Revision 0.1 – July 17, 2009

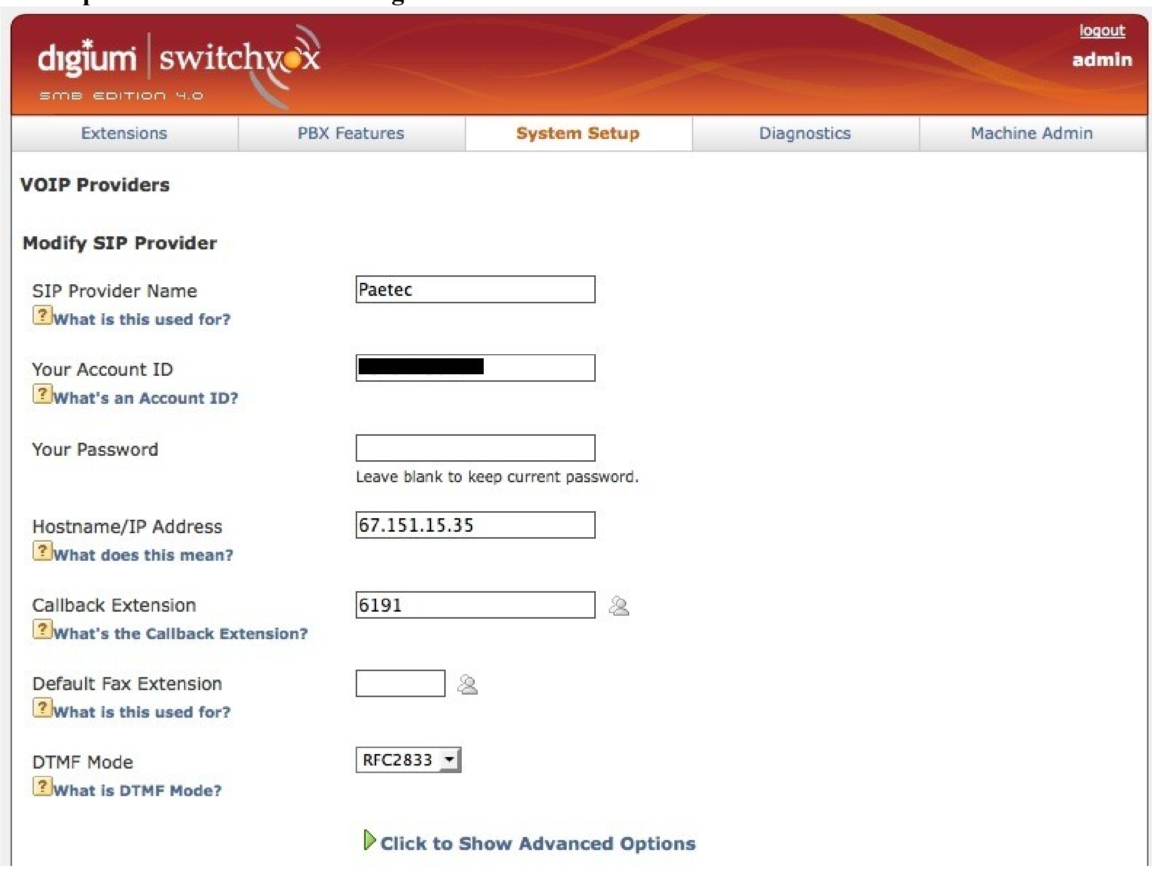
Configuration guide for Switchvox and PAETEC

This document will guide a Switchvox administrator through configuring the system to utilize PAETEC's Genband and LGP platforms.

Once logged into your Switchvox server follow these steps to configure PAETEC:

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 PAETEC (i.e. "Paetec"), since you will be using that name later to configure calling rules.

Your Account ID: is the username/DID PAETEC provided.

Your Password: Place your username here, this service doesn't require registration so this field is ignored.

Hostname/IP Address: The IP address for signaling PAETEC provides should go here.

