Configuration guide for Switchvox and Integra Telecom

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

Hardware and Software: the following hardware and software were employed to test interoperability between digium Switchvox and Integra Telecom.

Manufacturer	Model	Software Vesion
MetaSwitch	MetaSphere	7.3
Adtran	NV3305	17.09.02
digium	Switchvox	

Tested Features: The following is a list of interoperability features that were tested

Feature	Description	Issue (if any)
Basic Call	Making and receiving a call between the IP-PBX and Integra Telecom service provider with both G.711 and G.729 codec.	None
Call Hold	Placing a call in On Hold state and retreival of a call from same station.	None
Call Transfer	Relocation of an active call from one station to another. Both internal and external, attended and unattended transfers were tested.	None
Call Forward	Forwarding of calls from one station to another.	None
3-Party Conference	Conference call between internal and external participants.	None
Fax	Fax Transmission. Fallback to both G.711 and T.38 were tested.	T.38 fax not supported

Configuration Assumptions:

All SIP Signaling uses UDP port 5060
SIP Signaling uses Differentiated Services Code Point (DSCP) 24
Real-Time Transport Protocol (RTP) traffic uses DSCP 46

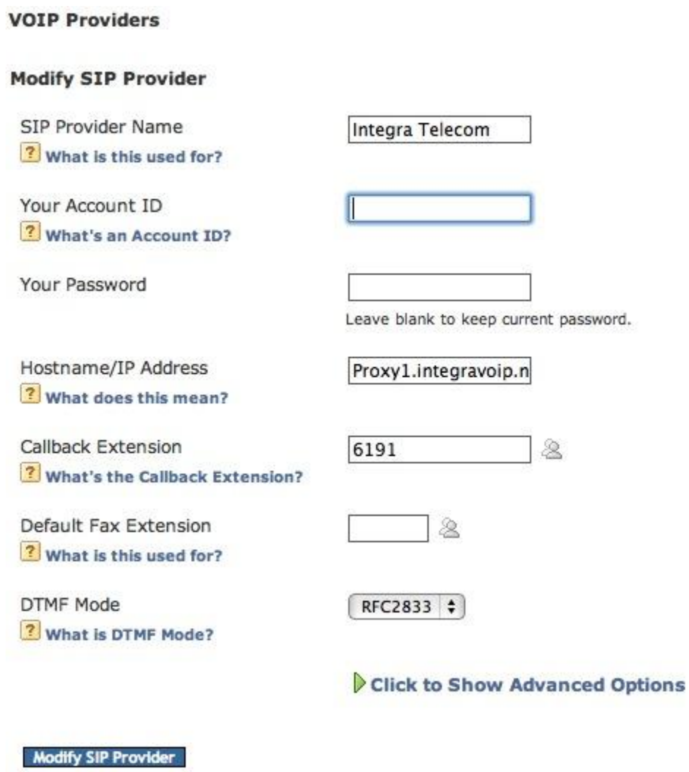
After you have your account information from the Integra Implementation Team, you will need to input this information into your Switchvox system through the admin web interface.

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

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:



* **Your Account ID**: is the username that Integra Telecom provided.

* **SIP Provider Name**: should be something logical that identifies this trunk as Integra Telecom (i.e. “INTEGRA TELECOM”), since you will be using that name later to configure calling rules.

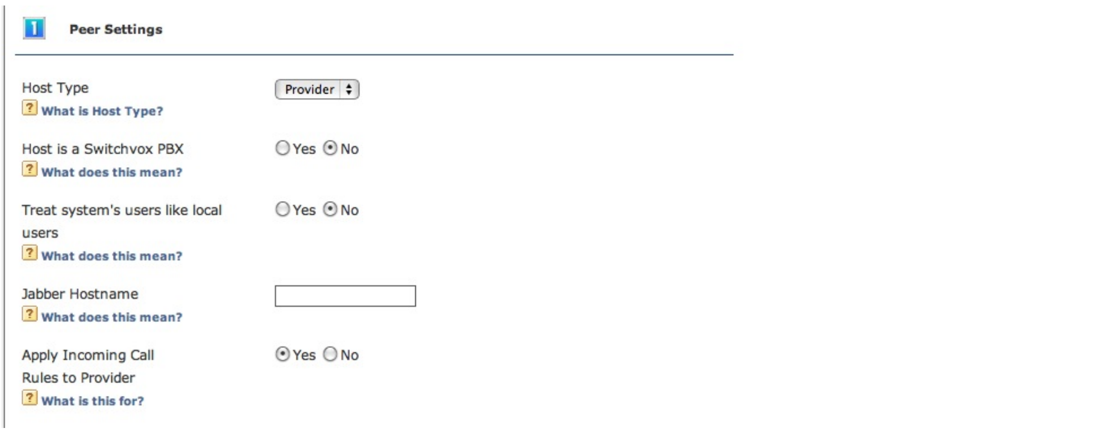
* **Your Password:** the password for digest challenge that Integra Telecom provided.

