Configuration guide for Switchvox and Speakeasy.

This document will guide a Switchvox administrator through configuring the system to utilize Speakeasy's SIP service.

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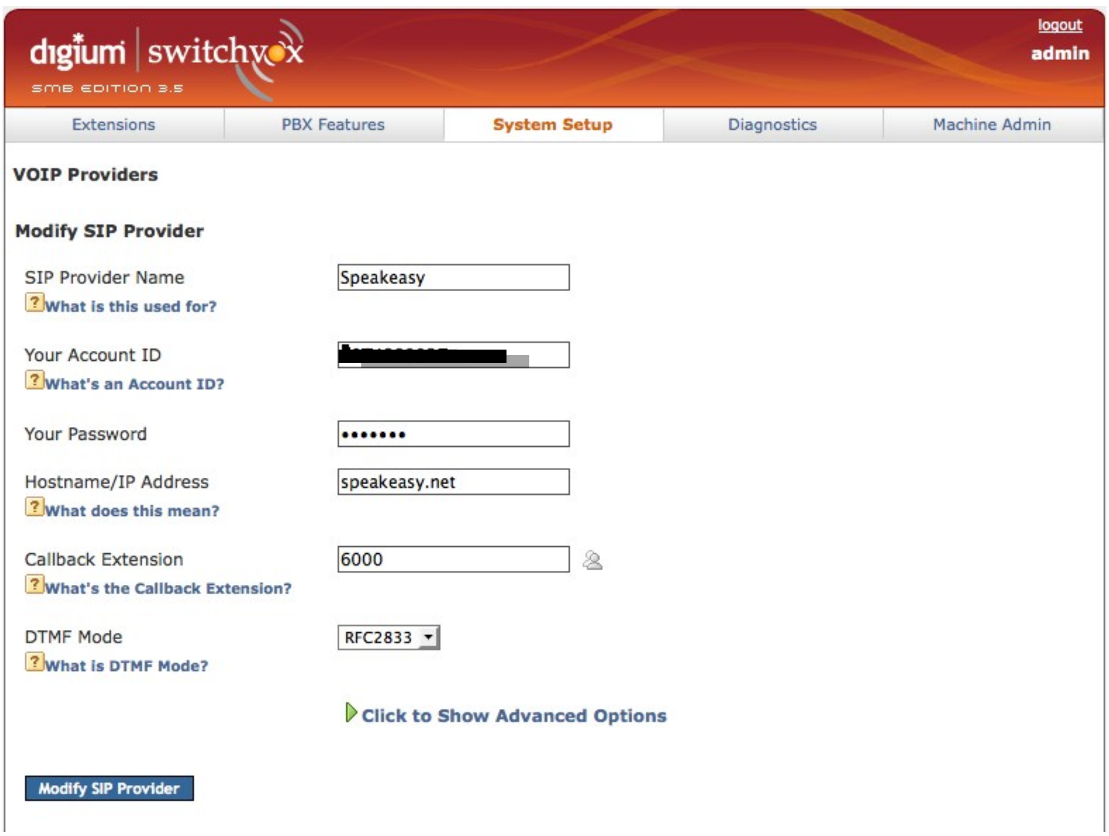
Once logged into the Admin interface of the Switchvox server follow these steps to configure Speakeasy:

Creating a SIP Account in Switchvox

* Navigate to System Setup > VOIP Providers

* Under "Add New" make sure the drop down box is selected for SIP provider and click the "**Go**" button and you will be presented with the following screen:



* **SIP Provider Name**: should be something logical that identifies this trunk as Speakeasy (i.e. "Speakeasy"), since you will be using that name later to configure calling rules.

* **Your Account ID**: is the username Speakeasy provided.

