Configuring Switchvox (Free Edition 1.0) for Spitfire SIP Trunks

This document is a guideline for configuring Spitfire SIP trunks onto Switchvox Free Edition and includes the settings required for Inbound DDI routing and Outbound CLI presentation. The settings contained within have been tested and are known to work at the time of testing.

SIP trunk details such as account number and password will be provided separately.

Provisioning a SIP trunk: *Page 2*

Inbound Routing – Reference only: *Page 7*

Outgoing CLI – Reference only: *Page 9*

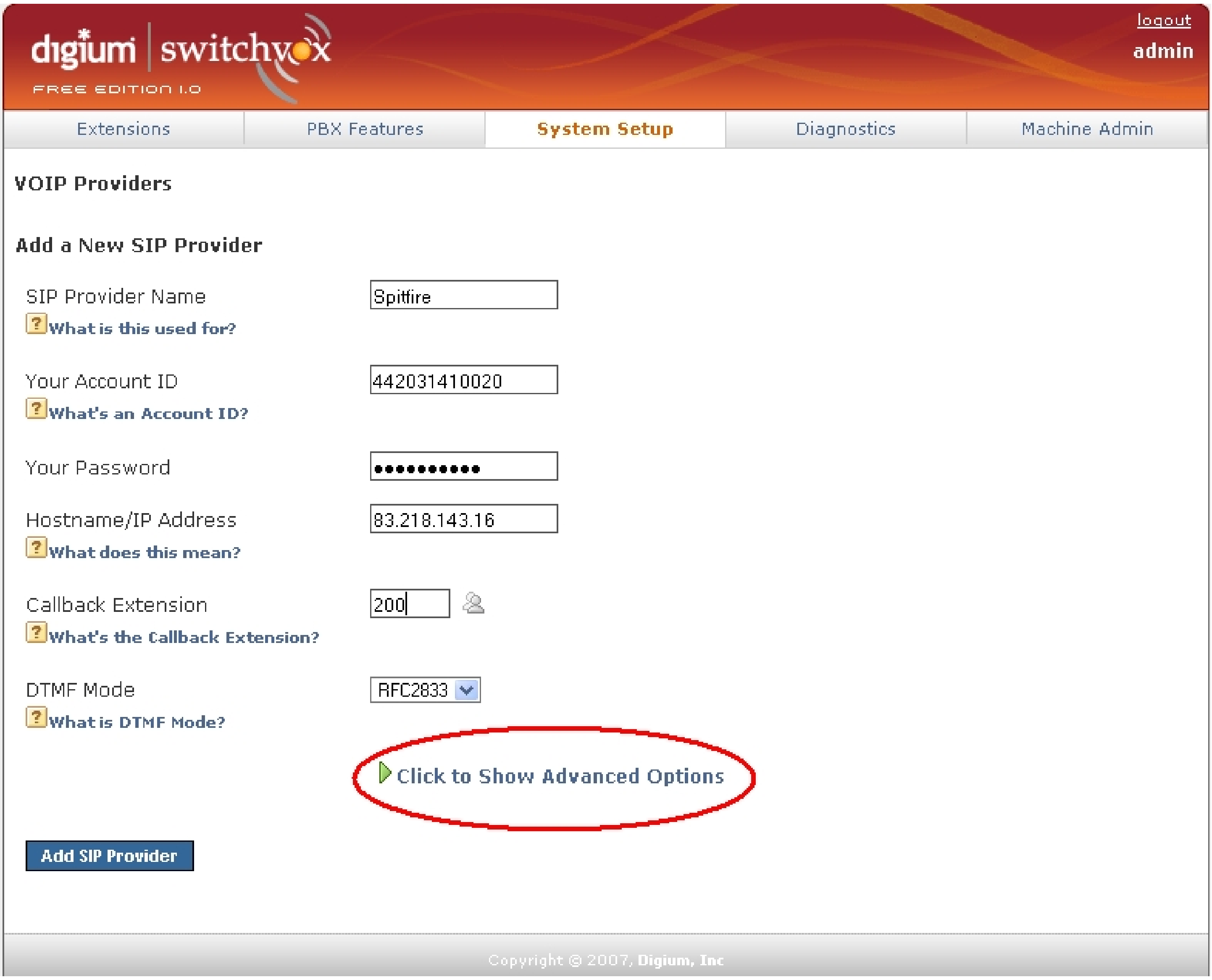
