

## Configuration guide for Switchvox and Bandwidth.com



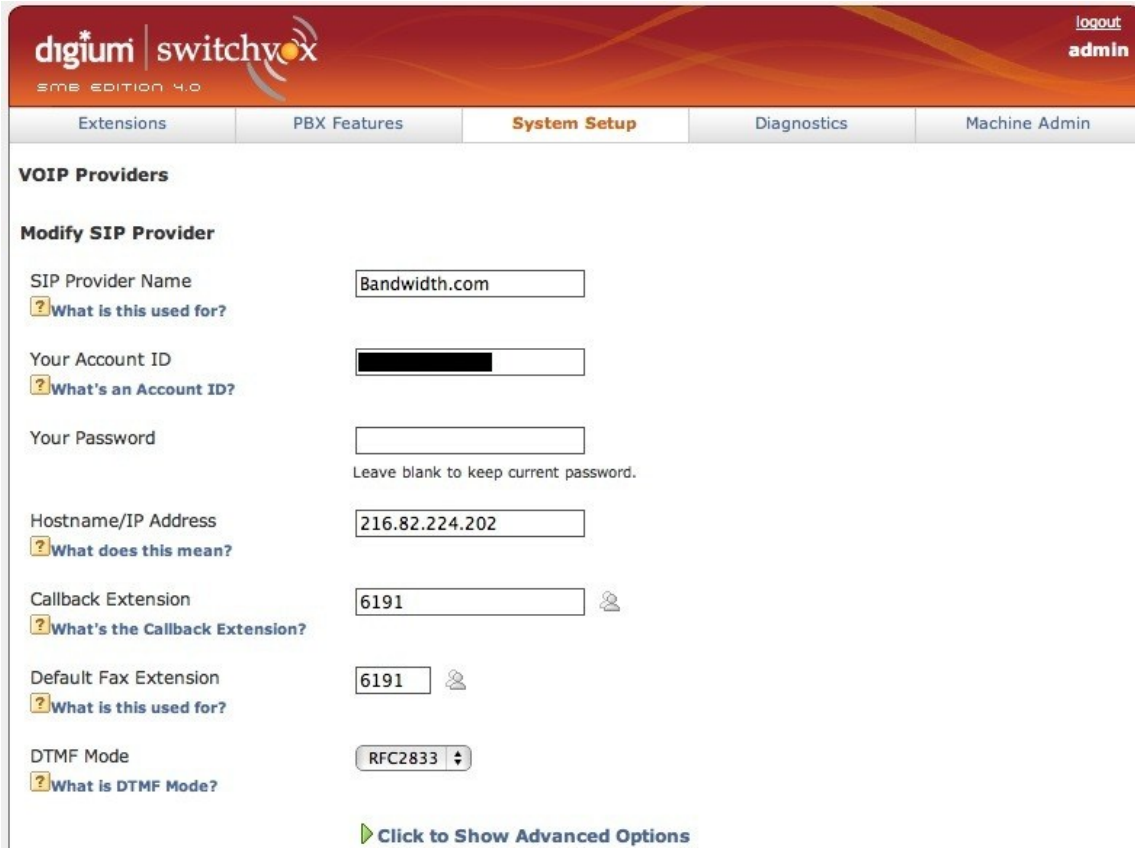
This document will guide a Switchvox administrator through configuring the system to utilize bandwidth.com's SIP Trunking Service.

After you have the Bandwidth.com account information from Bandwidth.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:



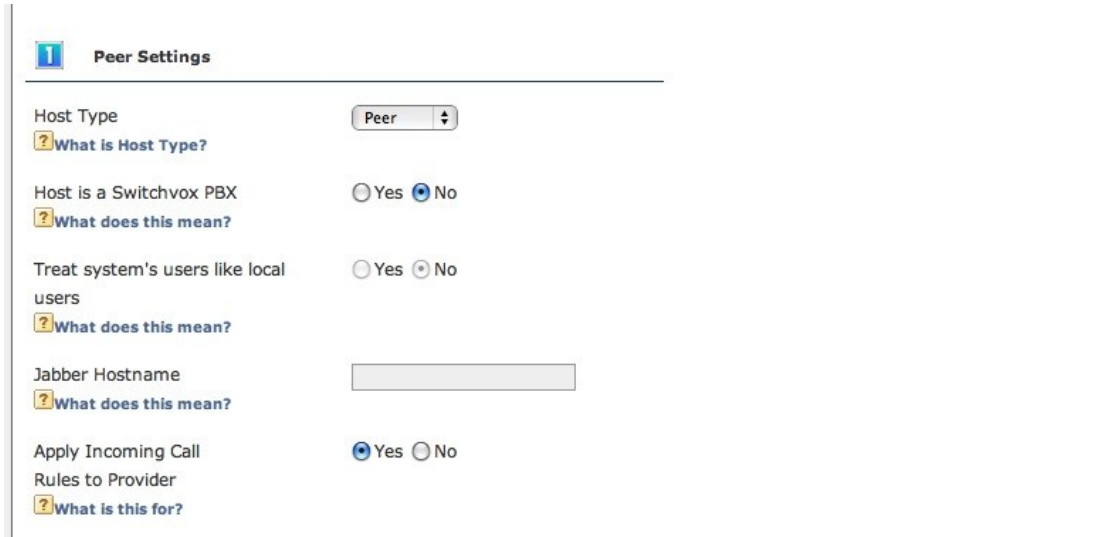
The screenshot shows the Switchvox administration interface. The top navigation bar includes 'Extensions', 'PBX Features', 'System Setup' (highlighted), 'Diagnostics', and 'Machine Admin'. The 'System Setup' section is titled 'VOIP Providers' and contains a 'Modify SIP Provider' form. The form fields are: SIP Provider Name (Bandwidth.com), Your Account ID (redacted), Your Password (empty), Hostname/IP Address (216.82.224.202), Callback Extension (6191), Default Fax Extension (6191), and DTMF Mode (RFC2833). A 'Click to Show Advanced Options' link is located at the bottom of the form.