* **Hostname/IP Address**: The IP address of Integra’s proxies 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 Integra Telecom. This should be set to ‘RFC2833’.

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

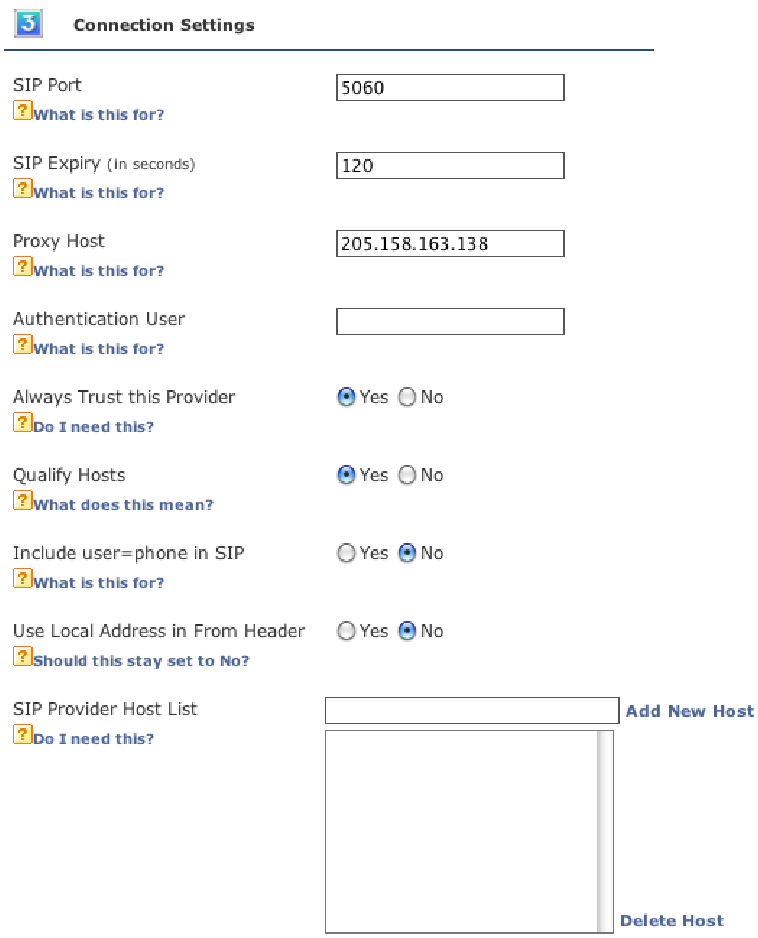


* **Host Type:** Host Type must be set to Provider.

* **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”

