



# **Digium Switchvox**

## IP PBX Provisioning Guide

Version 1.3  
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### DISCLAIMER

This document is provided as a basic guideline for setup and configuration of Digium Switchvox systems with MegaPath's SIP Trunking service, based on MegaPath's testing and validation process. It does not include advanced configurations to enable features such as voicemail, Find-Me-Follow-Me, etc. MegaPath is not responsible for customer IP PBX configurations. For more information on advanced features, please refer to your IP PBX documentation.

### OVERVIEW

Digium Switchvox ([www.switchvox.com](http://www.switchvox.com)) is a customizable telephony solution based on Asterisk open source software and providing PBX functionality such as call switching, route management, and advanced features including voicemail, conferencing, IVR, paging, and more. Both hardware-based appliances and software downloads are available. The steps outlined in this guide reflect the process used by MegaPath to successfully configure MegaPath's SIP trunks on the described software/hardware versions only.

### SUPPORTED SOFTWARE DEPLOYMENTS

The following software deployments are supported with this setup guide:

- Digium Switchvox 8634

### COMPATIBLE HARDWARE DEPLOYMENTS

The following hardware deployments are supported with this setup guide:

- Digium Switchvox SMB
- Digium Switchvox SOHO

### REQUIRED INFORMATION

The following information is required to complete the configuration. This information will be provided by your MegaPath Business Voice Implementation Engineer during the installation process.

Host \_\_\_\_\_

Username \_\_\_\_\_

Password \_\_\_\_\_

### CONFIGURATION

Navigate and Login to the admin section of the web portal <https://<IP ADDRESS>/admin/>

1. Click "System Setup", and then click "VOIP Providers"
2. Select Add new SIP Provider, and click the Go button and fill out the following Fields:

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- Set "SIP Provider Name" to MEGAPATH (Any Name will do here it is just a name)
  - Set "Your Account ID" To the trunk group username
  - Set "Your Password" To The Trunk group password
  - Set "Hostname/IP Address" to the Host name assigned to account (SIP Trunk Host provided by Moses)
    - If using an Edgemarc, enter the Edgemarc's LAN IP address
  - Select RFC2833 for "DTMF Mode"
3. Click the link titled "Click to show advanced options"
- Set "Supports Changing Caller ID" to Yes
  - Set "Proxy Host" to provided hostname
    - If using an Edgemarc, specify the Edgemarc's LAN IP address
  - Set "Authentication User" to provided username
  - Select the appropriate codec -- NOT SHOWN -> at bottom of page

### **ROUTING – OUTGOING CALLS**

Go to "**System Setup**" then to "**Outgoing Calls**"

All existing routes need to be modified by clicking the Modify button next to the route so that they send calls to the MEGAPATH VoIP provider defined in the previous section, and to send calls out with the appropriate E.164 formatting (+1NPANXXXXXX, or +CCXXXXXXX for International calls)

1. Add a "+1NPA" in the "prepend the digits" field
2. Change "Call Through" to SIP Provider
3. Change "SIP Provider" to MEGAPATH (or whatever name used in previous section).
4. This process should be repeated for all outgoing dialing patterns that you wish to send to MegaPath.

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**Note –** BY DEFAULT THERE IS NO OUTBOUND 911 ROUTE, PLEASE BE SURE TO SET ONE UP

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### **ROUTING – OUTBOUND CALLER-ID**

Go to "**System Setup**", "**Outgoing Calls**", then to "**Outgoing Caller ID Rules**".

1. Specify Extension number, and specify a Caller ID Number for this extension.

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**Note –** Each extension should have a Caller ID number specified in this manner

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### **ROUTING – INCOMING CALLS**

Go to "**System Setup**" then to "**Incoming Calls**"

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Every DID needs either an Incoming Call Rule (IE transfer to another number) or An Incoming Call Route (IE Route to an extension). Most cases will use an Incoming call route to send a call to an existing extension.

1. Click the "Add Route" Button:
2. Enter DID in the "Route Number" field
3. Enter the extension you want the DID routed to in the "Extension" field
4. Click the Save button

### **BLIND TRANSFERS**

Blind Transfer's do not work with Asterisks based systems. This issue can be resolved by setting CLID override on the trunk group.

### **BROADWORKS CPE TYPE**

Digium Switchvox was tested using **Generic SIP PBX Single Registration** as the CPE device type within Broadworks.

### **TESTED SOFTWARE VERSIONS**

This was tested with Digium Switchvox free addition version 8634.

### **OUTBOUND CALLER ID**

Recommended Outbound Caller ID settings are to be configured using the Outgoing Caller ID Rules. This will allow you to configure Outbound Caller ID on a per user (extension) basis. Switchvox does support Global Caller ID settings which is not recommended as remote users may need to have a different 911 address displayed for 911 calls.