- \* **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.



- \* **Hostname/IP Address:** The IP address of Bandwidth.com's proxies should go here. We currently don't recommend 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.



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

- Host Type:** A dropdown menu set to 'Peer'. A help icon and 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 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 text 'What does this mean?' are present.
- Jabber Hostname:** An empty text input field. A help icon and 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 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 page. It includes the following fields and options:

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

- \* **Supports Changing Caller ID:** Set to yes.
- \* **Caller-ID method:** Set to “From Header”



**3 Connection Settings**

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SIP Port   
? What is this for?

SIP Expiry (in seconds)   
? What is this for?

Proxy Host   
? What is this for?

Authentication User   
? What is this for?

Always Trust this Provider  Yes  No  
? Do I need this?

Qualify Hosts  Yes  No  
? What does this mean?

Include user=phone in SIP  Yes  No  
? What is this for?

Use Local Address in From Header  Yes  No  
? Should this stay set to No?

SIP Provider Host List   
? Do I need this?

216.82.225.202
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[Add New Host](#)

- \* **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.



**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

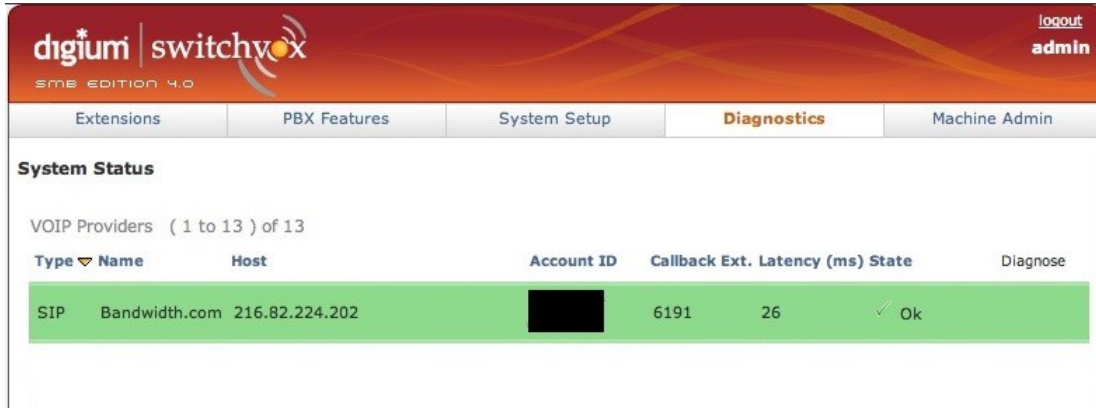
- \* **Provider Codec's:** Bandwidth.com supports G.711 uLaw and G.729.
- \* All other fields on this page will fill in automatically; don't worry if some are blank as they are not required.
- \* Click "Modify SIP Provider", your changes are now saved and the Provider should be successfully connected.