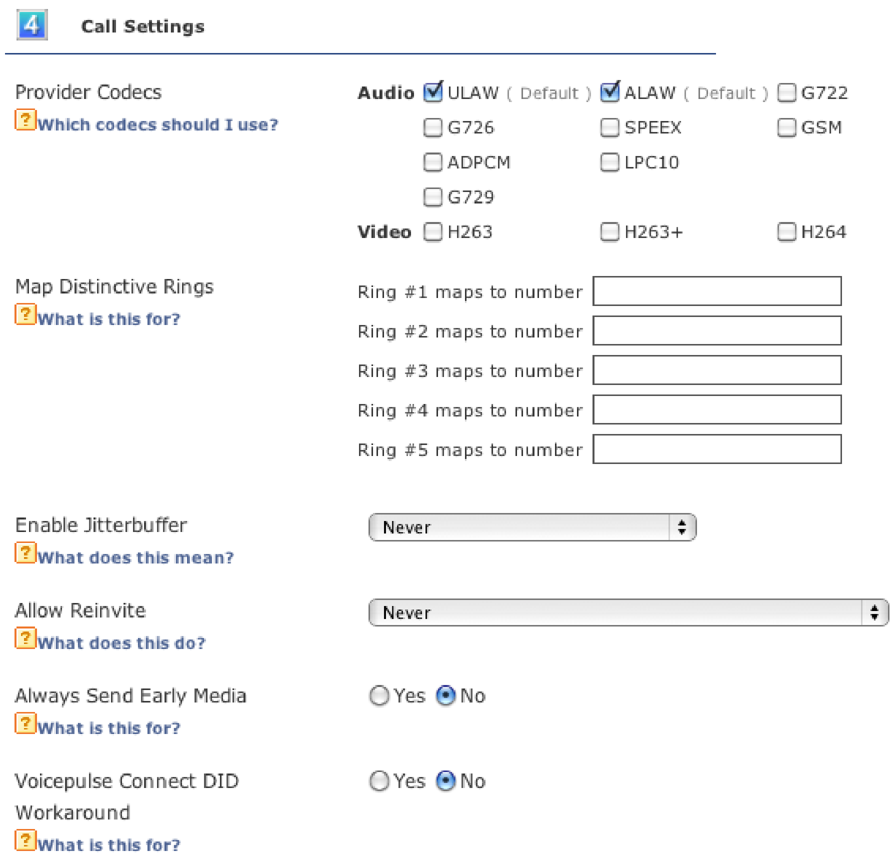


* **Sip Expiry**: The default value of 120.

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

* **Authentication User**: This field is automatically filled in with the Account ID from above.

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



* **Provider Codec's:** Integra Telecom supports G.711 ULAW/ALAW, G729 and G726.

* 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.

System Status

VOIP Providers (1 to 6) of 6

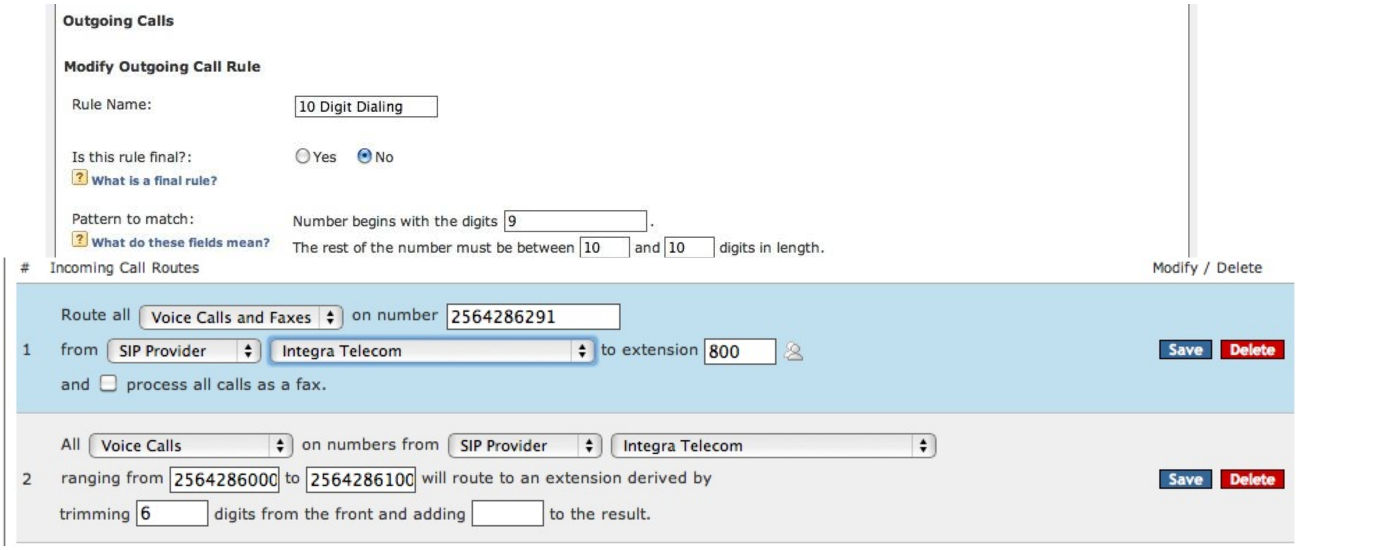
Type	Name	Host	Account ID	Callback Ext.	Latency (ms)	State	Diagnose
SIP	Integra Telecom	Proxy1.integravoip.net	3608529783	6191	78	✓ Registered	

- In the event there is an error connecting to Integra Telecom, 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 INTEGRA TELECOM. 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 9, truncate the 9 and send the call to INTEGRA TELECOM.

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

* Rule number 2 will match a range of DID's 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 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 Integra Telecom SIP Trunking. If you have any questions please contact Digium technical support at +1-256-428-6000 or Integra Telecom technical support at +1-866-871-1114

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