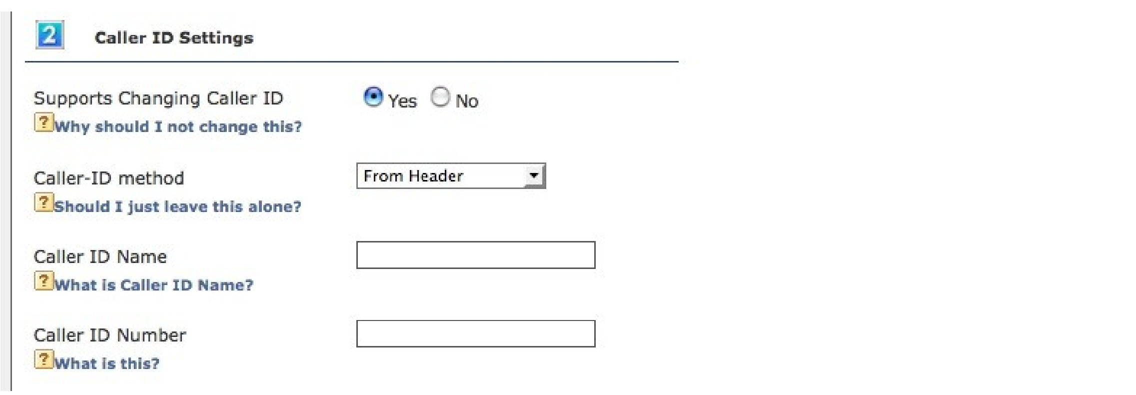
Callback Extension: The default extension to ring when receiving a call over this provider. (Operator extension or IVR)

DTMF Mode: The DTMF mode to use when sending and receiving DTMF tones to and from PAETEC. This should be set to 'RFC2833'.

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

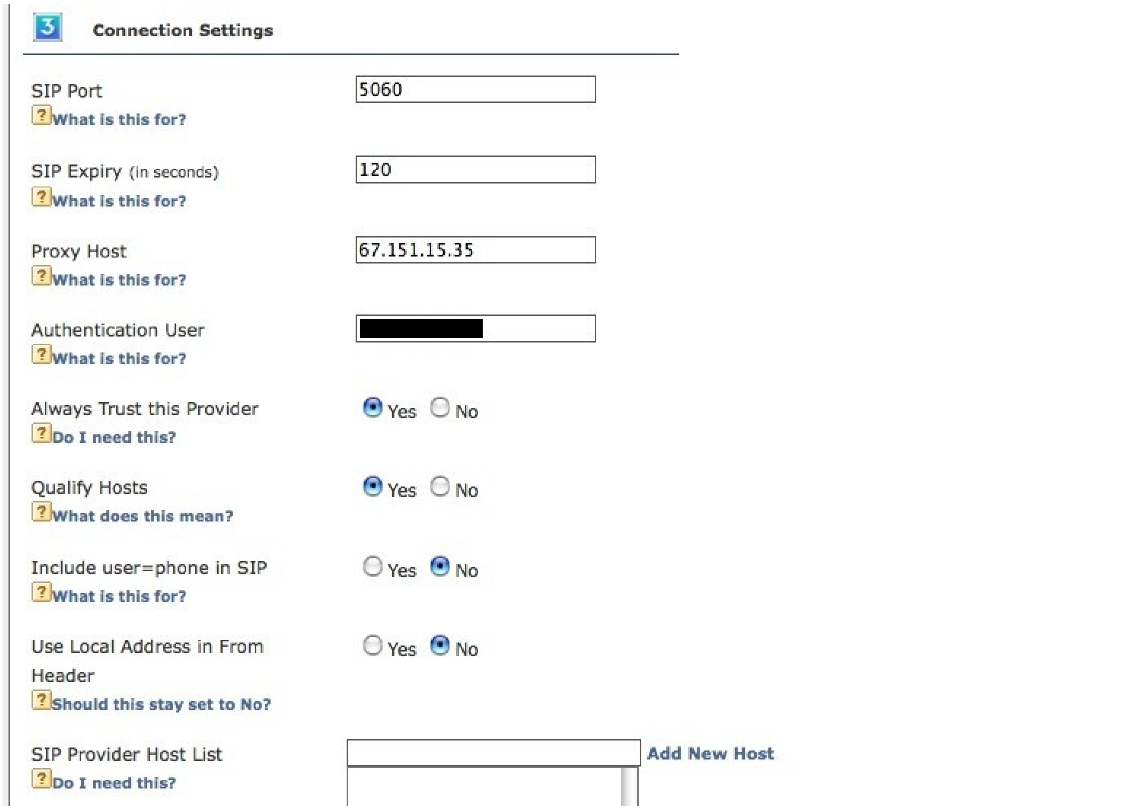
Host Type: Host Type must be set to Peer.