## Verifying the SIP Connection

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



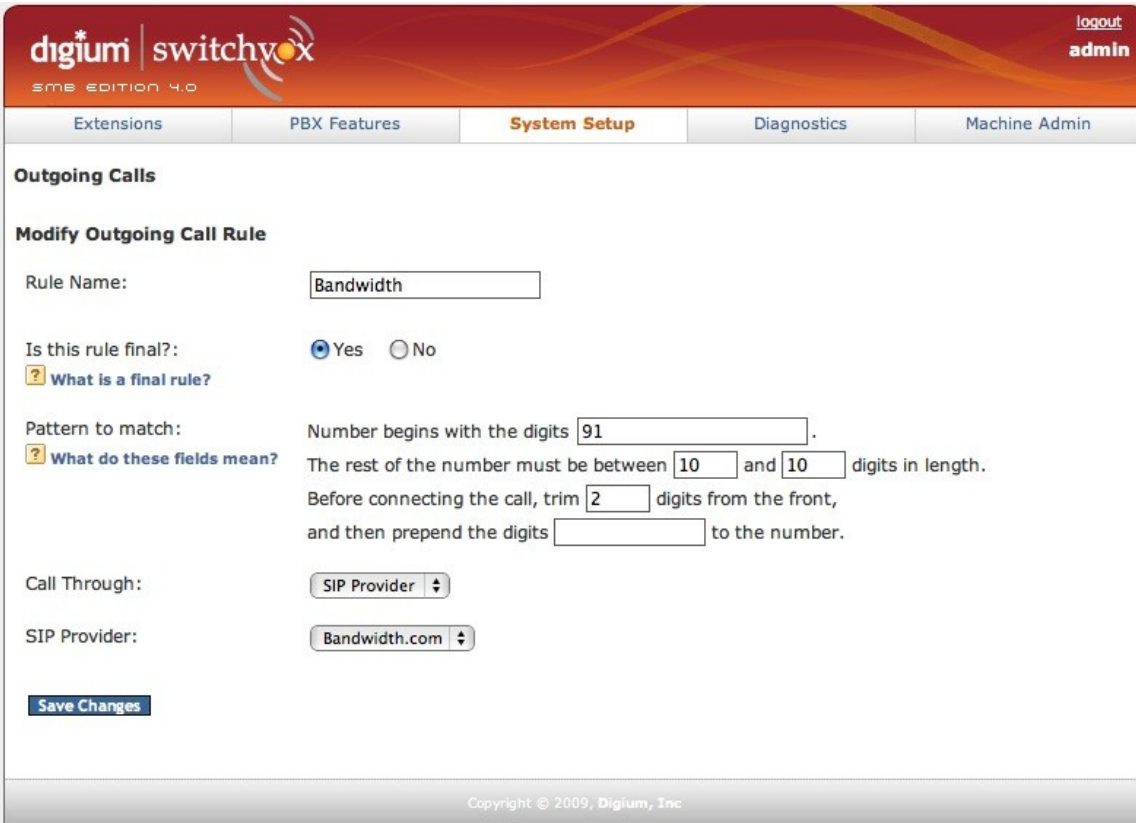
Type	Name	Host	Account ID	Callback Ext.	Latency (ms)	State	Diagnose
SIP	Bandwidth.com	216.82.224.202	[REDACTED]	6191	26	✓ Ok	

- \* The above picture shows Switchvox successfully connected to 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.



**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: Number begins with the digits .  
 ? 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:

[Save Changes](#)

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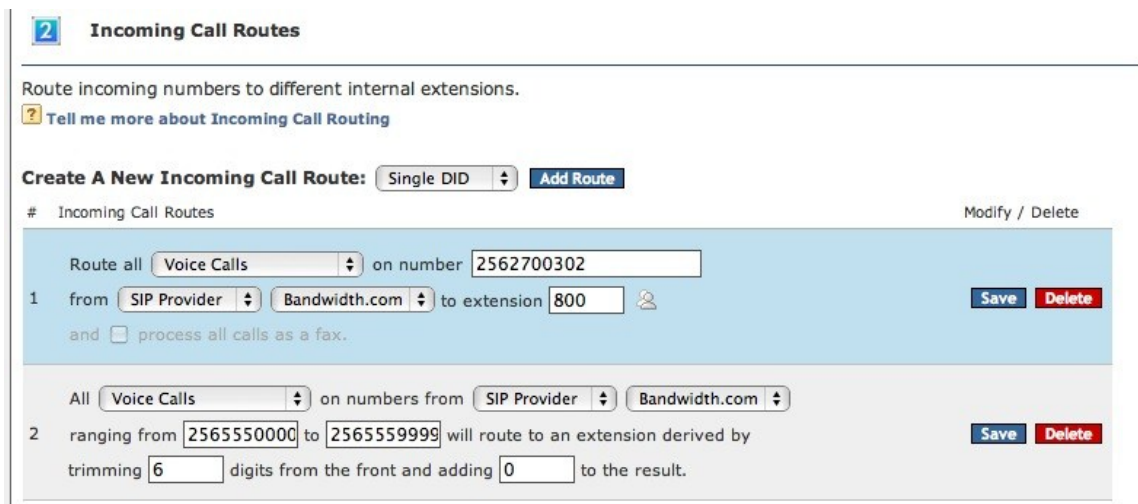
- \* The rule shown in the picture above will take a number beginning with 91 and , truncate the 91 and send the call to Bandwidth.com. You must contact Bandwidth.com installations and let them know that your call digits must be set at 10 digit.



## 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”



**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

Route all  on number

1 from   to extension

and  process all calls as a fax.

All  on numbers from

2 ranging from  to  will route to an extension derived by trimming  digits from the front and adding  to the result.

\* 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”



Allow Nat  Yes  No

Port Forwarding [What does this mean?](#)

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

