ELASTIX PBX APPLIANCE SOFTWARE + ASTERISK IP PBX

Aug. 2012

Configuring for Integra Telecom SIP Solutions

Document Index

Section **Page**

BACKGROUND

This document provides customer guidance for configuring the Elastix PBX Appliance Software with the underlying FreePBX graphical user interface (GUI) to properly interface to and interoperate with the Integra Telecom SIP Solutions trunks. The software applications are a graphical front-end to an underlying Asterisk open-source IP PBX; running on a Linux server.

The goal of this document is to ensure that—when properly configured—the subject customer supplied equipment will interface to and operate with the Integra Telecom equipment and network.

In order to establish a configuration and compatibility baseline, **Table 1** below shows the software versions used for testing and evaluation in the Integra Telecom Lab. Software versions above (newer than) the tested version are typically regression tested by the applicable vendor and generally considered to be acceptable as well.

Table 1

Software Versions Tested

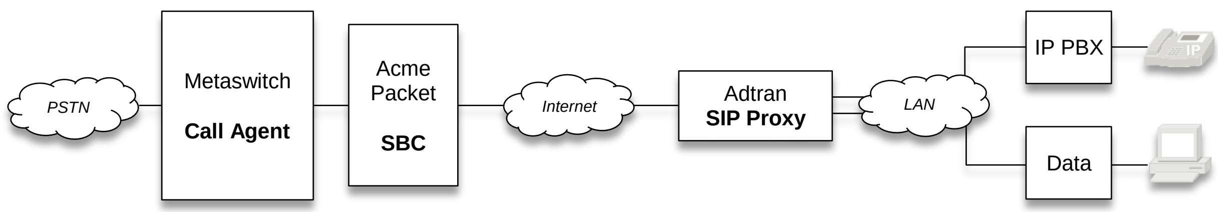
Network Element	Software
Metaswitch Call Agent	7.4.00 SU23 P90.00
Adtran NetVanta 3305	18.01.03.00
Elastix	**2.2.0-14**
elastix-firstboot	2.2.0-5
elastix-system	2.2.0-14
elastix-email_admin	2.2.0-9
elastix-vtigercrm	5.1.0-8
elastix-extras	2.0.4-4
elastix-asterisk-sounds	1.2.3-1
elastix-my_extension	2.2.0-5
elastix-agenda	2.2.0-5
elastix-a2billing	1.8.1-16
elastix-addons	2.2.0-4
elastix-im	2.0.4-2
elastix-pbx	2.2.0-14
elastix-security	2.2.0-7
elastix-reports	2.2.0-6
elastix-fax	2.2.0-4
FreePBX	**2.8.1-7**
Asterisk	**1.8.7.0-0**
asterisk-perl	0.10-2
asterisk-addons	1.8.7.0-0

CONFIGURATION

The generic IP PBX test configuration is shown in Figure 1 below. This configuration ensures proper SIP call handling between the customer-supplied IP PBX platform (Elastix software in this case) and the Integra Telecom equipment and network.

Figure 1

Generic IP PBX Test Configuration



Note that some 'schools of thought' like to consider everything outside of or beyond the IP PBX itself to be part of the PSTN. For this configuration document, the beginning of the PSTN does not matter; as 'in to' and 'out of' the IP PBX are the more important considerations.

PROVISIONING STEPS

The following steps provide provisioning guidance for configuring the Elastix PBX appliance software graphical user interface (GUI) parameters. This overlay GUI in turn pushes the configuration details to the underlying Asterisk open source IP PBX.

This process uses the web browser of choice. Specifics of the initial connectivity are outlined in the Elastix documentation supplied with the system.

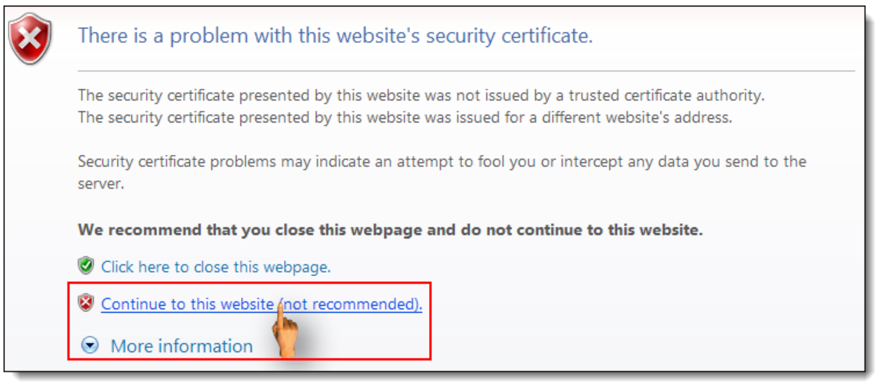
It is assumed that the appropriate hardware, software, and licenses have been procured, installed, and tested. As with all 'open source' software, the **onus** of technical support lies with the end user.

