Configuration guide for Switchvox and Bandwidth.com

This document will guide a Switchvox administrator through configuring the system to utilize bandwidth.com's SIP Trunking Service.

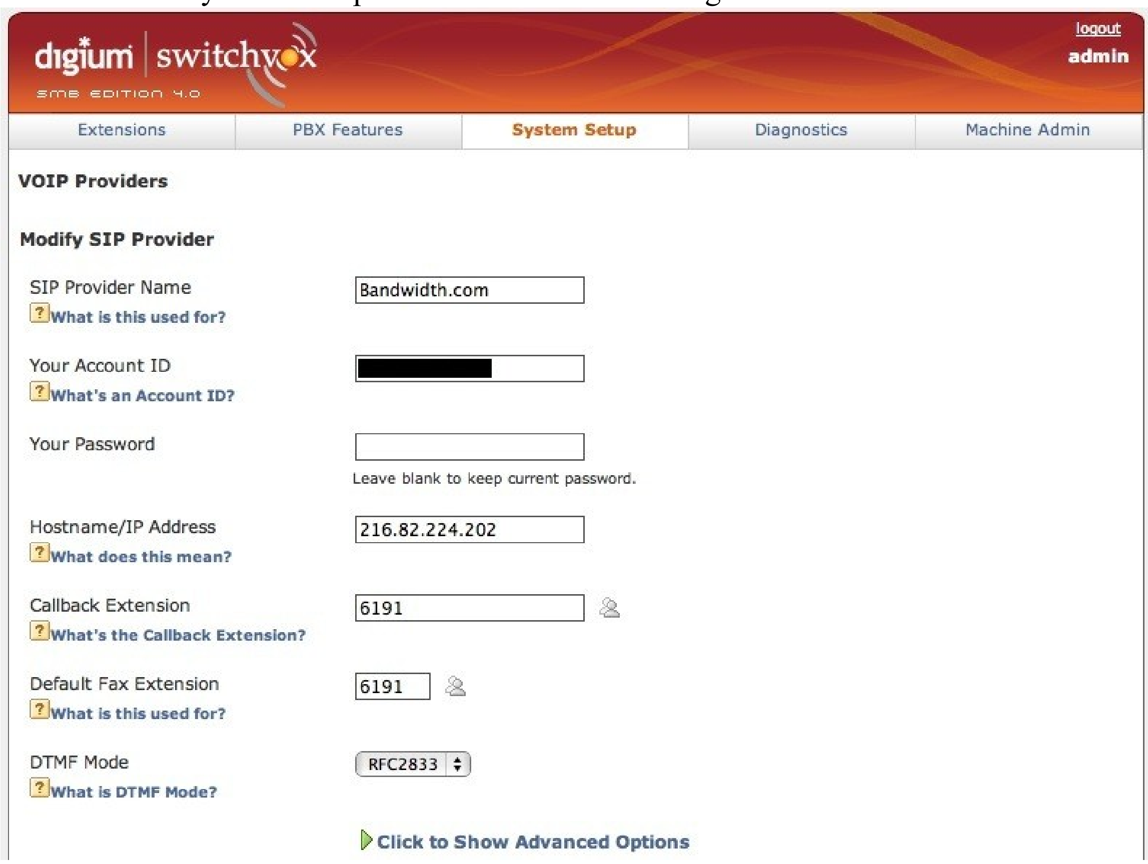
After you have the Banwidth.com account information from Bandwitdh.com, you will need to input this information into your Switchvox system through the admin web interface. The Bandwidth.com SIP Settings are found on the Order worksheet, which is provided to you by Bandwidth.com at the scheduled time of the install.

Once logged into your Switchvox server follow these steps to configure Bandwidth.com:

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 "Go" and you will be presented with the following screen:



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

* **Your Account ID**: is the username Bandwidth.com provided.

* **Your Password:** the password for digest challenge Bandwidth.com provided.

