

SWITCHVOX  
Technical Application Notes  
JULY 2016



## Document Revision History

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## ABOUT SWITCHVOX

Switchvox is more than an IP PBX, it's Digium's proven family of Business Phone Systems for small and medium businesses. Switchvox systems are designed to be easy to use, and easy to manage while bringing affordable Unified Communications solutions to SMBs.

Digium®, Inc., the Asterisk® Company, created, owns and is the innovative force behind Asterisk, the most widely used open source telephony software. Since its founding in 1999, Digium has become the open source alternative to proprietary communication providers, with offerings that cost as much as 80 percent less. Digium offers Asterisk software free to the open source community and offers [Switchvox IP PBX Software](#) to power a broad family of products for small, medium and large businesses. The company's product line includes a wide range of hardware to enable resellers and customers to implement [turnkey solutions](#) or to design their own voice over IP (VoIP) systems. More information is available at [www.digium.com](http://www.digium.com).

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## PURPOSE, SCOPE AND AUDIENCE

This technical application note describes the configuration of Switchvox for Fusion SIP Trunking service. This document is suitable for use by anyone deploying Fusion SIP Trunking service in conjunction with Switchvox. This document has a technical audience in mind – specifically IT professionals skilled in Linux with some experience in PBX administration and familiarity with VoIP technologies. This document is not for business administrators or people in other non-technical careers. In order to successfully use this document to deploy Fusion SIP Trunking service, you will need to possess the following skills, or have access to professionals or consultants with the following skills:

- Understanding of UNIX or Linux operating systems, including:
  - Understanding of file and directory structure on target OS
  - Understanding of firewall configuration on target OS
  - Understanding of network configuration on target OS
  - Understanding of service configuration on target OS
- Familiarity with network troubleshooting tools, including:
  - Wireshark/Ethereal
  - dig/nslookup
  - ping
  - traceroute
- Familiarity with PBX systems, including:
  - Trunk configuration
  - Calling plan configuration
  - Extension configuration
  - Mailbox configuration
- Familiarity with Session Initiation Protocol (SIP)
- An understanding of all seven layers of the Open System Interconnection (OSI) model
- A complete understanding of your internal network structure, Network Address Translation (NAT) setup, and firewall setup
- A complete understanding of your public Internet connectivity

Fusion can only provide support for Switchvox to the extent covered in this Technical Application Note and the included reference configuration, so if your level of technical expertise does not include the above skills, it is recommended that you obtain the services of a Switchvox professional.

## SWITCHVOX DEPLOYMENT INFORMATION

Before you begin deploying Switchvox, please locate the following information. If you have questions about any item, refer to the descriptions and additional details provided on the pages that follow.

Switchvox External IP Address or DNS: \_\_\_\_\_

<b>Preferred Codec:</b>	μlaw	g711	
<b>Alternate Codec:</b>	μlaw	g711	none
<b>Is Switchvox NATd:</b>	Yes	No	
<b>Is Switchvox Behind a Firewall:</b>	Yes	No	
<b>Is There a Local Firewall:</b>	Yes	No	

### Switchvox External IP Address

Your Switchvox server will either use a public IP address or a private IP address. If the IP address on your Switchvox server is of the form 192.168.x.x, 172.16.x.x – 172.31.x.x, or 10.x.x.x, then your Switchvox server uses an internal, private IP address. This internal address is not routable on the public Internet. In order for your Switchvox server to connect to the Fusion Session Border Controller, you must either have a public IP address on your Switchvox server or you must translate your private IP address into a public IP address using a Network Address Translator (NAT).

If your Switchvox server is behind a NAT, your public IP address will typically be the public IP address of your NAT. You may also have a static, one-to-one mapping of a public IP address to your private IP address. In this case, your public IP address will not match the IP address of your NAT, but you can look up the correct public IP address in your NAT configuration. If in doubt, you can perform a network packet capture using Wireshark (previously called Ethereal) on the public side of your NAT while simultaneously issuing some form of Internet request on your Switchvox server.

## Sending Calls to Fusion

In your Welcome letter, Fusion provides DNS records to which you may send calls, and from which you should be prepared to receive calls. Switchvox supports DNS A records, DNS SRV records, and IP addresses. Please note, however, that Switchvox only resolves the DNS records during startup and will only utilize the first IP address in any multiple-IP record.

