

## Configuration guide for Switchvox and PAETEC



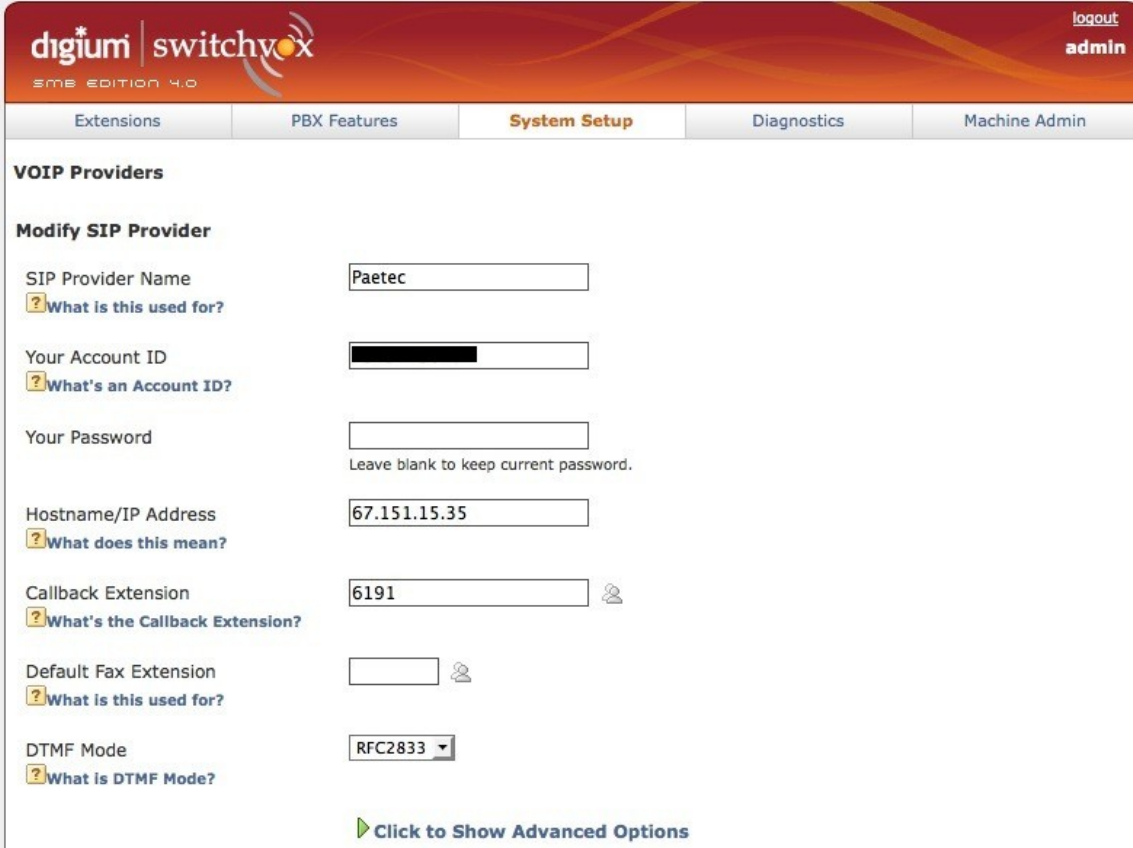
**This document will guide a Switchvox administrator through configuring the system to utilize PAETEC's Genband and LGP platforms.**



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:



The screenshot shows the 'System Setup' tab selected in the navigation bar. The 'VOIP Providers' section is active, displaying the 'Modify SIP Provider' form. The form includes the following fields and values:

- SIP Provider Name: Paetec
- Your Account ID: [Redacted]
- Your Password: [Empty]
- Hostname/IP Address: 67.151.15.35
- Callback Extension: 6191
- Default Fax Extension: [Empty]
- DTMF Mode: RFC2833

A link labeled 'Click to Show Advanced Options' is located at the bottom of the form.