o·nus /ˈōnəs/

Noun: Used to refer to something that is one's duty or responsibility: "the onus is on you to show that you have suffered".

Synonyms: burden - charge - responsibility - weight - liability

- **Step 1: Security Warning —** When the web browser of choice is first connected to the Elastix web server, a certificate authority warning is typically displayed. Therefore, it is necessary to manually proceed to the login page. In this example, the screen shot below is from Microsoft Internet Explorer.



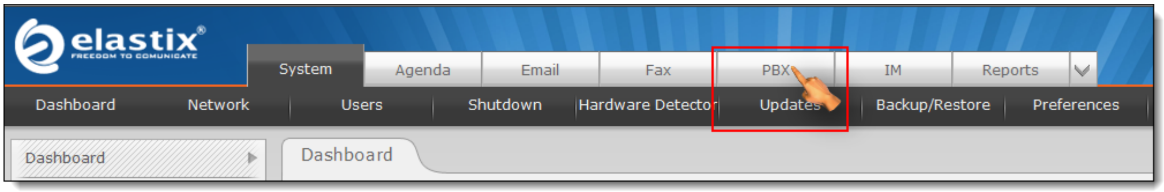
Select "Continue to this website (not recommended)." or equivalent continue message option (on the selected web browser) to proceed to the Elastix login page.

- **Step 2: Elastix Login —** Enter the appropriate administrative login credentials to access Elastix. Enter the username and password previously selected.

Press Submit.

After successfully accessing the Elastix system, the landing page shown in the next step will be displayed.

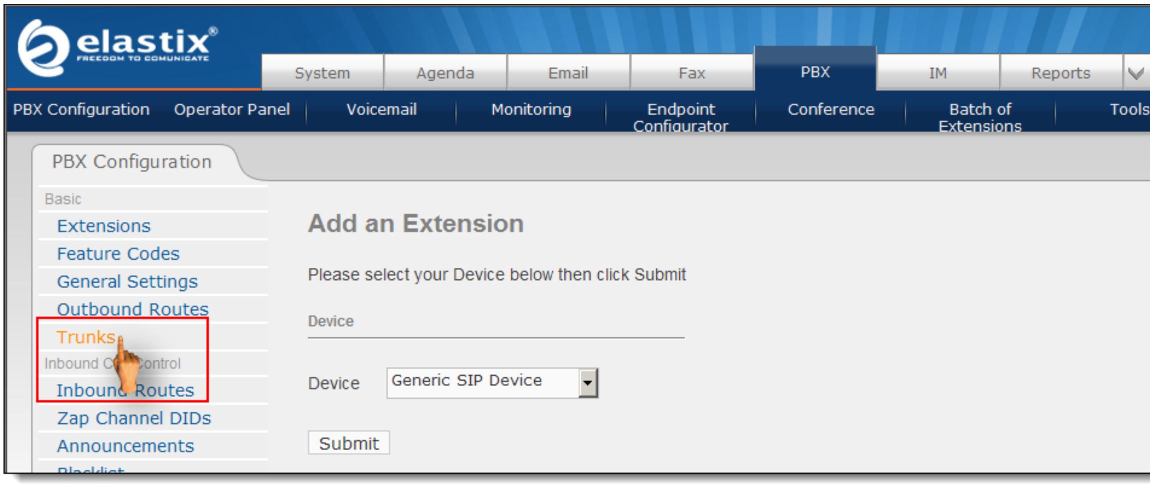
- **Step 3: Navigate to PBX —** Using the top menu tabs, navigate to PBX.



Select PBX.