Provisioning a SIP Trunk

Select **"SIP Provider"** & click on **"Go"** next to Add New

Add New SIP Provider

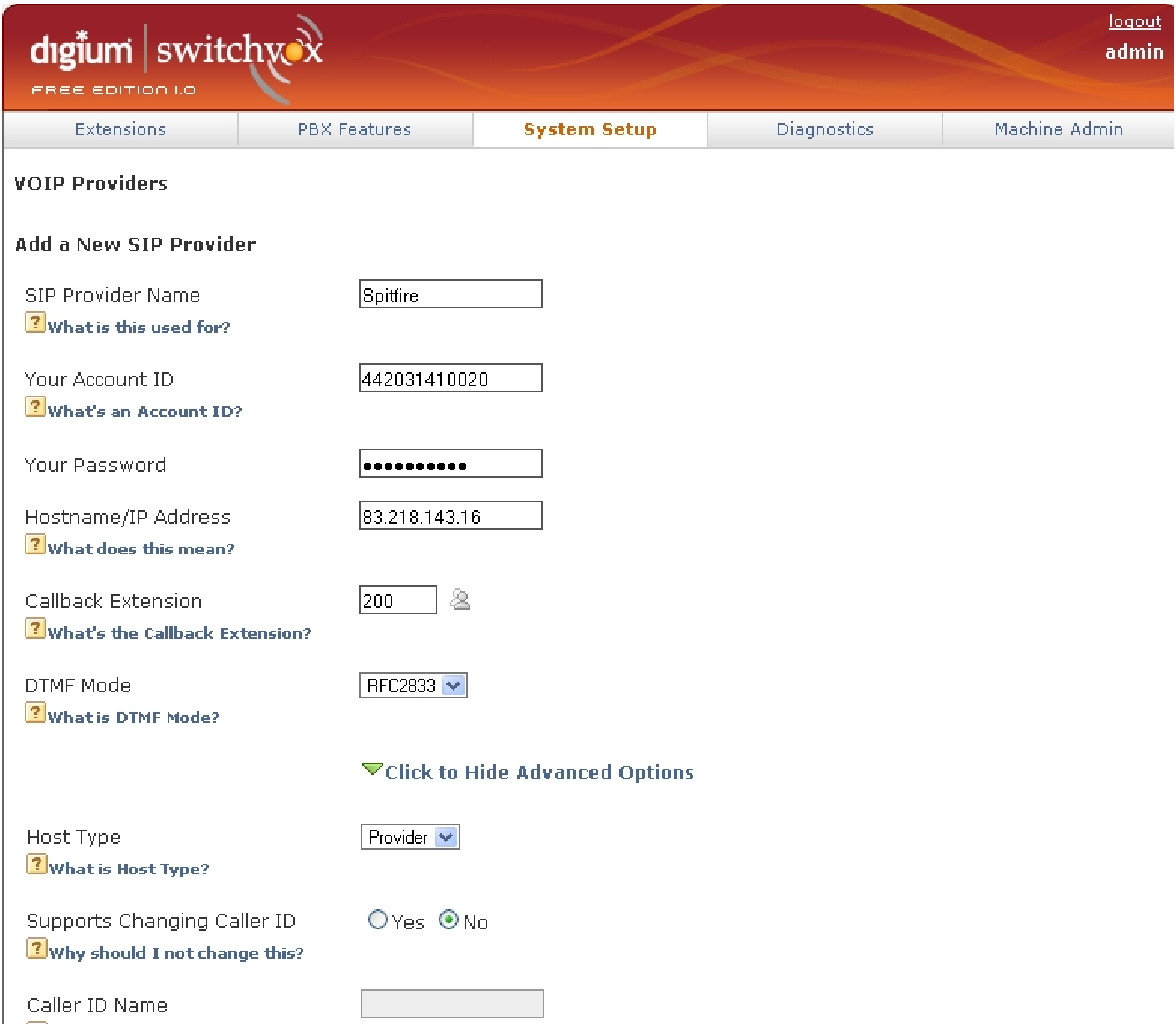
1. **SIP Provider Name:** Enter a logical name for the SIP Trunk
2. **Your Account ID:** Enter the **User name as supplied by Spitfire.** Please note that this is just the number and **DOES NOT** include @spitfiretsp.net
3. **Your Password:** Enter the **password as supplied by Spitfire**
4. **Hostname/IP Address:** Enter **83.218.143.16 (or spitfiretsp.net)**
5. **Callback Extension:** This is determined by the installer
6. **DTMF Mode:** Leave at RFC2833