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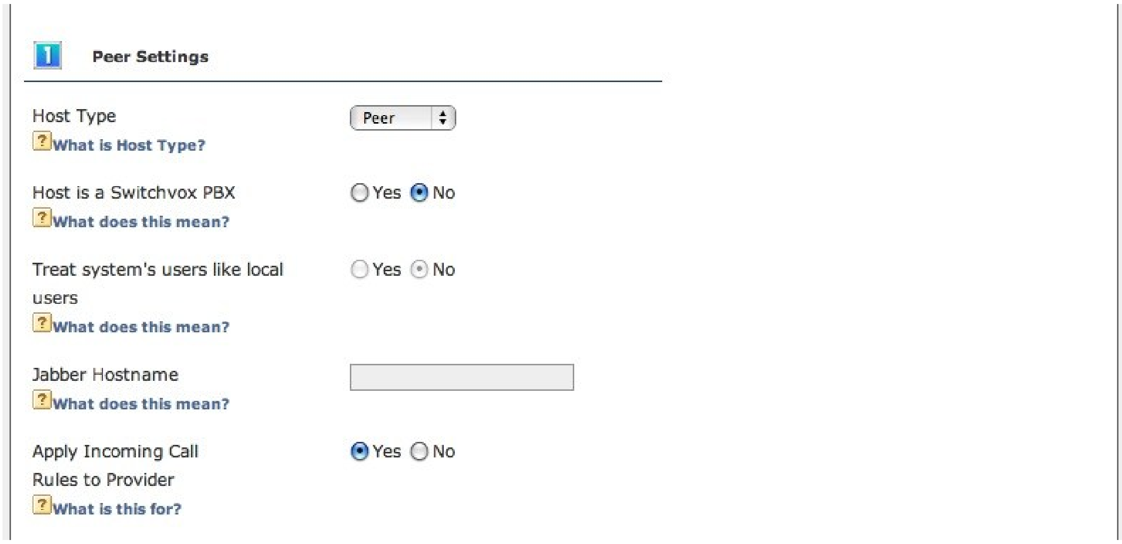
* **Hostname/IP Address**: The IP address of Bandwidth.com's proxies should go here. We currently don't reccomend using the DNS SRV method of connecting to Bandwidth.com, you could have issues with failover in the event of a failure. We expect this to be fixed in a future version of Switchvox.

* **Callback Extension**: The default extension to ring when receiving a call over this provider. (Operator extension or IVR)

* **DTMF Mode**: The DTMF mode to use when sending and receiving DTMF tones to and from Bandwidth.com. This should be set to 'Auto'.