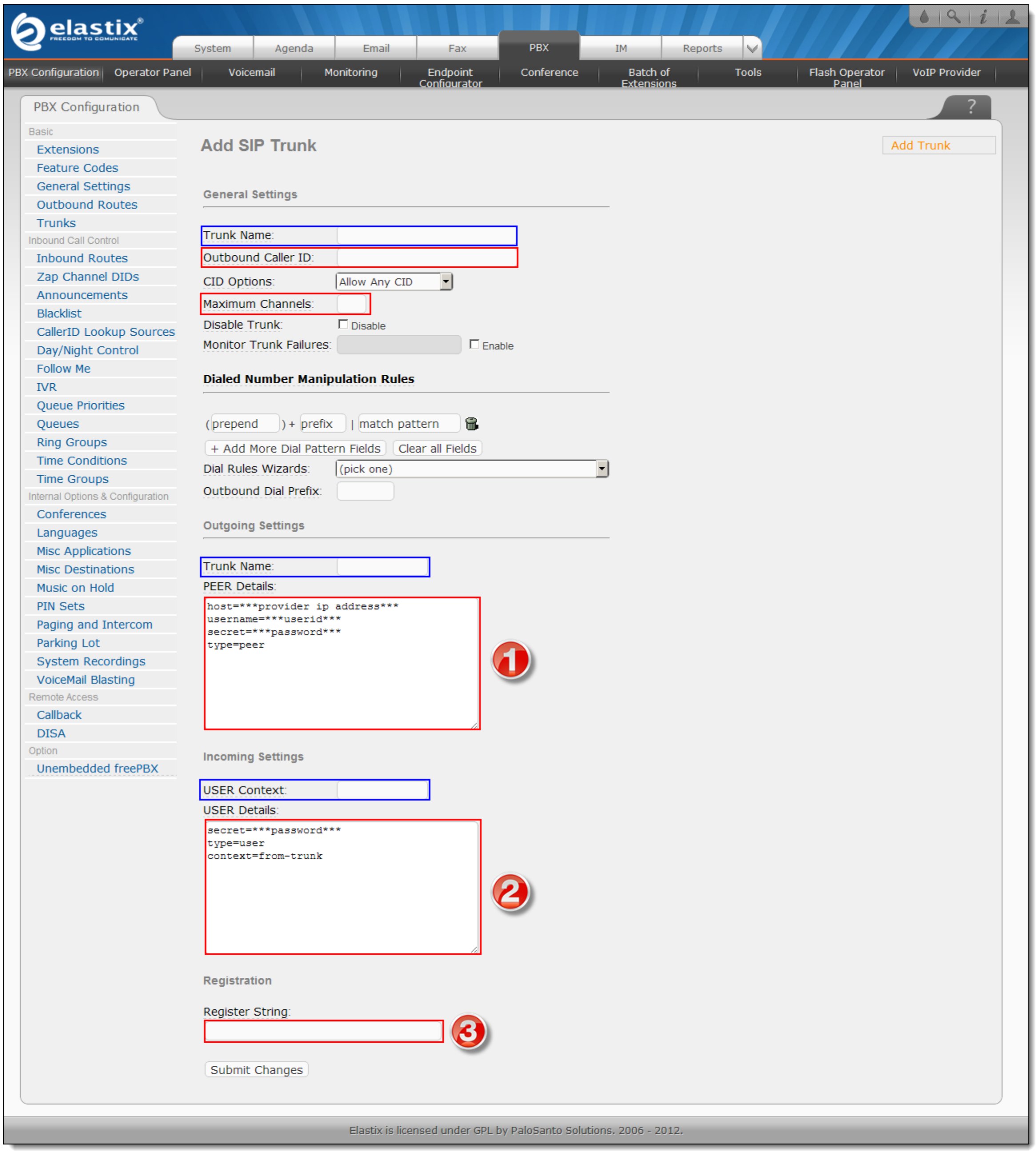
After selecting **PBX**, the **PBX Configuration** tab will be displayed as shown in the next step.

- **Step 4: Add a SIP Trunk (a) —** Using the left side menu tree, select **Trunks**.



- **Step 5: Add a SIP Trunk (b) —** On the **Add a Trunk** page, select **Add SIP Trunk**.

- **Step 6: Add a SIP Trunk (c) —** The **Add SIP Trunk** configuration page is shown below.



The **Trunk Name** and **User Context** fields (**outlined in blue**) may be filled in with a SIP Solutions trunk name and context for easy identification.