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.

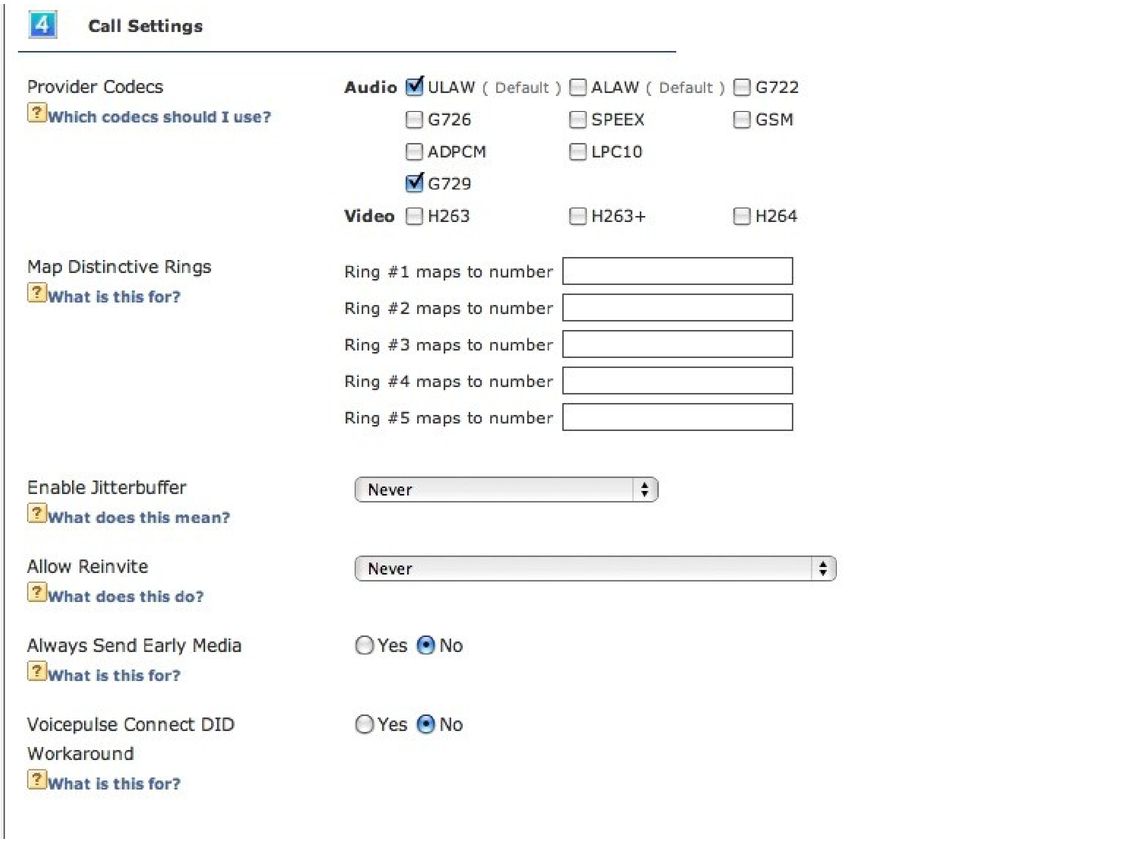
Caller-ID method: Set to "From Header"



Sip Expiry: The default value of 120.

