to the FXS1 port. The call takes this route provided the dialed number is 0101 (This is the extension number of the analog phone connected in the first FXS).

Figure 4-24. Incoming_FXO profile



Incoming_FXS

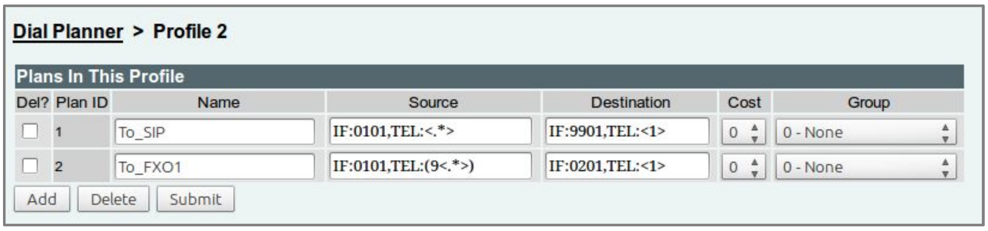
ID	Name	Source	Destination
1	To_SIP	IF:0101,TEL:<.*>	IF:9901,TEL:<1>
2	To_FXO1	IF:0101,TEL:(9<.*>)	IF:0201,TEL:<1>

The first rule will route the calls to the SIP Trunk. The dialed number can be any but it has not to begin with 9.

The second rule will route the call to the FXO1 port. The call takes this route provided the dialed number begins with 9.

.

Figure 4-25. Incoming_FXS



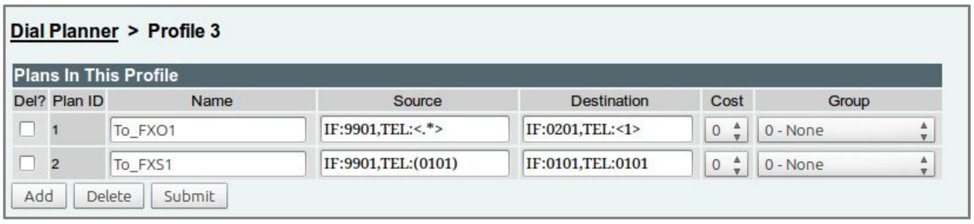
Incoming_SIP

ID	Name	Source	Destination
1	To_FXO1	IF:9901,TEL:<.*>	IF:0201,TEL:<1>
2	To_FXS1	IF:9901,TEL:(0101)	IF:0101,TEL:0101

The first rule will route the calls to the FXO1 port. The dialed number can be any except 0101.

The second rule will route the calls to FXS1 port provided the dialed number is 0101 (This is the extension number of the analog phone connected in the first FXS).

Figure 4-26. Incoming_SIP profile



28. For more help using expressions you can take a look on the **Regular Expression Help** and **Token Help** section located below of the plans of each dial plan profile. Don’t forget to Apply and Save changes clicking on the red buttons located on the left.

29. Now that you have configured the dial plan, you are able to make the following calls:

    From PSTN to Elastix server,
    From PSTN to Analog phones connected to Vega
    From Analog phones to Elastix Server
    From Analog phones to PSTN
    From Elastix Server to Analog phones connected to Vega
    From Elastix Server to PSTN