Then click on **"Click to Show Advanced Options"**



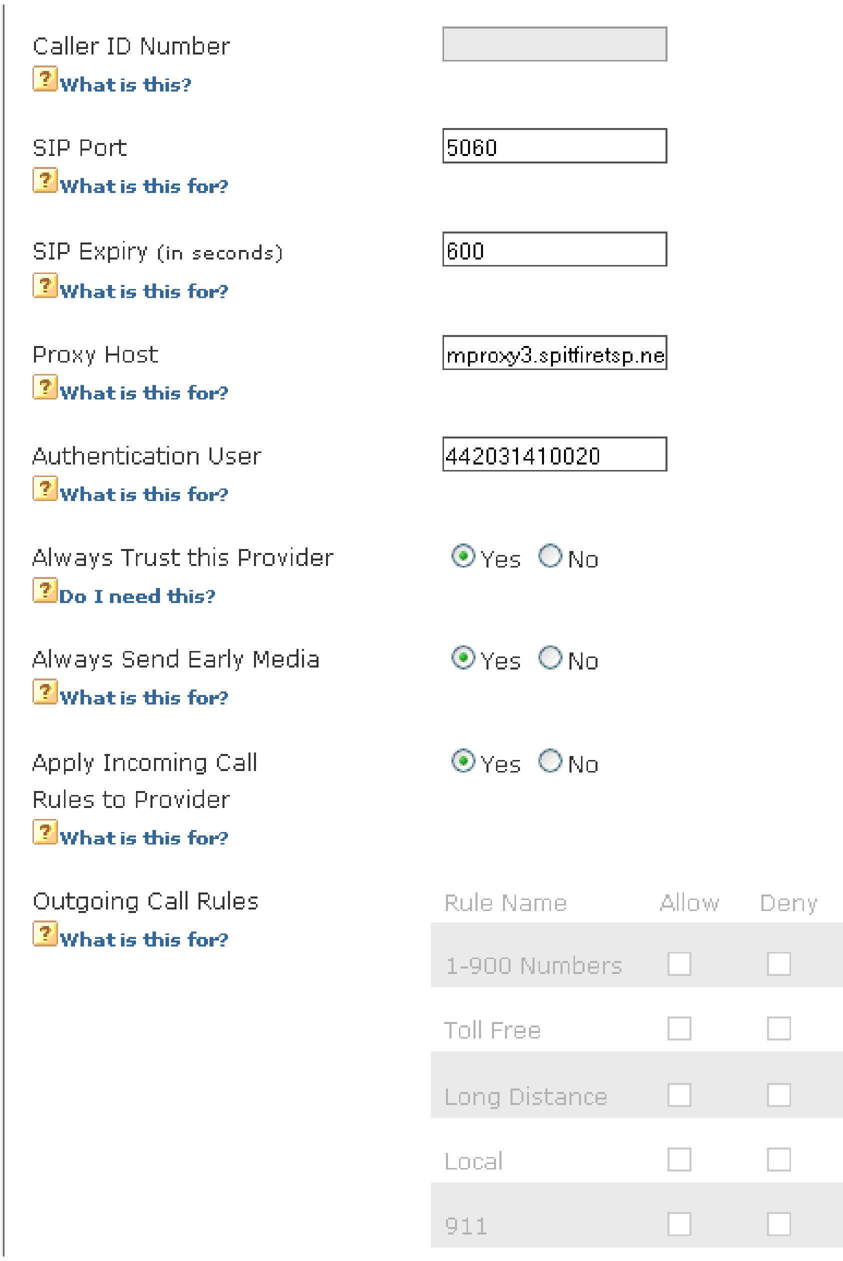
Add New SIP Provider continued

7. **Host Type:** Leave the Host Type as Provider
8. **Supports Changing Caller ID:** Leave at No
9. **Caller ID Name:** N/A