The parameters shown above that are **outlined in red**, **MUST** be configured to match the information as discussed below.

Outbound Caller ID: *Caller ID number as provided by Integra Telecom.*

Maximum Channels: *The number of SIP sessions ordered by the customer.*

(1) PEER Details:
*host=**Integra Telecom provided SIP domain name.***
*username=**Integra Telecom provided SIP user.***
*secret=**Integra Telecom provided SIP password.***
type=peer
disallow=all
allow=g729&ulaw (Assumes customer has purchased the appropriate number of G.729 Codec licenses from Digium.)

(2) USER Details:
disallow=all
allow=g729&ulaw (Assumes customer has purchased the appropriate number of G.729 Codec licenses from Digium.)
context=from-trunk
insecure=very
dtmfmode=auto
*fromdomain=**Integra Telecom provided private IP address of the Adtran SIP proxy CPE device.***
*fromuser=**Integra Telecom provided SIP user.***
*host=**Integra Telecom provided private IP address of the Adtran SIP proxy CPE device.***
nat=no
qualify=yes
type=peer
*username=**Integra Telecom provided SIP user.***
*secret=**Integra Telecom provided SIP password.***
port=5060

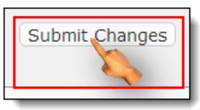
(3) Register String: *Username:Secret@Host/Username*

Username ***Colon*** *Secret* ***Ampersand*** *Host* ***Slash*** *Username*

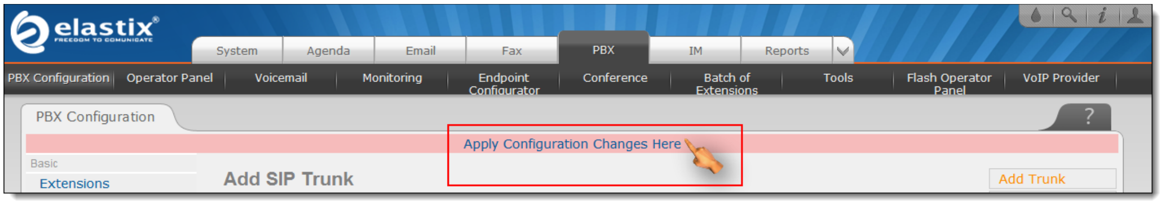
The example string below shows the properly formatted Register String using the Integra Telecom provided data fill.

3608523950:Int3graPassw0rd@proxy1.integravoip.net/3608523950

After the parameters are properly configured and double checked for accuracy, **Select Submit Changes** at the bottom of the page.

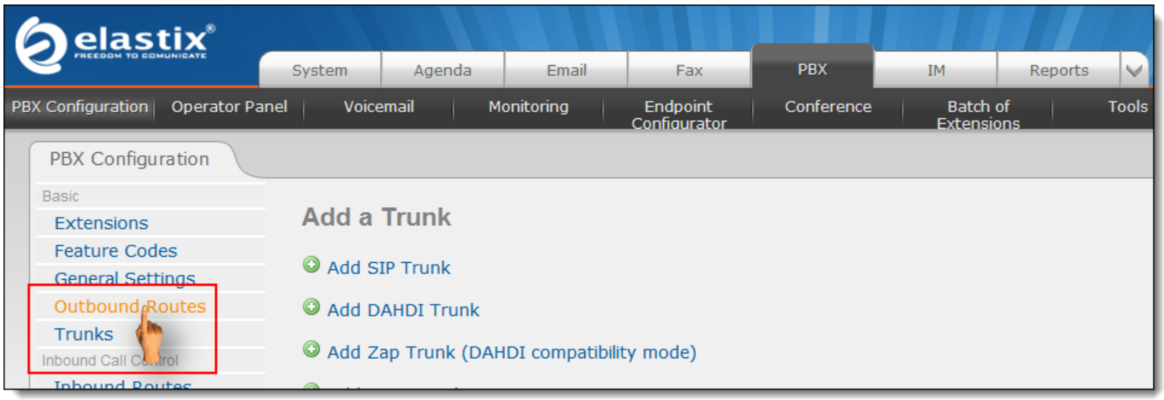


- **Step 7: Add a SIP Trunk (d) —** At the top of the **PBX Configuration** tab, select **Apply Configuration Changes Here** to reload the Asterisk PBX with the updated configuration.