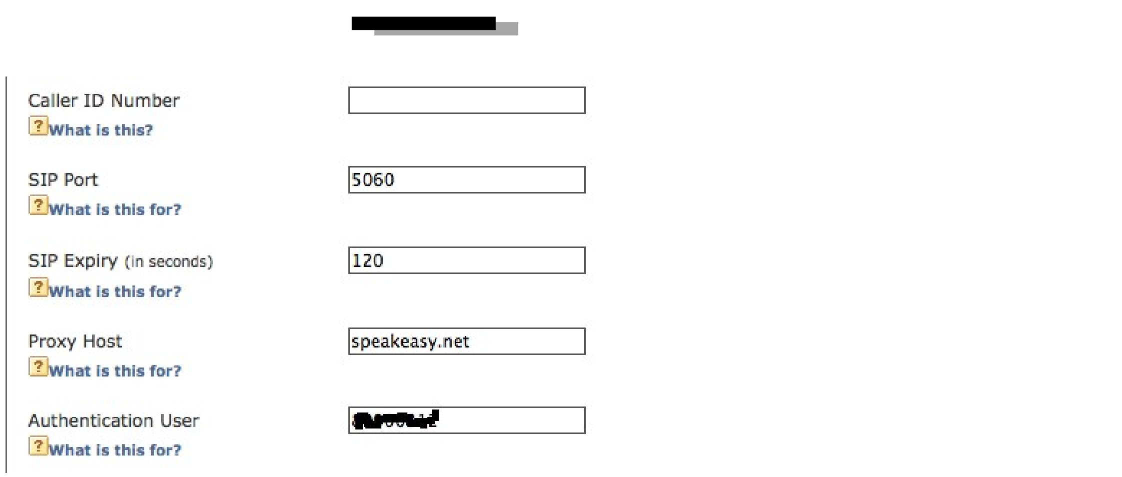
* **Your Password**: the password for digest challenge Speakeasy provided.

* **Hostname/IP Address**: This is the hostname or IP address Speakeasy has provided you for sending registrations to.

* **Callback Extension**: The default extension to ring when receiving a call over this provider. (Operator extension or IVR) This is where you tell the DID where to terminate on the Line side, either to the IVR/AA or to an Extension directly.

* **DTMF Mode**: The DTMF mode to use when sending and receiving DTMF tones to and from Speakeasy. This should be set to ‘RFC2833’.

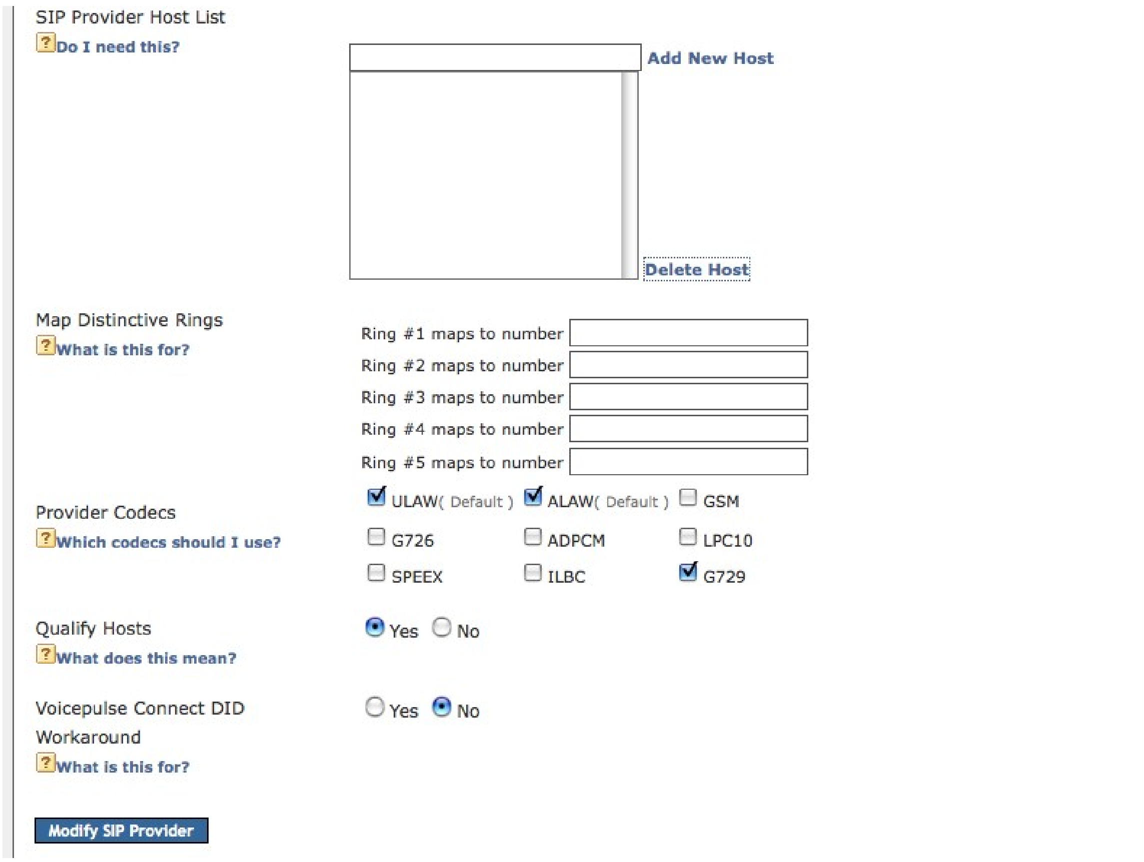
  Now click on the “**Click to Show Advanced Options**”, additional options will now appear.