Add New SIP Provider continued

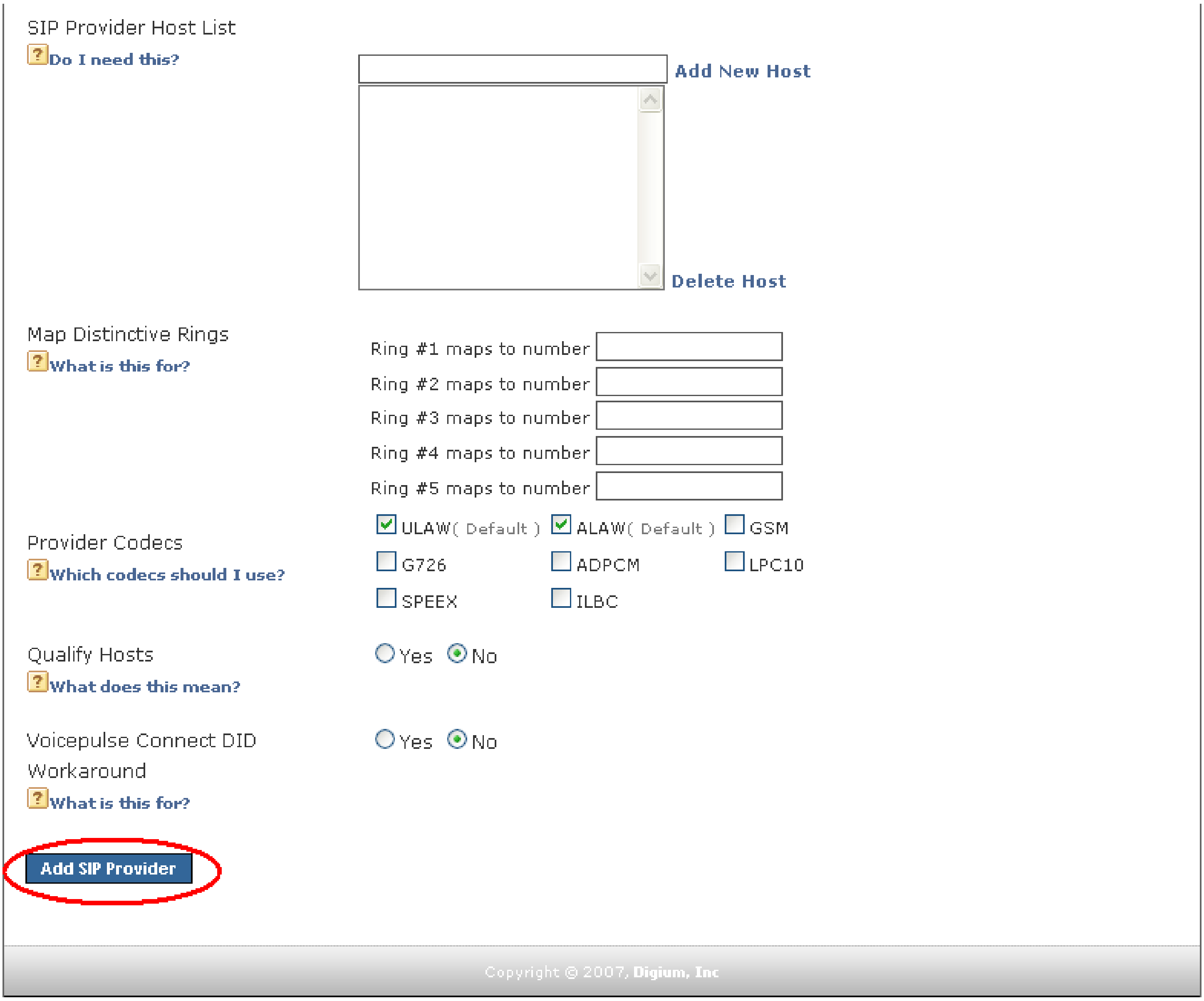
10.	**Caller ID Number:**	N/A
11.	**SIP Port:**	5060
12.	**Expiry Timer:**	Can be a maximum of 3600 – **Change to a minimum of 600**
13.	**Proxy Host:**	Change the Proxy Host to **mproxy3.spitfiretsp.net**
14.	**Authentication User:**	This is copied from the Account ID. Leave as is
15.	**Always Trust:**	Leave at Yes
16.	**Send Early Media:**	Change to **Yes**
17.	**Incoming Call Rules:**	Leave at Yes
18.	**Outgoing Call Rules:**	Determined by the installer