**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.

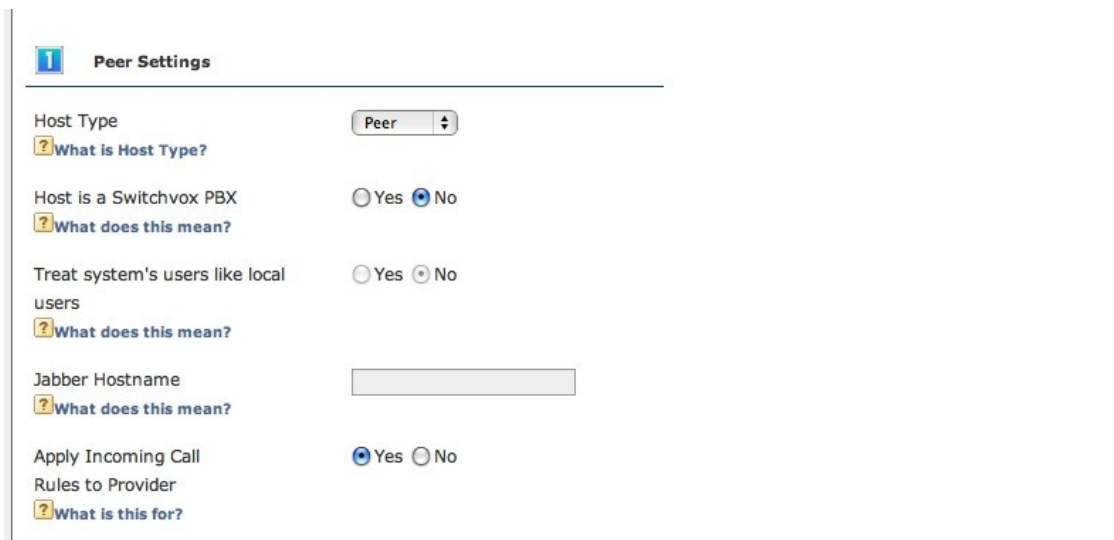


**Hostname/IP Address:** The IP address for signaling PAETEC provides should go here.

**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.

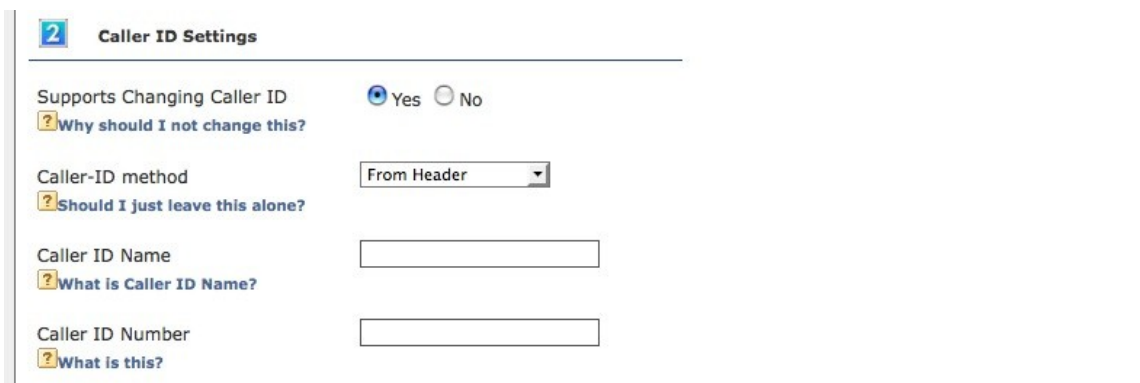


The screenshot shows the 'Peer Settings' section of a configuration interface. It includes the following fields and options:

- Host Type:** A dropdown menu set to 'Peer'. A help icon and the text '? What is Host Type?' are present.
- Host is a Switchvox PBX:** Radio buttons for 'Yes' and 'No', with 'No' selected. A help icon and the text '? What does this mean?' are present.
- Treat system's users like local users:** Radio buttons for 'Yes' and 'No', with 'No' selected. A help icon and the text '? What does this mean?' are present.
- Jabber Hostname:** An empty text input field. A help icon and the text '? What does this mean?' are present.
- Apply Incoming Call Rules to Provider:** Radio buttons for 'Yes' and 'No', with 'Yes' selected. A help icon and the text '? What is this for?' are present.

