



Configuring Switchvox (Free Edition 1.0) for Spitzfire SIP Trunks

This document is a guideline for configuring Spitzfire SIP trunks onto Switchvox Free Edition and includes the settings required for Inbound DDI routing and Outbound CLI presentation. The settings contained within have been tested and are known to work at the time of testing.

SIP trunk details such as account number and password will be provided separately.

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Provisioning a SIP Trunk

Select “SIP Provider” & click on “Go” next to Add New

The screenshot shows the Digium Switchvox administration interface. At the top left is the logo for Digium Switchvox, with 'FREE EDITION 1.0' below it. At the top right, there are links for 'logout' and 'admin'. Below the header is a navigation menu with tabs for 'Extensions', 'PBX Features', 'System Setup' (which is highlighted), 'Diagnostics', and 'Machine Admin'. The main content area is titled 'VOIP Providers'. Under this title, there is a section '1 Manage VOIP Providers' with a help icon. Below this is an 'Add New:' section with a dropdown menu set to 'SIP Provider' and a 'Go' button. A horizontal line separates this from the 'VOIP Providers' list, which contains the message: 'You have not added any VOIP Providers yet for this system.' Below this is a section '2 RSA Key' with a help icon. It includes a 'Download Current RSA key (pbx41184)' link with a 'Download' button, followed by '-OR-' and a 'Rename RSA key:' label with an input field and a 'Rename' button. At the bottom of the page, there is a copyright notice: 'Copyright © 2007, Digium, Inc'.

Add New SIP Provider

1. **SIP Provider Name:** Enter a logical name for the SIP Trunk
2. **Your Account ID:** Enter the **User name as supplied by Spitfire**. Please note that this is just the number and **DOES NOT** include @spitfiresp.net
3. **Your Password:** Enter the **password as supplied by Spitfire**
4. **Hostname/IP Address:** Enter **83.218.143.16 (or spitfiresp.net)**
5. **Callback Extension:** This is determined by the installer
6. **DTMF Mode:** Leave at RFC2833

Then click on **“Click to Show Advanced Options”**

The screenshot shows the Digium Switchvox System Setup interface. The top navigation bar includes "logout" and "admin" links. The main menu has tabs for "Extensions", "PBX Features", "System Setup" (selected), "Diagnostics", and "Machine Admin". The "VOIP Providers" section is active, showing the "Add a New SIP Provider" form. The form fields are: SIP Provider Name (Spitfire), Your Account ID (442031410020), Your Password (masked with dots), Hostname/IP Address (83.218.143.16), Callback Extension (200), and DTMF Mode (RFC2833). A red oval highlights the "Click to Show Advanced Options" button. An "Add SIP Provider" button is located at the bottom left of the form area. The footer contains the copyright notice: "Copyright © 2007, Digium, Inc".

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logout
admin

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

VOIP Providers

Add a New SIP Provider

SIP Provider Name
 What is this used for?

Your Account ID
 What's an Account ID?

Your Password
 What's the Password?

Hostname/IP Address
 What does this mean?

Callback Extension
 What's the Callback Extension?

DTMF Mode
 What is DTMF Mode?

Click to Show Advanced Options

Add SIP Provider

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Add New SIP Provider continued

- 7. **Host Type:** Leave the Host Type as Provider
- 8. **Supports Changing Caller ID:** Leave at No
- 9. **Caller ID Name:** N/A

[logout](#)
admin

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


VOIP Providers


Add a New SIP Provider


| | |
|---------------------------------------|---|
| SIP Provider Name | Spitfire |
| What is this used for? | |
| Your Account ID | 442031410020 |
| What's an Account ID? | |
| Your Password | ●●●●●●●● |
| Hostname/IP Address | 83.218.143.16 |
| What does this mean? | |
| Callback Extension | 200 |
| What's the Callback Extension? | |
| DTMF Mode | RFC2833 |
| What is DTMF Mode? | |
| Click to Hide Advanced Options | |
| Host Type | Provider |
| What is Host Type? | |
| Supports Changing Caller ID | <input type="radio"/> Yes <input checked="" type="radio"/> No |
| Why should I not change this? | |
| Caller ID Name | |


Add New SIP Provider continued


- 10. **Caller ID Number:** N/A
- 11. **SIP Port:** 5060
- 12. **Expiry Timer:** Can be a maximum of 3600 – **Change to a minimum of 600**
- 13. **Proxy Host:** Change the Proxy Host to **mproxy3.spitfiretsp.net**
- 14. **Authentication User:** This is copied from the Account ID. Leave as is
- 15. **Always Trust:** Leave at Yes
- 16. **Send Early Media:** Change to **Yes**
- 17. **Incoming Call Rules:** Leave at Yes
- 18. **Outgoing Call Rules:** Determined by the installer


Caller ID Number
 **What is this?**


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?**

Always Send Early Media Yes No
 **What is this for?**

Apply Incoming Call Rules to Provider Yes No
 **What is this for?**

Outgoing Call Rules  **What is this for?**

| Rule Name | Allow | Deny |
|---------------|--------------------------|--------------------------|
| 1-900 Numbers | <input type="checkbox"/> | <input type="checkbox"/> |
| Toll Free | <input type="checkbox"/> | <input type="checkbox"/> |
| Long Distance | <input type="checkbox"/> | <input type="checkbox"/> |
| Local | <input type="checkbox"/> | <input type="checkbox"/> |
| 911 | <input type="checkbox"/> | <input type="checkbox"/> |

Add New SIP Provider continued

- 19. **SIP Provider Host List:** Leave empty
- 20. **Map Distinctive Rings:** Leave empty
- 21. **Provider Codecs:** Leave as default (uLaw & aLaw)
- 22. **Qualify Hosts:** No
- 23. **Voicepulse Workaround** No
- 24. THEN click on **“Add SIP Provider”**

If registration is not successful, check your settings & reload the software

SIP Provider Host List

Do I need this?