Add New SIP Provider continued

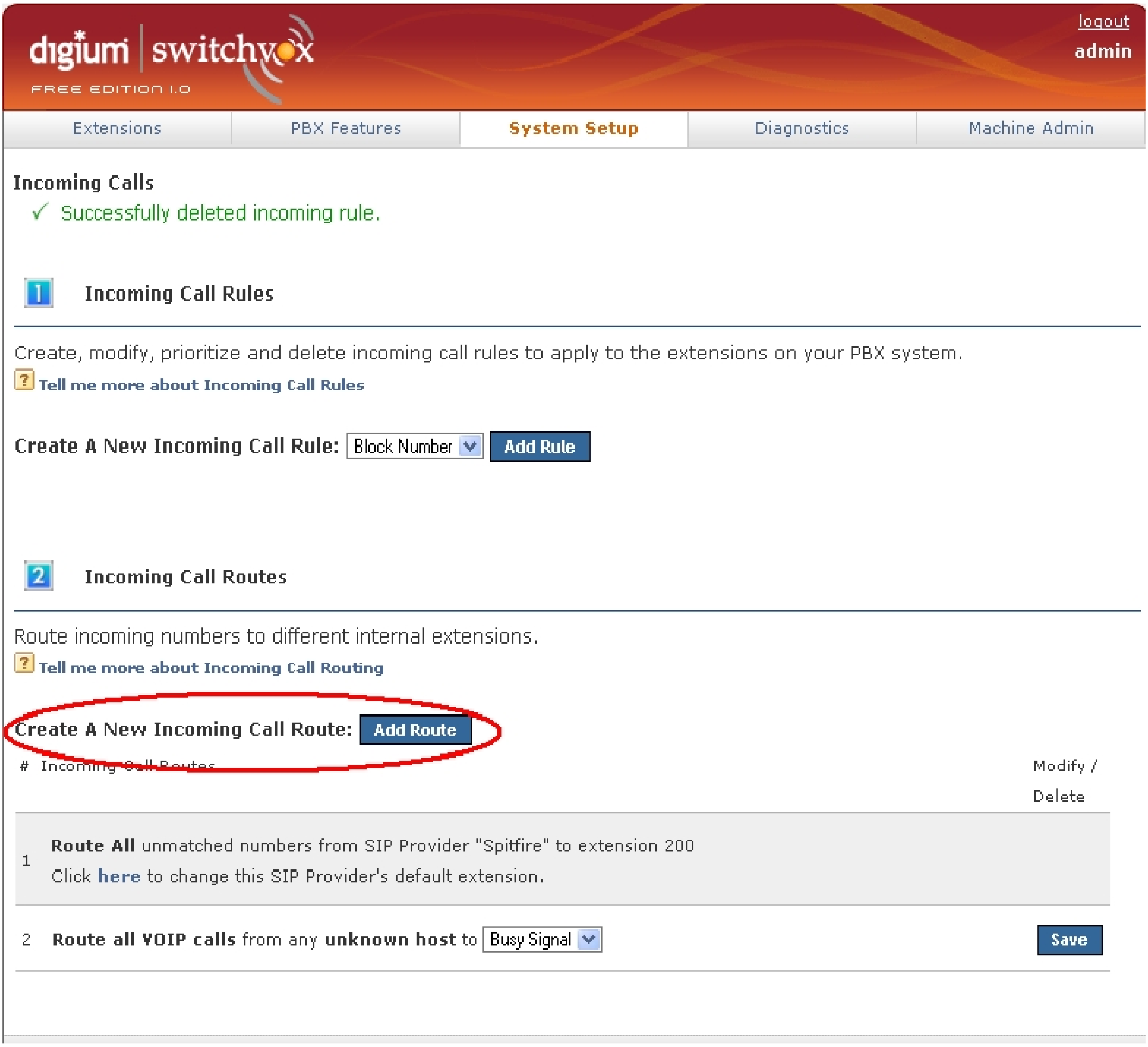
19. **SIP Provider Host List:** Leave empty
20. **Map Distinctive Rings:** Leave empty
21. **Provider Codecs:** Leave as default (uLaw & aLaw)
22. **Qualify Hosts:** No
23. **Voicepulse Workaround** No
24. THEN click on **"Add SIP Provider"**