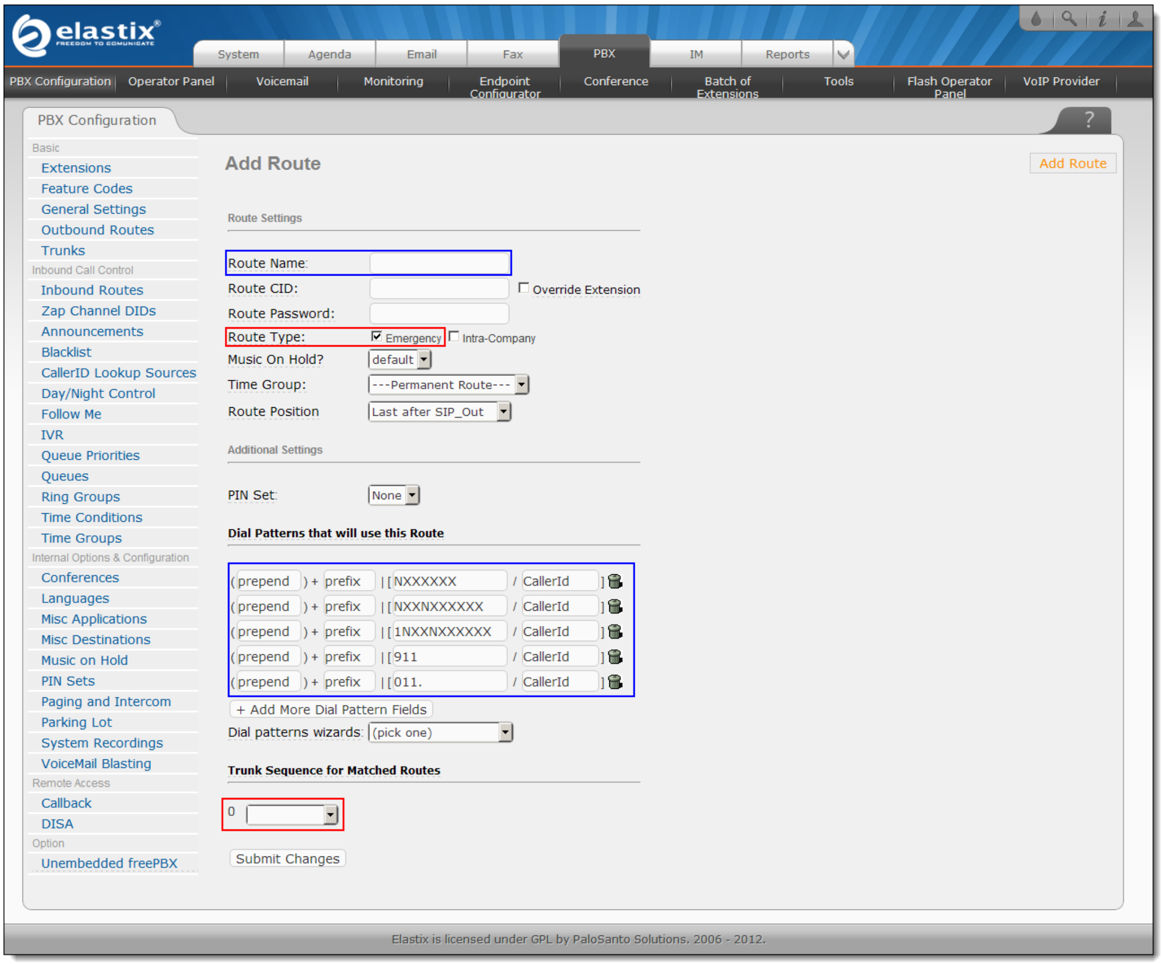


Asterisk PBX will reload. This <u>should</u> not be a service affecting operation.

- **Step 8: Add Outbound Routes (a) —** After the SIP trunk is added, and applied, using the left side menu tree select **Outbound Routes**.



- **Step 9: Add Outbound Routes (b) —** The Add Route configuration page is shown below.



The **Route Name** field (**outlined in blue**) may be filled in with a friendly route name for easy identification.

The **Dial Patterns that will use this route** fields (**outlined in blue**) are the customer defined patterns that will result in outbound calls using the SIP trunk provisioned in Steps 4-7.

In the above example, the entries are as follows.