[Add New Host](#)

[Delete Host](#)

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

Provider Codecs

Which codecs should I use?

ULAW(Default) ALAW(Default) GSM

G726 ADPCM LPC10

SPEEX ILBC

Qualify Hosts

What does this mean?

Yes No

Voicepulse Connect DID Workaround

What is this for?

Yes No

Add SIP Provider

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Inbound Routing – Reference only

Select “**New Incoming Call Route**”

The screenshot shows the Digium Switchvox administration interface. At the top, there is a navigation bar with the following tabs: Extensions, PBX Features, System Setup (highlighted), Diagnostics, and Machine Admin. The top right corner contains links for 'logout' and 'admin'. Below the navigation bar, the 'Incoming Calls' section displays a green message: 'Successfully deleted incoming rule.' Below this is a section titled '1 Incoming Call Rules' with a sub-header 'Incoming Call Rules'. The text below reads: 'Create, modify, prioritize and delete incoming call rules to apply to the extensions on your PBX system.' There is a help icon and a link: '? Tell me more about Incoming Call Rules'. Below this is a form to 'Create A New Incoming Call Rule:' with a dropdown menu set to 'Block Number' and an 'Add Rule' button. The next section is '2 Incoming Call Routes' with a sub-header 'Incoming Call Routes'. The text below reads: 'Route incoming numbers to different internal extensions.' There is a help icon and a link: '? Tell me more about Incoming Call Routing'. Below this is a form to 'Create A New Incoming Call Route:' with an 'Add Route' button. The 'Add Route' button is circled in red. Below the form is a table of existing routes. The table has a header row with '# Incoming Call Routes' and 'Modify / Delete'. The first row is highlighted and contains: '1 Route All unmatched numbers from SIP Provider "Spitfire" to extension 200. Click here to change this SIP Provider's default extension.' The second row contains: '2 Route all VOIP calls from any unknown host to Busy Signal' with a dropdown menu set to 'Busy Signal' and a 'Save' button.

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

Incoming Calls

✓ Successfully deleted incoming rule.

1 Incoming Call Rules

Create, modify, prioritize and delete incoming call rules to apply to the extensions on your PBX system.

? Tell me more about Incoming Call Rules

Create A New Incoming Call Rule: Block Number Add Rule

2 Incoming Call Routes

Route incoming numbers to different internal extensions.

? Tell me more about Incoming Call Routing

Create A New Incoming Call Route: Add Route

| # Incoming Call Routes | Modify / Delete |
|--|-----------------|
| 1 | |
| Route All unmatched numbers from SIP Provider "Spitfire" to extension 200 Click here to change this SIP Provider's default extension. | |
| 2 | |
| Route all VOIP calls from any unknown host to Busy Signal | Save |

Inbound Routing continued

1. **Route Number:** Enter the Route Number in the format **44xxxxxxxxxx** (as below)
2. **From:** Change to **SIP Provider**, and then **select** the name you have given to the trunk (Spitfire in this case)
3. **To Extension:** Enter the extension number you wish to route the call to and click **Save**.

The screenshot shows the Digium Switchvox System Setup interface. The top navigation bar includes "Extensions", "PBX Features", "System Setup" (highlighted), "Diagnostics", and "Machine Admin". The "logout" and "admin" links are in the top right corner. The "Incoming Calls" section shows a green checkmark and the message "Successfully added new rule." Below this is a section titled "1 Incoming Call Rules" with a description: "Create, modify, prioritize and delete incoming call rules to apply to the extensions on your PBX system." A link "Tell me more about Incoming Call Rules" is provided. The "Create A New Incoming Call Rule:" section has a "Block Number" dropdown and an "Add Rule" button. The "2 Incoming Call Routes" section has a description: "Route incoming numbers to different internal extensions." and a link "Tell me more about Incoming Call Routing". The "Create A New Incoming Call Route:" section has an "Add Route" button. Below this is a table of Incoming Call Routes with columns for "#", "Route number", "from", "to extension", and "Modify / Delete". The first row is highlighted in blue and circled in red. It shows route number 1, route number 442031410021, from SIP Provider (dropdown), Spitfire (dropdown), to extension 200, with Save and Delete buttons. Below the table, there are two more routes: "Route All unmatched numbers from SIP Provider 'Spitfire' to extension 200" and "Route all VOIP calls from any unknown host to Busy Signal" with a Save button.

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

Incoming Calls

✓ Successfully added new rule.

1 Incoming Call Rules

Create, modify, prioritize and delete incoming call rules to apply to the extensions on your PBX system.
[? Tell me more about Incoming Call Rules](#)

Create A New Incoming Call Rule:

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 number <input type="text" value="442031410021"/> from <input type="text" value="SIP Provider"/> <input type="text" value="Spitfire"/> to extension <input type="text" value="200"/> | <input type="button" value="Save"/> <input type="button" value="Delete"/> |
| 2 | Route All unmatched numbers from SIP Provider "Spitfire" to extension 200 Click here to change this SIP Provider's default extension. | |
| 3 | Route all VOIP calls from any unknown host to <input type="text" value="Busy Signal"/> | <input type="button" value="Save"/> |

Outgoing CLI – Reference only

Note: Individual CLI presentation is not available on the Switchvox free edition 1.0.

Supports changing Caller ID on the VoIP trunk is left at “No” because you cannot select RPID as a Caller ID Method on the Free Edition. Selecting “No” ignores any Caller ID Rule assigned on a per extension basis and simply presents the main number.

If you select “Yes”, and then apply individual Caller ID Rules on a per extension basis the SIP messaging will be incorrect as Spitfire expects to see it and the call will fail.