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



* **Host Type:** Host Type must be set to Peer.
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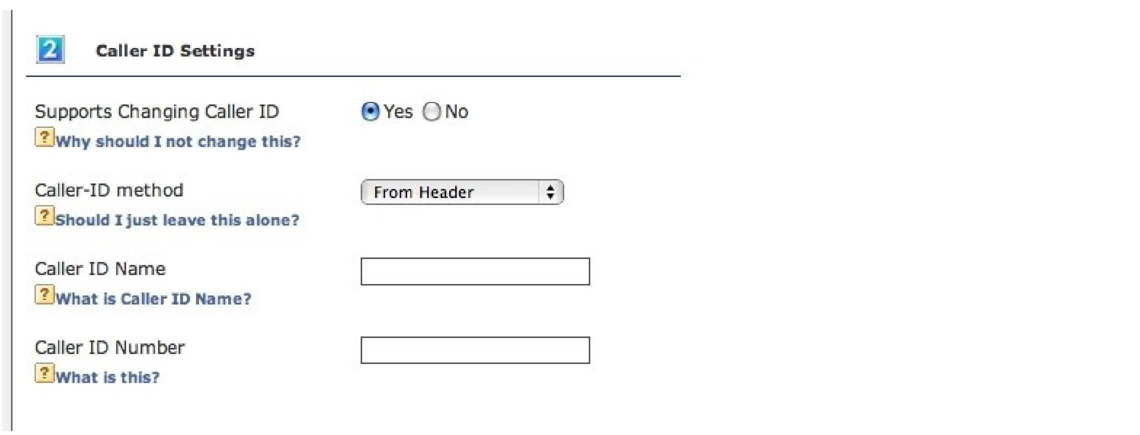
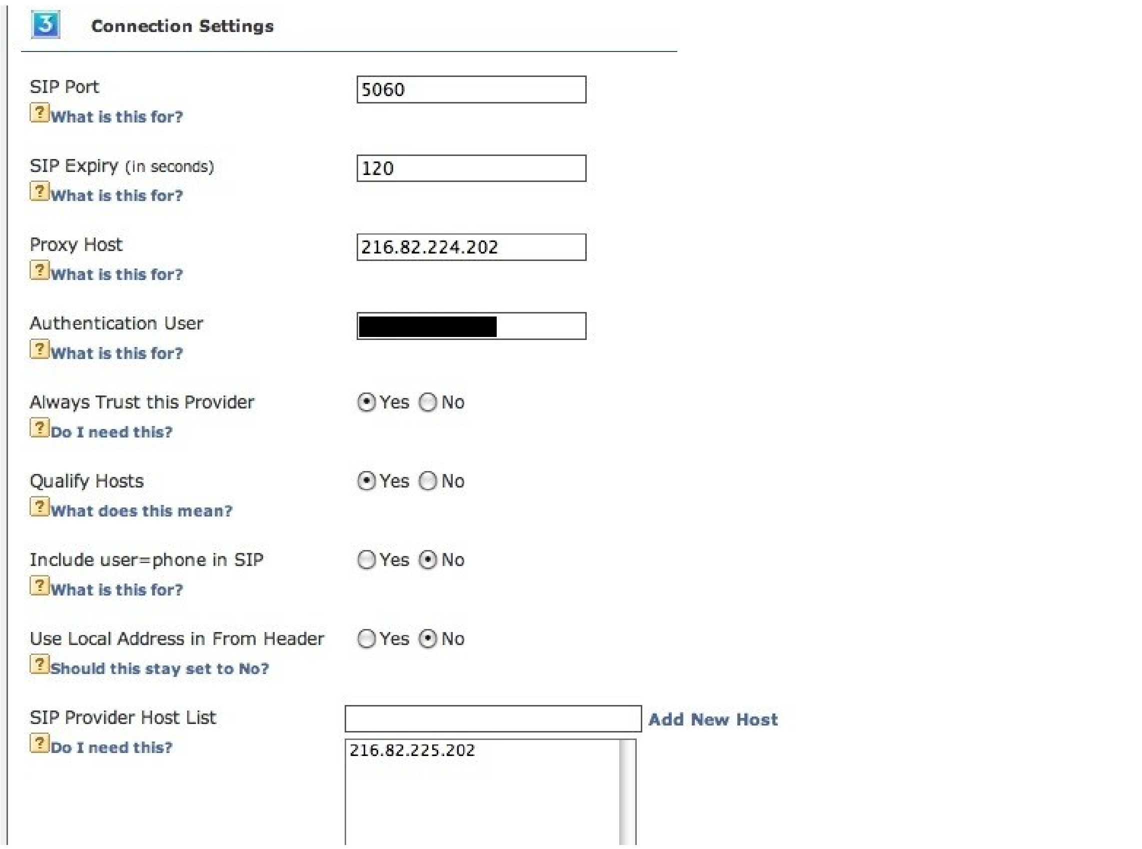
* **Apply Incoming Call Rules to Provider:** Must be set to yes in order to route calls correctly in Switchvox.



* **Supports Changing Caller ID:** Set to yes.