**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.

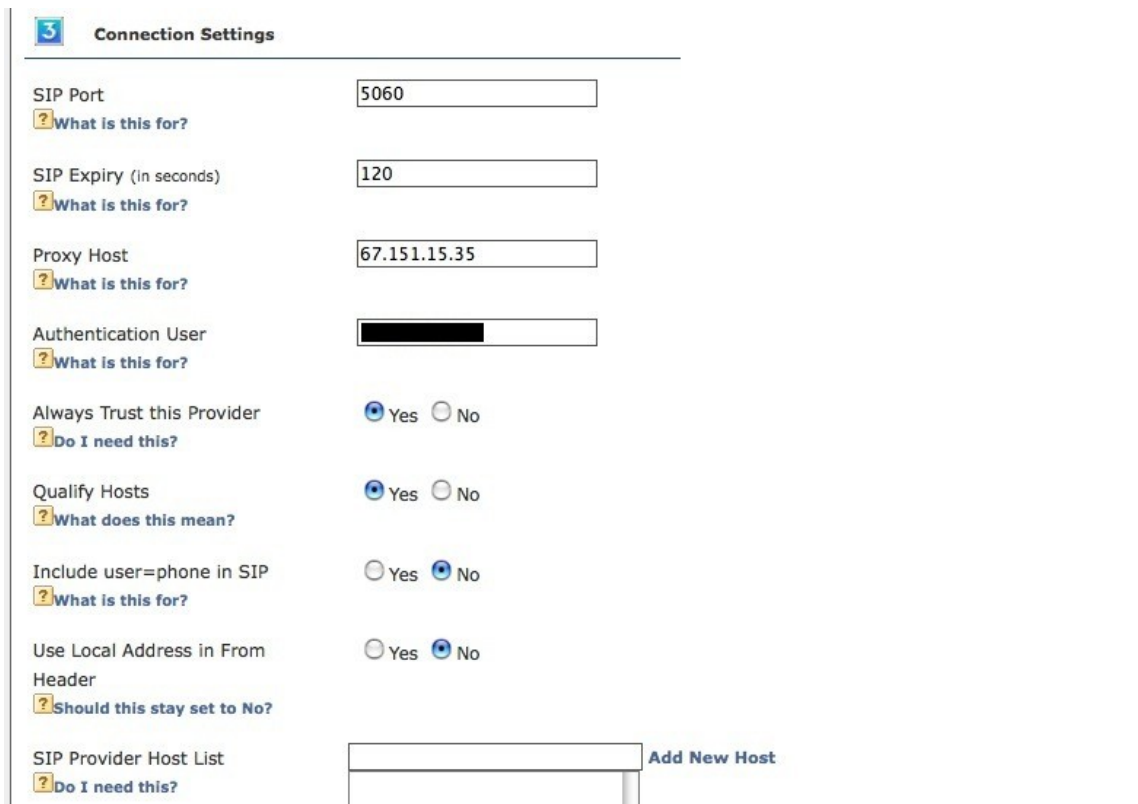


The screenshot shows the 'Caller ID Settings' section of a configuration interface. It includes the following fields and options:

- Supports Changing Caller ID:** Radio buttons for 'Yes' and 'No', with 'Yes' selected. A help icon and the text '? Why should I not change this?' are present.
- Caller-ID method:** A dropdown menu set to 'From Header'. A help icon and the text '? Should I just leave this alone?' are present.
- Caller ID Name:** An empty text input field. A help icon and the text '? What is Caller ID Name?' are present.
- Caller ID Number:** An empty text input field. A help icon and the text '? What is this?' are present.

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

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



The screenshot shows the 'Connection Settings' page in Asterisk. It contains several configuration fields and options:

- SIP Port:** 5060
- SIP Expiry (in seconds):** 120
- Proxy Host:** 67.151.15.35
- Authentication User:** [Redacted]
- Always Trust this Provider:**  Yes  No
- Qualify Hosts:**  Yes  No
- Include user=phone in SIP:**  Yes  No
- Use Local Address in From Header:**  Yes  No
- SIP Provider Host List:** [Empty table] [Add New Host](#)

**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.**



4 **Call Settings**

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**Provider Codecs**  
? What codecs should I use?