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

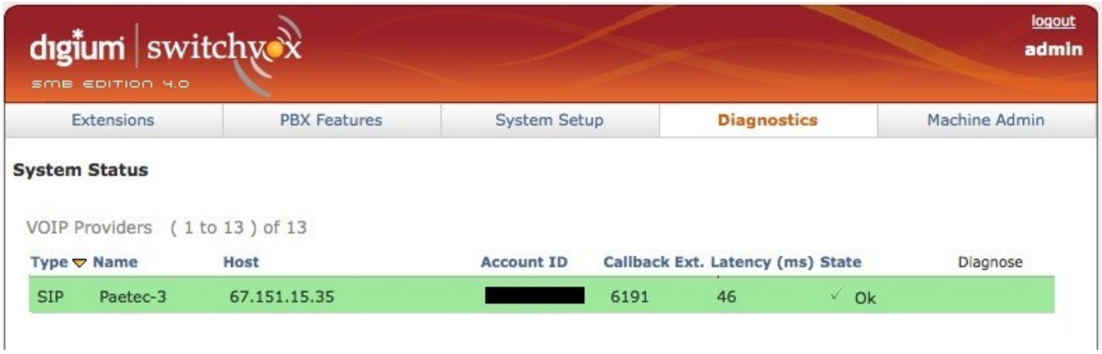
Qualify Hosts: This field is optional; enabling this option allows you to view your latency to PAETEC.



Provider Codec's: PAETEC 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.
Navigate to "Diagnostics > System Status", this page shows the status of all VOIP peers.