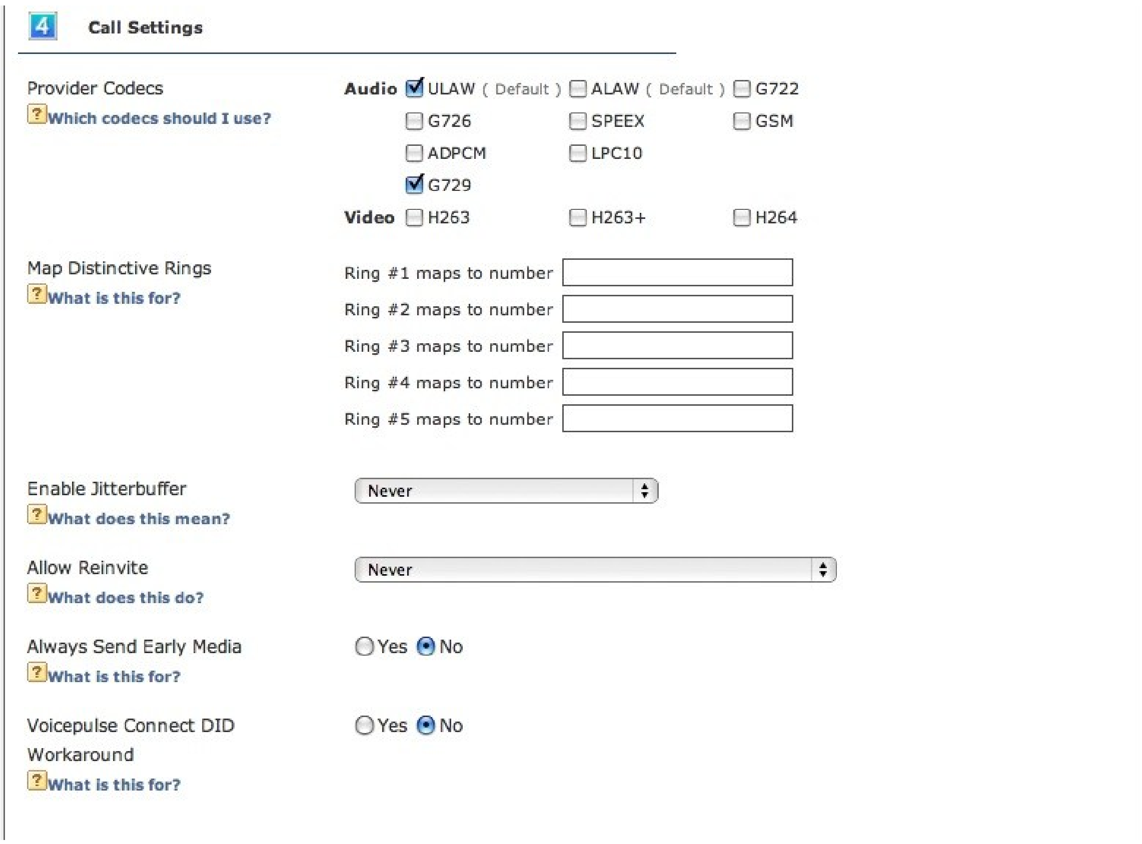
* **Caller-ID method:** Set to “From Header”



* **Sip Expiry**: The default value of 120.

* **Proxy Host**: This field is automatically filled in with the IP address used above.

* **Sip Provider Host List:** Add the other proxy IP address that Bandwidth.com provides (216.82.225.202) so that Switchvox will accept signaling from either address.
* **Qualify Hosts**: This field is optional; enabling this option allows you to view your latency to Bandwidth.com.



* **Provider Codec's:** Bandwidth.com supports G.711 uLaw and G.729.
* All other fields on this page will fill in automatically; don't worry if some are blank as they are not required.
* Click "Modify SIP Provider", your changes are now saved and the Provider should be successfully connected.

Verifying the SIP Connection

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

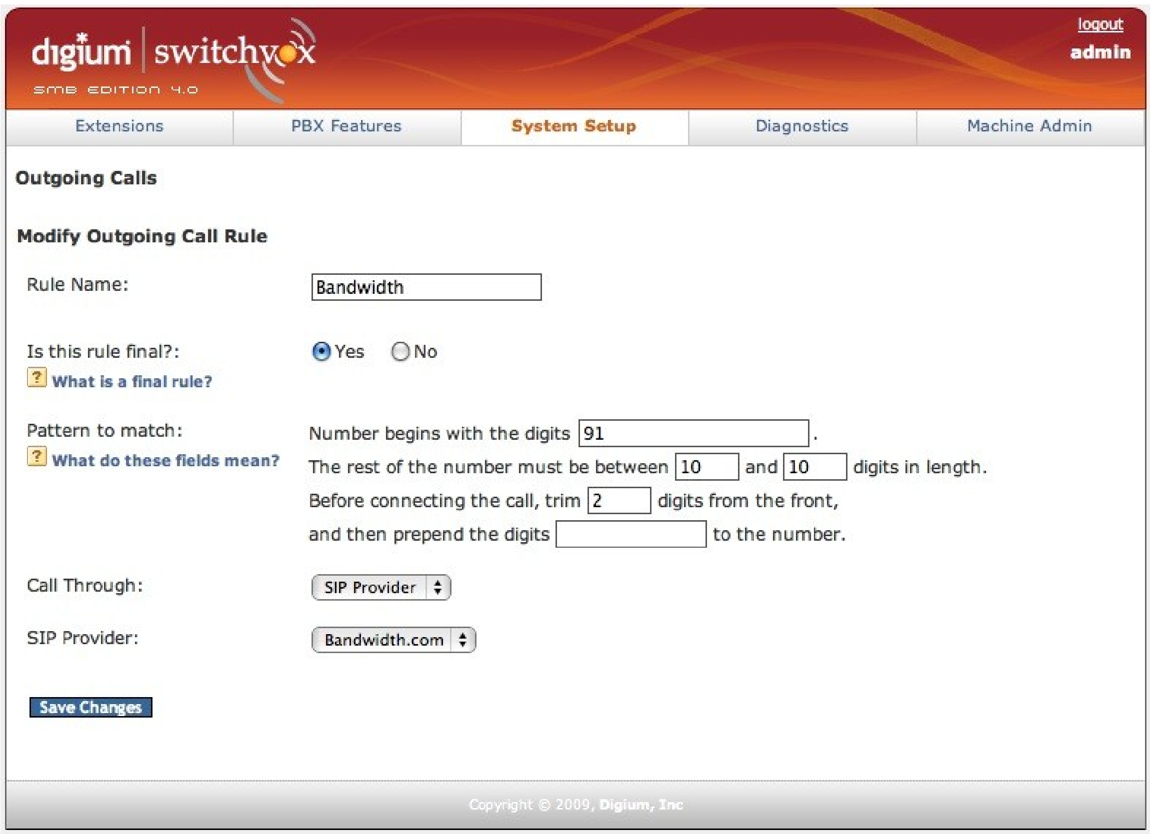
* The above picture shows Switchvox successfully connectedto Bandwidth.com. If the VoIP Provider is highlighted in green and the state is "Ok", Switchvox is connected and authenticated with Bandwidth.com.