Line 1 NXXXXXX is 7-digit local calling.
Line 2 NXXNXXXXXX is 10-digit local calling.
Line 3 1NXXNXXXXXX is 11-digit long distance (1+10 digits) dialing.
Line 4: 911 is emergency services dialing.
Line 5: 011. Is international long distance (011+X digits+#) dialing.

Note that E.164 dialing is not provisioned and is not supported in the above configuration statements.

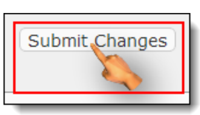
Again, the customer decides which dialing patterns are supposed to egress the Integra Telecom SIP Solutions trunk(s).

The parameters shown above that are **outlined in red**, **MUST** be configured to match the information as discussed below.

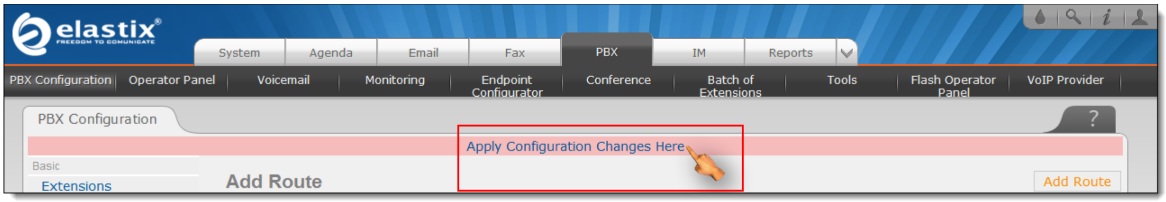
Route Type: *If this has been designated by the customer as a 911 emergency services route, the **Emergency** check box must be selected.*

Trunk Sequence for Matched Routes: *The Integra Telecom SIP Solutions **Trunk Name**, provisioned in Step 6, must be selected from the drop down menu list in this field. Additional (non-Integra Telecom) trunk routes may also be selected as alternate choices, if desired. Likewise, the Integra Telecom trunk routes may be selected as alternate choices for other primary trunks.*

After the parameters are properly configured and double checked for accuracy, **Select Submit Changes** at the bottom of the page.

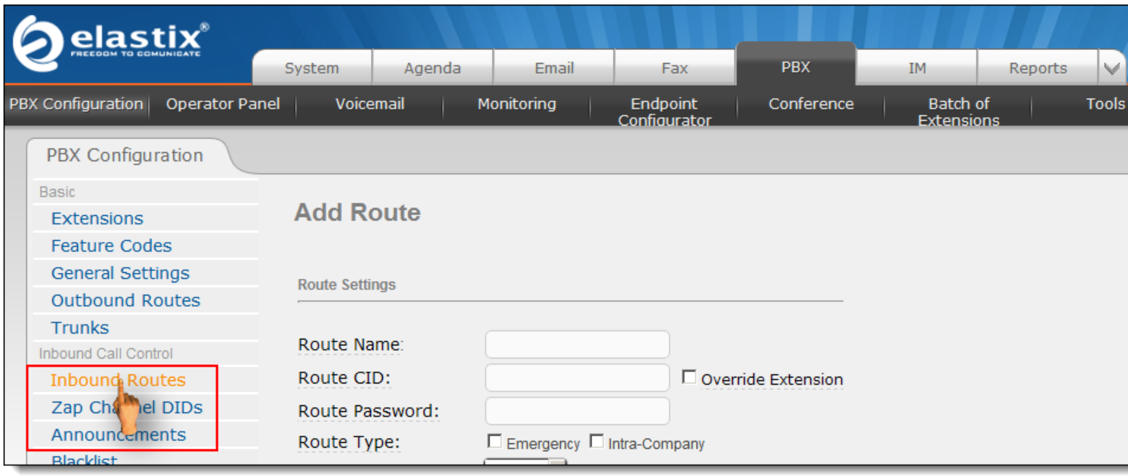


- **Step 10: Add Outbound Routes (c) —** At the top of the **PBX Configuration** tab, select **Apply Configuration Changes Here** to reload the Asterisk PBX with the updated configuration.