**Audio**  ULAW ( Default )  ALAW ( Default )  G722  
 G726  SPEEX  GSM  
 ADPCM  LPC10  
 G729

**Video**  H263  H263+  H264

**Map Distinctive Rings**  
? What is this for?

Ring #1 maps to number

Ring #2 maps to number

Ring #3 maps to number

Ring #4 maps to number

Ring #5 maps to number

**Enable Jitterbuffer**  
? What does this mean?

**Allow Reinvite**  
? What does this do?

**Always Send Early Media**  
? What is this for?

Yes  No

**Voicepulse Connect DID Workaround**  
? What is this for?

Yes  No

**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 screenshot shows the 'Diagnostics' tab in the digium switchvox interface. Under 'System Status', there is a section for 'VOIP Providers ( 1 to 13 ) of 13'. A table lists the providers, with one entry highlighted in green:

Type	Name	Host	Account ID	Callback Ext.	Latency (ms)	State	Diagnose
SIP	Paetec-3	67.151.15.35	[REDACTED]	6191	46	✓ Ok	

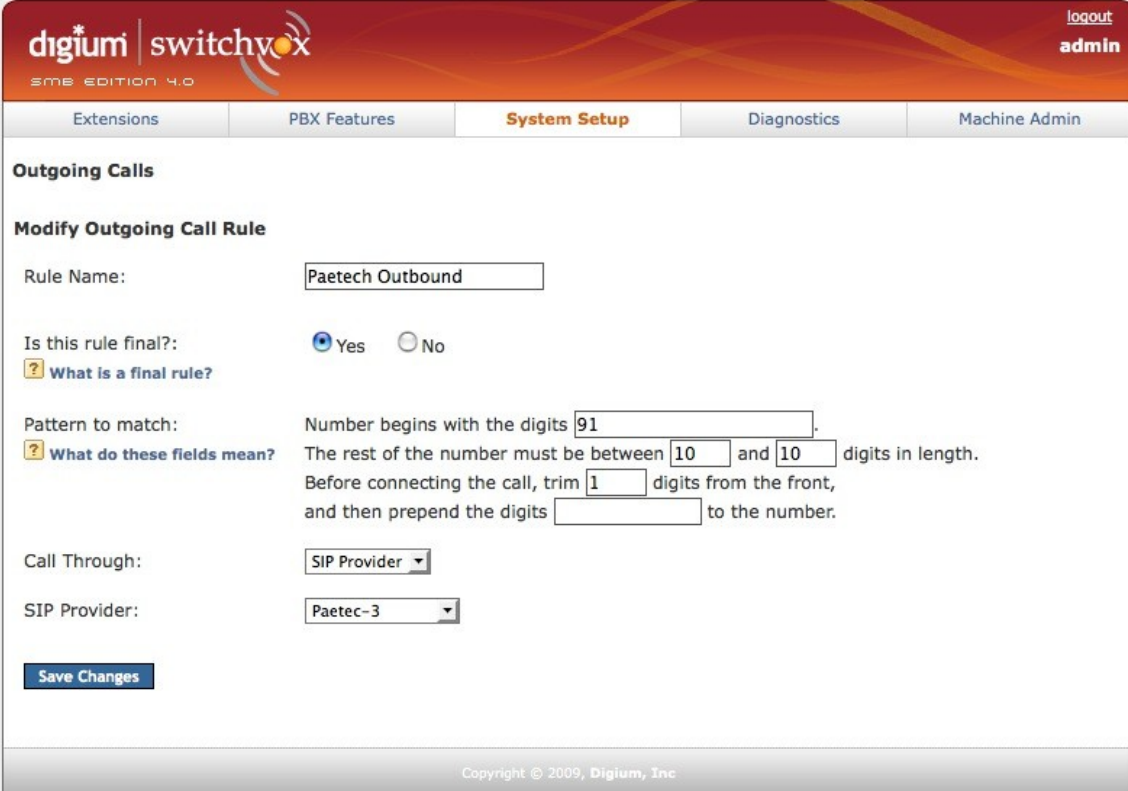
The above picture shows Switchvox successfully connected to PAETEC. If the VoIP Provider is highlighted in green and the state is “Ok”, Switchvox is connected and authenticated with PAETEC.