At the top of your Welcome Letter you should see the following:

**Account Name:**    Name  
**Account Number:** 99XXXXXXXXX  
**Order Number:**    XXXXXX-X

In your Welcome letter, Fusion Connect provides IP addresses to which you may send calls and from which you will should be prepared to receive calls. FreePBX support both DNS A records and IP addresses. Both should be listed in your Welcome Letter and either can be used. This IP will be dependent on if the trunk is configured as Static or Registration. For deployment of the Switchvox we recommended that the trunks be set as registration.

Traffic Type	IP Address	Domain Name	Protocol	Port Range
SIP Registration	216.86.41.69	sip2.thevoicemanager.com	UDP and TCP	5060/5061
SIP Static	216.86.41.68	peer2.thevoicemanager.com	UDP and TCP	5060
RTP – Registration	216.86.41.69	N/A	UDP and TCP	10000 to 65535
RTP – Static	216.86.41.68	N/A	UDP and TCP	10000 to 65535

### Preferred and Alternate Codecs

Fusion allows you to select preferred and alternate codecs to simultaneously meet your bandwidth requirements and provide greater end-to-end support. In the event that your destination party or your destination party's carrier cannot support your preferred codec or alternate codecs, Fusion will automatically transcode your call to a supported codec.

When configuring codecs, please keep in mind that G.711  $\mu$ Law (ulaw) consumes approximately 87.2 Kbps of bandwidth per simultaneous call over Ethernet. G.729 Annex A (g729) will consume approximately 31.2 Kbps of bandwidth per simultaneous call over Ethernet. Also, keep in mind that G.711 offers superior call quality when compared to G.729, but only if you have enough bandwidth to support all of your simultaneous calls.

### Preferred and Alternate Codecs

If your Switchvox server uses an Internet-facing IP address of the form 192.168.x.x, 172.16.x.x – 172.31.x.x, or 10.x.x.x, then it is almost certainly behind a Network Address Translation (NAT) device. If your server uses an address of that form and is not behind a NAT, then it has no connectivity to the Internet. Even if your server uses an IP address that does not match the forms above, it is still possible (though very unlikely) that it is behind a NAT. If Switchvox is behind a NAT, you may need to perform port forwarding, set up a DMZ host or configure a one-to-one static IP map.

### Is Switchvox Behind a Firewall

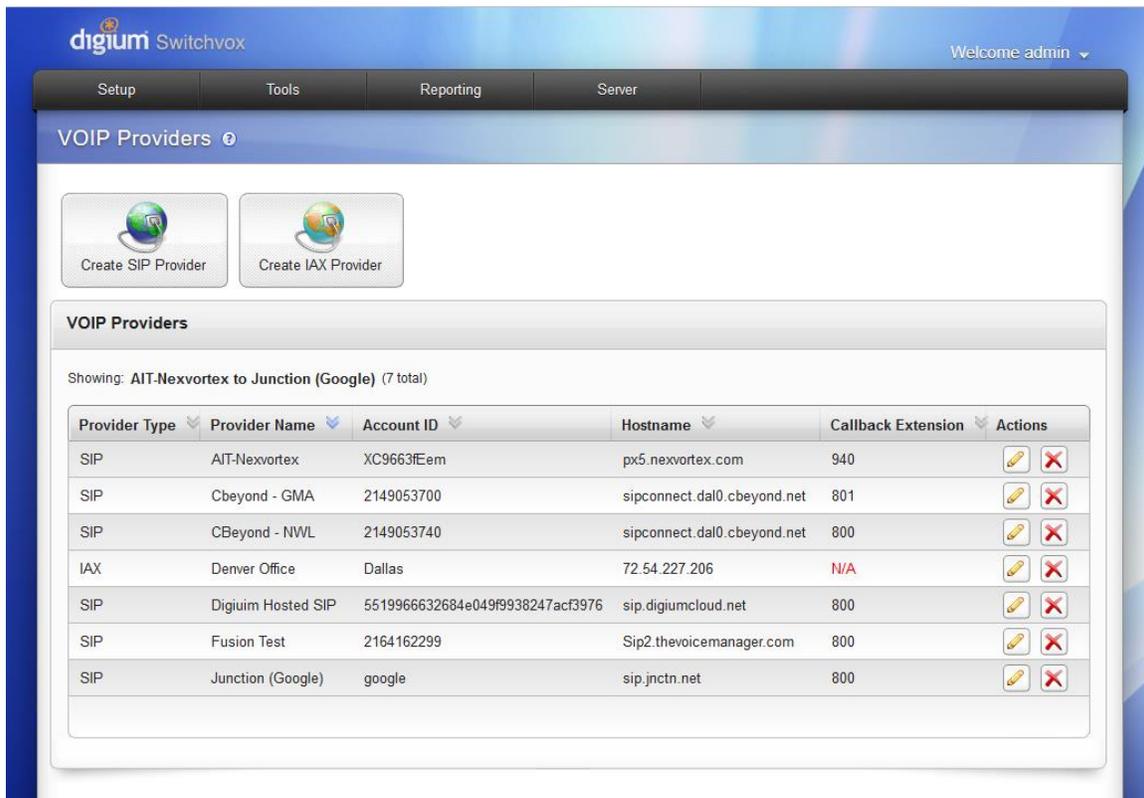
If Switchvox is behind a NAT, then it is almost certainly behind a firewall. It is also possible that Switchvox uses a public IP address but is still behind a firewall. If you use a Cisco PIX, SonicWALL, Shorewall, Firebox, or any other brand of firewall, you may need to perform additional configuration steps on the firewall device to allow Switchvox to function properly. Additionally, you may be using an Application Gateway such as an Ingate SIParator. These devices will also need additional configuration to allow Switchvox to function properly. Configuring your firewall or application gateway is beyond the scope of this document. In general, you will need to allow UDP port 5060 in both directions, as well as UDP ports 1024 to 65535 for RTP. However, you may need a larger range of ports for RTP.