* **Sip Expiry**: This field should be left at the default value of 120.

- **Proxy Host**: This is the Domain Name or IP address that Speakeasy provides you for sending a registration to. It is the Fully Qualified Domain Name for the appropriate Speakeasy Session Border Controller.

- **Authentication User**: *** What is this set to******



- **SIP Provider Host List:** If Speakeasy provides you with additional IP addresses they will be sending media on, they need be entered into this field.

- **Provider Codec's:** Speakeasy supports G.711 Ulaw and G.729

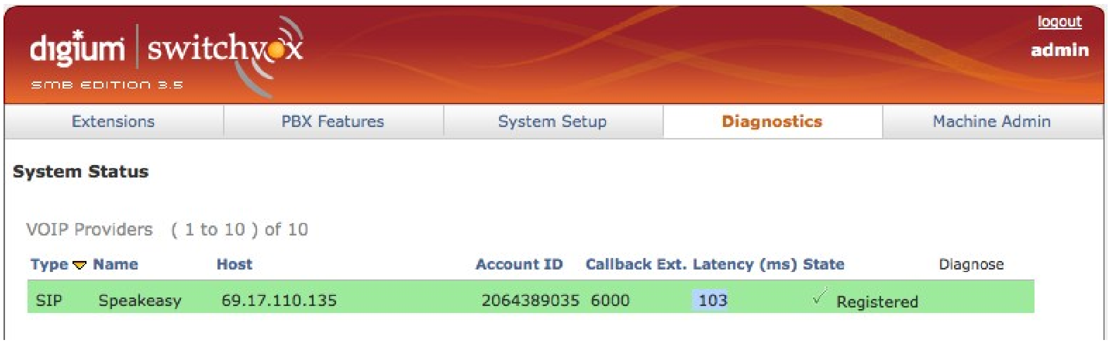
- **Qualify Hosts**: This field is optional; enabling this option allows you to view your latency to Speakeasy.

* All other fields in this section will fill in automatically; don't worry if some are blank as they are not required.

* Click the "**Modify SIP Provider**" button, your changes are now saved and the Provider should be successfully registered.

Verifying the SIP Connection

* Navigate to “Diagnostics > System Status”, this page shows the status of all VOIP peers.