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.



**digium | switchvox** logout  
admin  
SMB EDITION 4.0

Extensions | PBX Features | **System Setup** | Diagnostics | Machine Admin

**Outgoing Calls**

**Modify Outgoing Call Rule**

Rule Name:

Is this rule final?:  Yes  No  
 ? What is a final rule?

Pattern to match: .  
 ? What do these fields mean? The rest of the number must be between  and  digits in length.  
 Before connecting the call, trim  digits from the front,  
 and then prepend the digits  to the number.

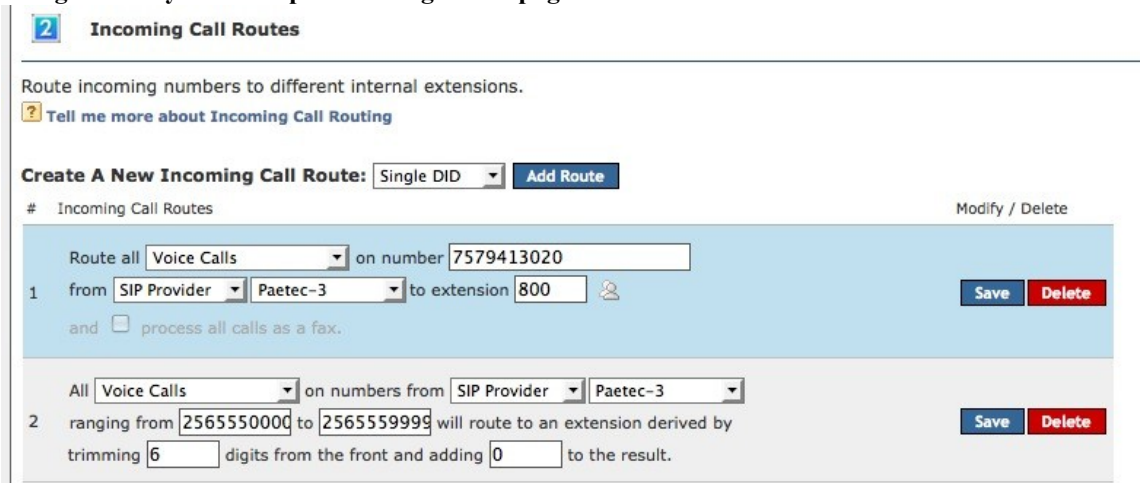
Call Through:

SIP Provider:

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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”



**2 Incoming Call Routes**

Route incoming numbers to different internal extensions.  
[? Tell me more about Incoming Call Routing](#)

**Create A New Incoming Call Route:**

#	Incoming Call Routes	Modify / Delete
1	Route all <input type="text" value="Voice Calls"/> on number <input type="text" value="7579413020"/> from <input type="text" value="SIP Provider"/> <input type="text" value="Paetec-3"/> to extension <input type="text" value="800"/> <input type="button" value="Save"/> <input type="button" value="Delete"/> and <input type="checkbox"/> process all calls as a fax.	
2	All <input type="text" value="Voice Calls"/> on numbers from <input type="text" value="SIP Provider"/> <input type="text" value="Paetec-3"/> ranging from <input type="text" value="2565550000"/> to <input type="text" value="2565559999"/> will route to an extension derived by trimming <input type="text" value="6"/> digits from the front and adding <input type="text" value="0"/> to the result. <input type="button" value="Save"/> <input type="button" value="Delete"/>	

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”



Allow Nat  Yes  No  
Port Forwarding  
[? What does this mean?](#)

Switchvox is now fully configured for PAETEC SIP Trunking. If you have any questions please contact Digium technical support at +1-256-428-6000.





**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](mailto: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