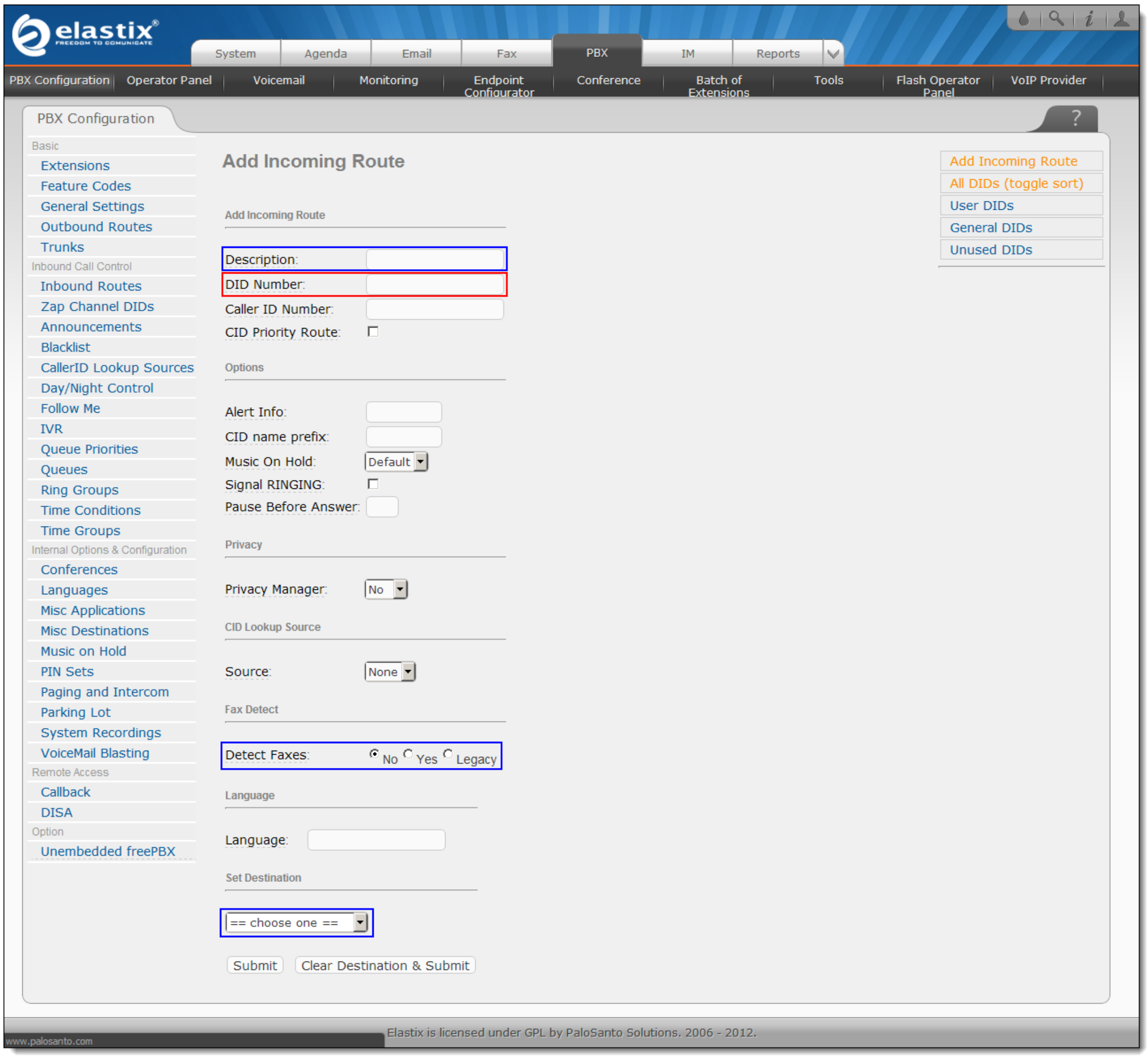


Asterisk PBX will reload. This should not be a service affecting operation.

- **Step 11: Add Inbound Routes (a) —** After the outbound route is added, and applied, using the left side menu tree select **Inbound Routes**.



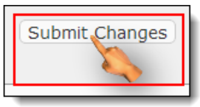
- **Step 12: Add Inbound Routes (b) —** The Add Incoming Route configuration page is shown below.