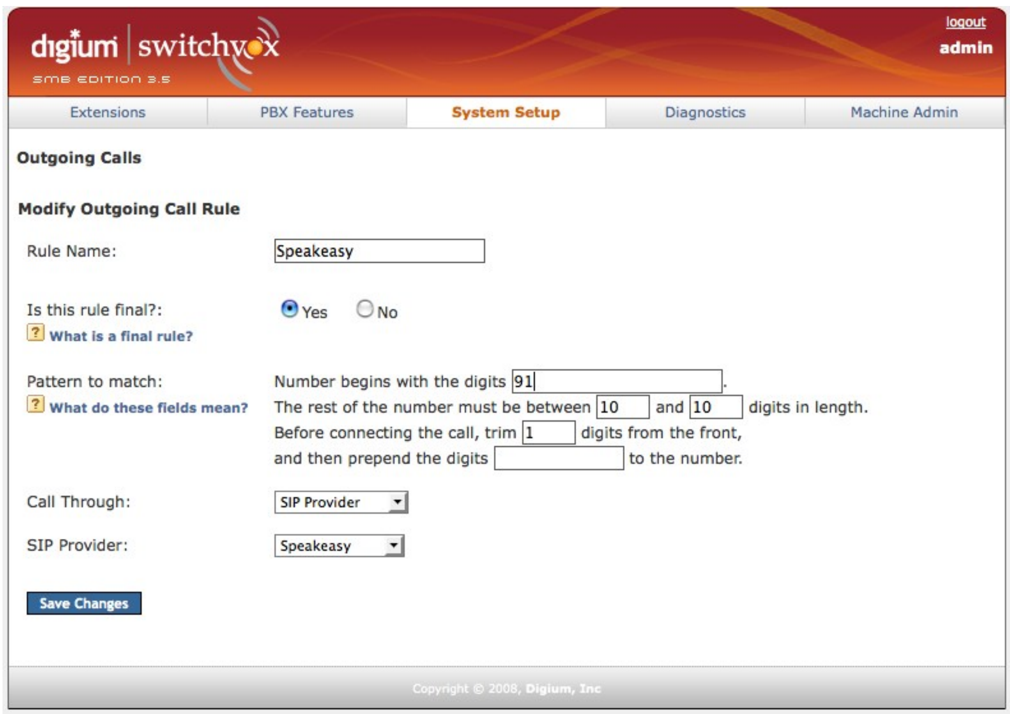
  * The above picture shows Switchvox successfully registered to Speakeasy. If the VoIP Provider is highlighted in green and the state is “Registered”, Switchvox is registered and authenticated with Speakeasy.

  * In the event there is an error registering to Speakeasy, the VoIP Provider will be highlighted in red and you will have the option to diagnose the problem with the built in mechanism.

Creating Outgoing Call Rules in Switchvox

The next step is to setup calling rules to determine which calls go through Speakeasy; Here is a standard example.

- Navigate to “System Setup > Outgoing Calls” page and click on the “**Add New Outgoing Rule**” button



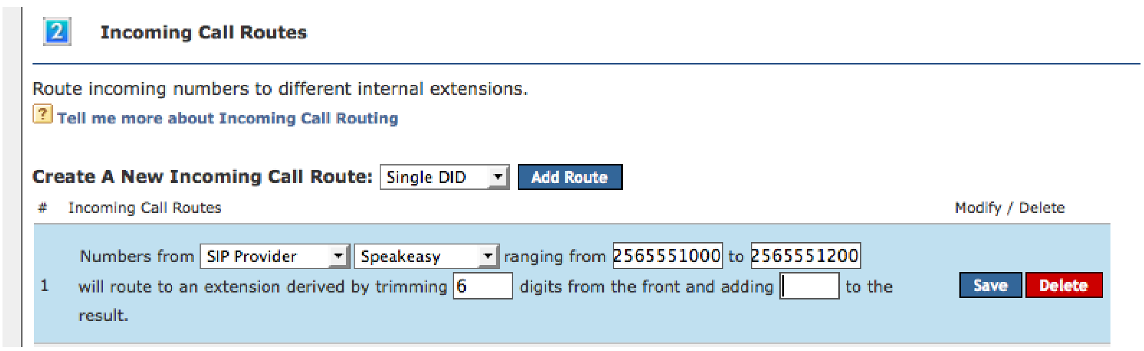
These are examples and your rules may vary based upon requirements.

- The rule shown in the picture above will take a number beginning with 91 and truncate the 9 and send the call to Speakeasy. This would be an example of dialing a long distance call beginning with a 9.

Creating Incoming Call Rules in Switchvox

The next step is to determine how calls received through Speakeasy are routed in Switchvox.

* Navigate to “System Setup >Incoming Calls” page and click on the “**Add Route**” botton



* These are examples and your rules may vary based upon requirements.

* Rule number 1 will match a range of DID’s (Direct Inward Dial numbers) and send them to the matching extension on the system.

Optional Network Configuration

If your Switchvox PBX is behind a router that performs NAT and/or there will be phones connected to Switchvox from outside the network, you will need to enable NAT port forwarding in Switchvox.

* Navigate to “Machine Admin -> Network Settings”
* Make sure the yes is selected next to “Allow Nat Port Forwarding”

Switchvox is now fully configured for Speakeasy SIP Service. If you have any questions please contact your Digium reseller or if you have a telephone support subscription you can contact Digium technical support directly at Digium technical support at (256) 428-6000.