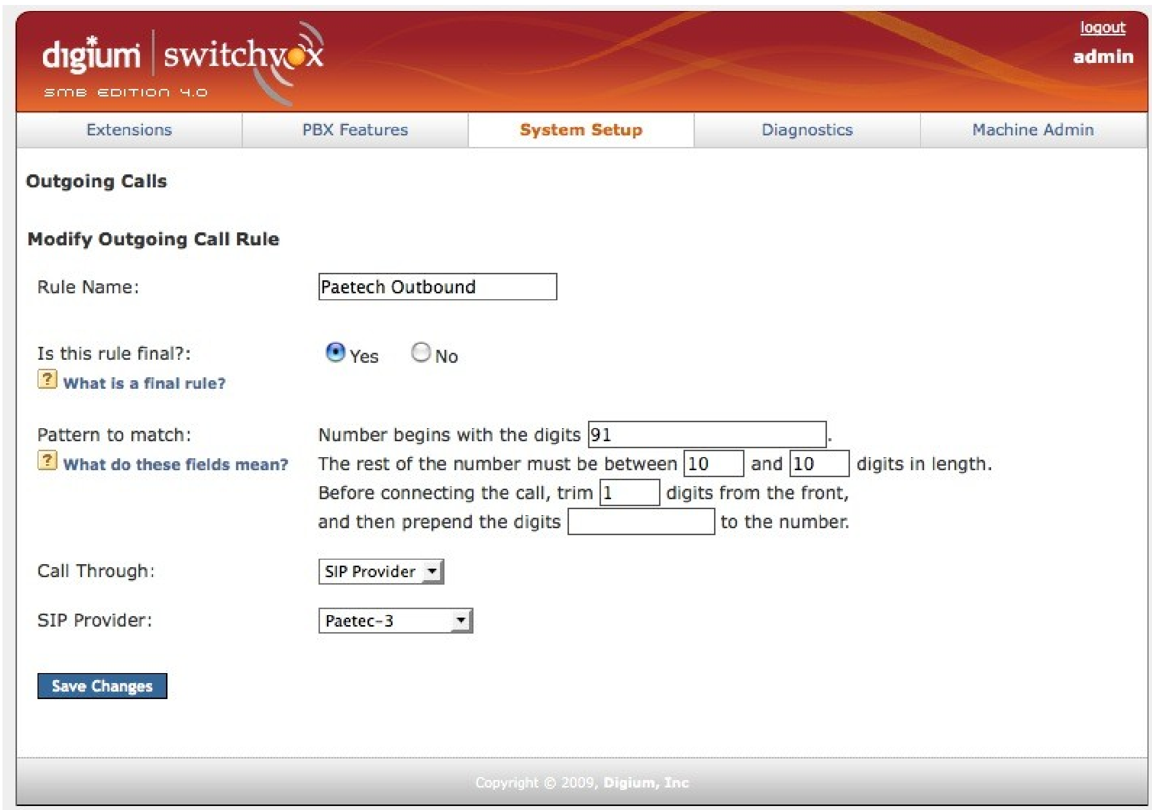


The above picture shows Switchvox successfully connected to PAETEC. If the VoIP Provider is highlighted in green and the state is “Ok”, Switchvox is connected and authenticated with PAETEC.

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

The next step is to setup calling rules to determine which calls go through PAETEC; 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.



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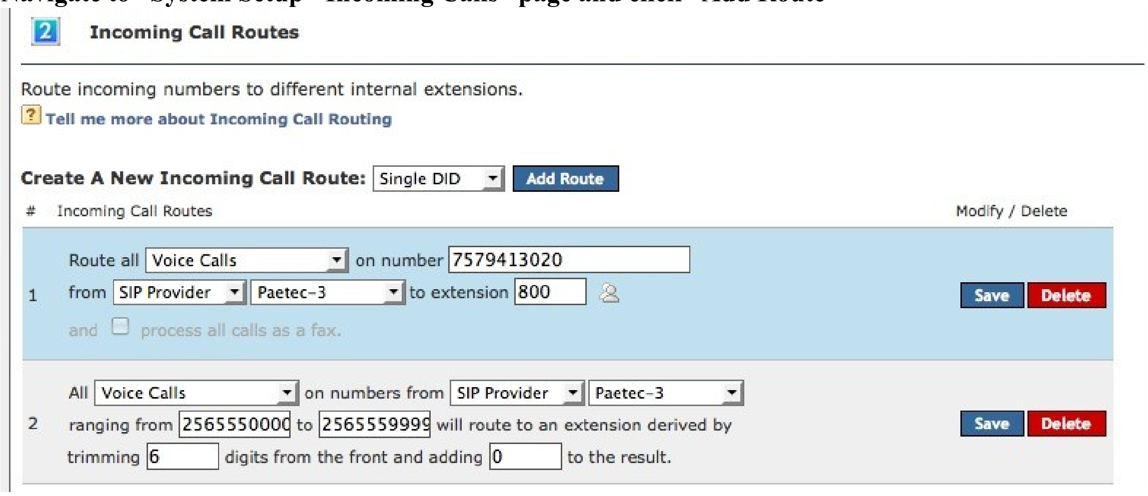
The rule shown in the picture above will take a number beginning with 91 and , truncate the 9 and send the call to PAETEC. Now that outgoing calls route correctly, you will need to setup where incoming calls are routed.

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 one DID and send it to an IVR. (e.g. the company number)

Rule number 2 will match a range of DID’s and send them to the matching extension on the system.

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 PAETEC SIP Trunking. If you have any questions please contact Digium technical support at +1-256-428-6000.

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PAETEC Configuration & Support
PAETEC will configure SIP trunks on its network and provide customers with IP addresses of SIP Proxy, and phone numbers assigned to customers before scheduled service activation date. SIP trunks are offered through MPLS connections with proper QoS to guarantee the security and quality of voice traffic. MPLS configuration will be done by PAETEC throughout its network and all the way to the CPEs provided by PAETEC.

If you are in need of technical support for PAETEC's Dynamic IP SIP Trunking, please contact our Customer Care department at 877.340.2600 or customercare@paetec.com. When contacting Customer Care, please include the customer account number.

Version Information

PAETEC's Dynamic IP SIP Trunking	
LGP MGC/MG	6.3.1.2SP8
Genband G9	6.0.60
Acme Packet Session Border Controller	2.1.0 Patch 67

Digium	
Switchvox	4.0 version 18775