* In the event there is an error connecting to Bandwidth.com, 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 Bandwidth.com; Here is a standard example.

* Navigate to “System Setup > Outgoing Calls” page and click “Add New Outgoing Rule” These are examples and your rules may vary based upon requirements.

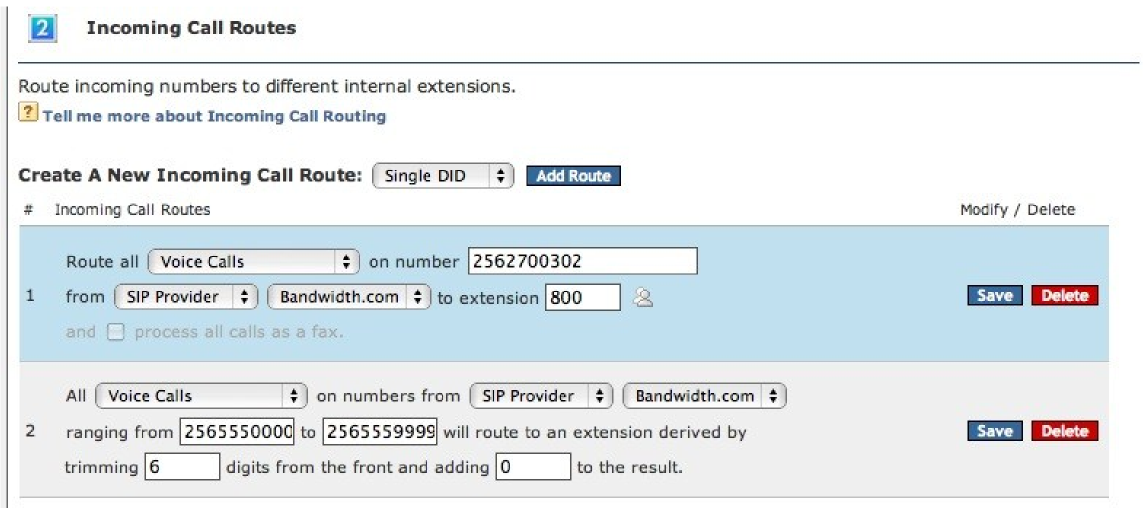


* The rule shown in the picture above will take a number beginning with 91 and , truncate the 91 and send the call to Bandwidth.com. You must contact Bandwidth.com installations and let them know that your call digits must be set at 10 digit.

Creating Incoming Call Rules in Switchvox

Now that outgoing calls route correctly, you will need to setup where incoming calls are routed.

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



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

* Rule number 1 will match a range of DID’s and send them to the matching extension on the system.
* Rule number 2 will match one DID and send it to an IVR. (e.g. the Bandwidth.com company number)

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 need to set an option 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 Bandwidth.com SIP Trunking. If you have any questions please contact Digium technical support at +1-256-428-6000 or Bandwidth.com technical support at +1-919-297-1100