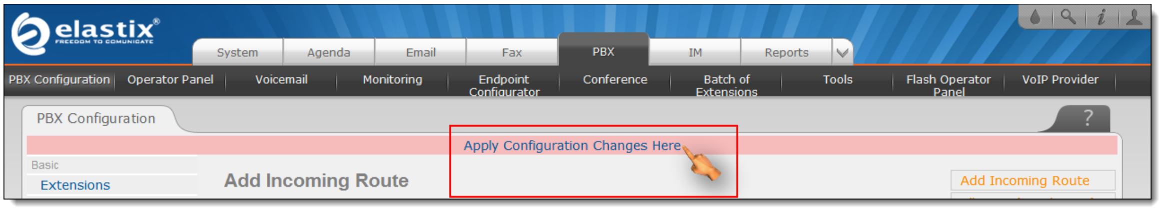
The majority of the provisioning on this page is customer determined and thus customer defined. The incoming routes specify the extension (or other selected option) that will terminate the incoming DID number(s).

As a result, the **DID Number** field (**outlined in red**), **MUST** be configured to match the information provided by Integra Telecom. The DID values and/or ranges will mirror the quantity ordered by the customer. One route for each individual DID number.

After the parameters are properly configured and double checked for accuracy, **Select Submit Changes** at the bottom of the page.



- **Step 13: Add Inbound Routes (c) —** At the top of the **PBX Configuration** tab, select **Apply Configuration Changes Here** to reload the Asterisk PBX with the updated configuration.



Asterisk PBX will reload. This <u>should</u> not be a service affecting operation.

End of Provisioning Procedure.

At this point, the Elastix and underlying should be properly configured to interface to the Integra Telecom SIP Solutions trunk. Calls in to and out of the Elastix host IP PBX should be possible at this time. If not, proceed to the next section: **Troubleshooting**.

TROUBLESHOOTING

Comprehensive troubleshooting of a red status (not registered) SIP Solutions trunk is not part of this configuration document. However, experience has shown that there is a reasonable list of basic **Items to Check** as part of the nominal customer turn up process.

Items to Check

- ✓ **1. Confirm that the Linux (or UNIX equivalent) operating system machine which hosts the Elastix software has the proper TCP/IP interface configuration (IP address, Subnet mask, Default gateway, Primary and Secondary DNS server).**
- ✓ **2. Confirm, as a minimum, that the Telephone Service, Fax Service, Database Service, and Web Server of the Elastix services are showing "Running" status on the Linux operating system machine.**
- ✓ **3. Confirm the Elastix host machine can ping the LAN IP (private) address of the Adtran SIP proxy CPE at the customer site.**
- ✓ **4. Confirm the Elastix host can ping the WAN IP (public) address of the Adtran SIP proxy CPE at the customer site.**
- ✓ **5. Confirm the Elastix host can resolve, ping, and trace route to the FQDN (provided by Integra Telecom) of the Metaswitch Call Agent serving the customer site.**
- ✓ **6. Confirm with Integra Telecom that the SIP account information provided for the Elastix configuration (FQDN, Username, Secret, etc.) is correct.**
- ✓ **7. Confirm that the information provided by Integra Telecom is properly entered in the Elastix host.**

- ✓ 8. Confirm with Integra Telecom that they can 'see' the SIP "Register" messages from the Elastix host system within the BrixWorks call monitoring system.

- ✓ 9. Confirm with Integra Telecom that they can ping and trace route to the WAN IP (public) address of the Adtran SIP proxy CPE at the customer site.

- ✓ 10. Confirm with Integra Telecom that they can resolve, ping, and trace route from within the Adtran SIP proxy CPE to the FQDN of the Metaswitch Call Agent serving the customer site.

- ✓ 11. Confirm with Integra Telecom that they can ping the LAN IP (private) address from within the Adtran SIP proxy CPE at the customer site.

- ✓ 12, Confirm with Integra Telecom that they can ping the Elastix host IP address (private) from within the Adtran SIP proxy CPE at the customer site.

Summary

Traditionally, one or more of the above steps of confirmation and basic troubleshooting will yield the location of any problem(s) with the exchange of SIP messaging. Confirming the SIP "Register" message within the Integra Telecom's BrixWorks monitoring system generally provides a wealth of information on the nature of the issue; including any possible improper configuration. Therefore, it is appropriate to initiate that step only after the initial customer site configuration confirmation and connectivity testing has been completed (Steps 1-7 inclusive). This section represents a logical troubleshooting methodology from the customer location to the Metaswitch call agent.