### CONFIGURING SWITCHVOX

The following entails configuring both outbound and inbound routes for calls to and from Fusion. Inbound routes control how DIDs provided by Fusion are routed to your IVR and/or phones. Outbound routes determine how calls from your phones are routed out toward Fusion.

Once you have logged into the Switchvox, select the System Setup from the list of options. This is where most of the PBX configuration is going to be done. The System Setup screen can be seen below.

First, go to the "VOIP Providers" page, seen below. On this page you will be setting up the trunk that will communicate with Fusion.



Switchvox VOIP Providers Screen

Begin by selecting SIP Provider from the drop down menu under Manage VOIP Providers and click “Go”. Fill in the information as listed below.

**NOTE:** Actual IP/Hostname information may vary, so please consult your technical letter for the most accurate information.

VoIP Provider Information Screen

<b>SIP Provider Name:</b>	Fusion
<b>Your Account ID:</b>	BTN
<b>Password:</b>	Provided in technical letter
<b>Hostname/IP Address:</b>	Sip2.thevoicemanager.com
<b>Callback extension:</b>	customer choice (800 is default IVR)
<b>Default Fax Extension:</b>	
<b>DTMF Mode:</b>	RFC2833

Click on the “Connection Settings” tab which will allow you to make some additional configuration selections.

The screenshot shows the 'Create SIP Provider' configuration page with the 'Connection Settings' tab active. The configuration fields are as follows:

- SIP Port: 5060
- SIP Expiry (in seconds): 120
- Proxy Host: [Empty]
- Authentication User: [Empty]
- Always Trust this Provider: YES (checked)
- Qualify Hosts: NO
- Include user=phone in SIP: NO
- Use Local Address in From Header: NO

Below these settings is a 'SIP Provider Host List' section with a search bar and a 'Host' dropdown menu. A 'Save SIP Provider' button is located at the bottom of the form.