If registration is not successful, check your settings & reload the software



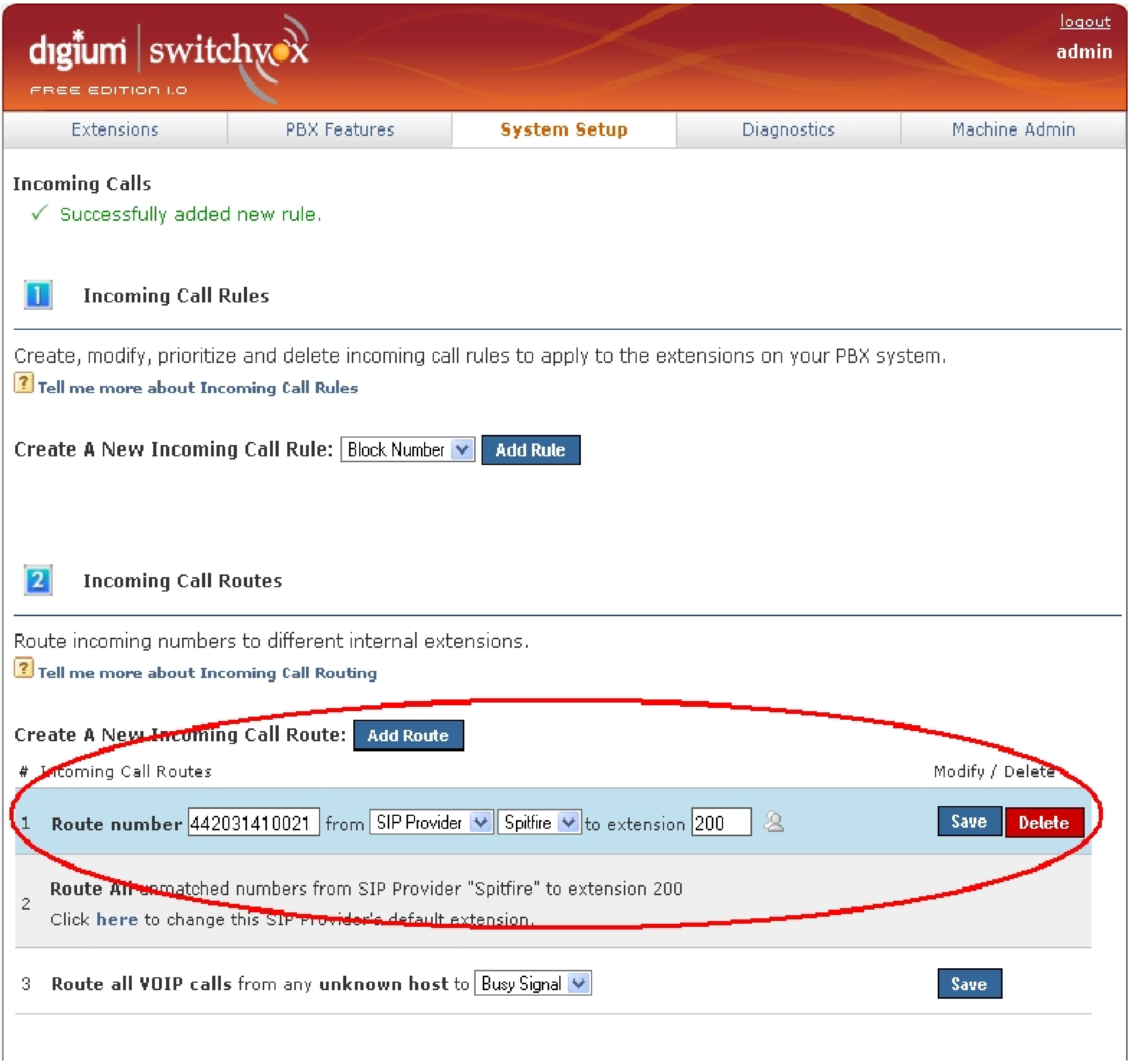
Inbound Routing – Reference only

Select "**New Incoming Call Route**"



Inbound Routing continued

1. **Route Number:** Enter the Route Number in the format **44xxxxxxxxxx** (as below)
2. **From:** Change to **SIP Provider**, and then **select** the name you have given to the trunk (Spitfire in this case)
3. **To Extension:** Enter the extension number you wish to route the call to and click **Save**.



Outgoing CLI – Reference only

<u>Note:</u> Individual CLI presentation is not available on the Switchvox free edition 1.0.

Supports changing Caller ID on the VoIP trunk is left at “No” because you cannot select RPID as a Caller ID Method on the Free Edition. Selecting “No” ignores any Caller ID Rule assigned on a per extension basis and simply presents the main number.

If you select “Yes”, and then apply individual Caller ID Rules on a per extension basis the SIP messaging will be incorrect as Spitfire expects to see it and the call will fail.