Switchvox Connections Settings Screen

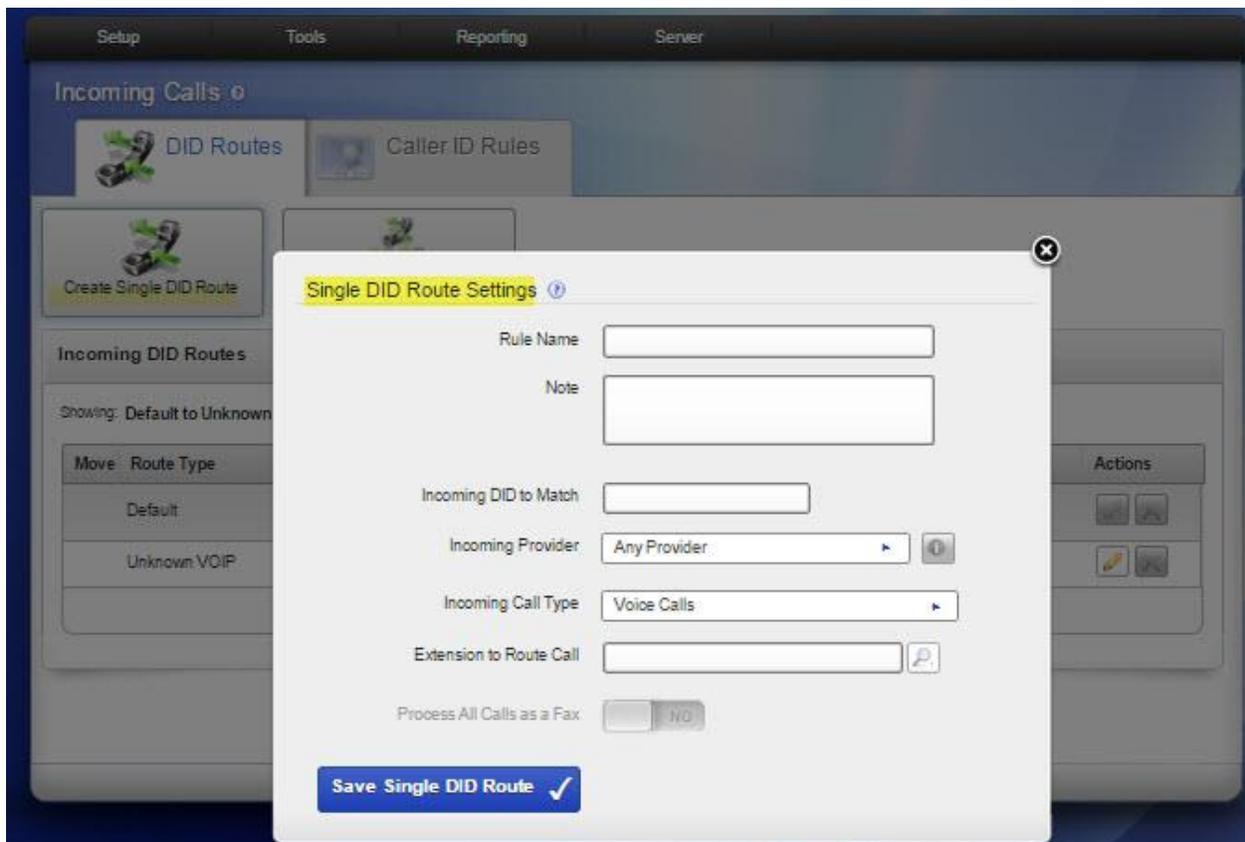
<b>SIP Port:</b>	5060
<b>SIP Expiry:</b>	300
<b>Proxy Host:</b>	sip2.thevoicemanager.com
<b>Authentication User:</b>	BTN
<b>Always Trust this Provider:</b>	Yes
<b>Qualify Hosts:</b>	No
<b>Include user=phone in SIP:</b>	No
<b>Use Local Address in From Header:</b>	Yes
<b>SIP Provider Host List:</b>	sip2.thevoicemanager.com

Currently, Fusion only supports the ULAW audio codec from the selections in section 4; "Call Settings". It is recommended that you unselect ALAW from the default list.

At the bottom of the page, save the settings by clicking on the **Save SIP Provider** button. This will take you back to the "VOIP Providers" screen.

Fusion utilizes 10-digits in all routing. All digits including the area code must be present in any outgoing calls to Fusion for the call to complete properly.

You will now want to create inbound routes to route DIDs to extensions. Creating inbound routes will mostly depend on how you would like to set up DIDs to route. This is another instance where this should be curtailed to meet your specific business needs.



**NOTE:** There is a 30 second default interval before Switchvox will reroute a call that it doesn't get a response to. This can cause some extended wait time or "post-dial-delay" if the first host is down for some reason. This is extremely rare. Generally, when this happens you will see the host become unregistered. To verify this, choose Diagnostics > System Status.