

# SIP Trunking using the Optimum Business SIP Trunk Adaptor and the Digium IP-PBX

## Table of Contents

1. Overview	3
2. SIP Trunk Adaptor Set-up Instructions	3
3. Additional Set-up Information	7
4. International Calling	8
5. Digium PBX Configuration	9
5.1 SIP Trunking	
5.2 Extensions/DID	
5.3 Dial Plan	
5.4 Backup/Restore	

## 1 Overview

The purpose of this configuration guide is to describe the steps needed to configure the Digium IP PBX for proper operation Optimum Business Sip Trunking.

## 2 SIP Trunk Adaptor Set-up Instructions

These instructions describe the steps needed to configure the LAN side of the Optimum Business SIP Trunk Adaptor.

### Step 1:

#### Log on to the Optimum Business SIP Trunk Adaptor

1. Connect a PC to port 4 of the Optimum Business SIP Trunk Adaptor, the silver device labeled Edgewater Networks, 4550 series.



2. Open a Web browser and go to IP Address <http://10.10.200.1>. A login box will appear.

3. Enter login and password and click 'OK'.

Login: pbxinstall

Password: slptrunk



### Step 2:

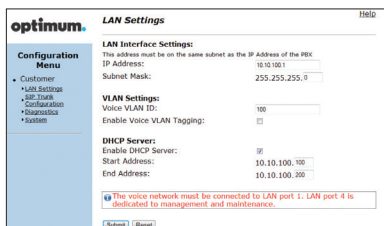
#### Click on the LAN Settings Link

1. Assign an IP Address to the LAN interface of the SIP Trunk Adaptor. The IP address must be on the same subnet as the IP PBX. This changes the address on port 1 of the Optimum Business SIP Trunk Adaptor.

Note: This will become your local SIP proxy IP address. No other IP addresses will be provided by Cablevision.

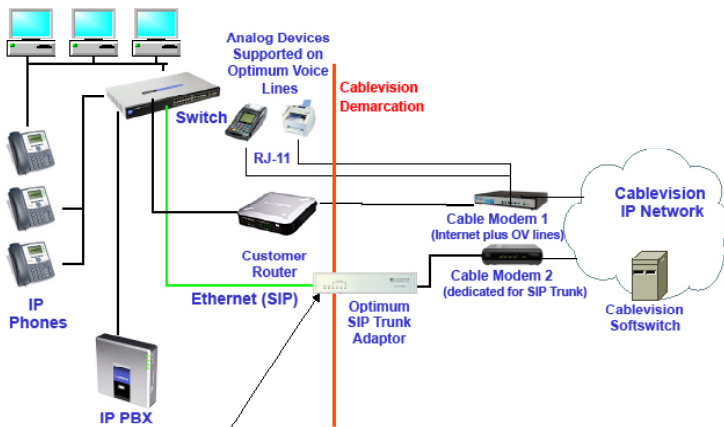
2. Optional: Specify a VLAN for your voice traffic. Click the 'Enable Voice VLAN Tagging' check box. The default VLAN ID is 100.

Note: VLAN 200 should not be used. It is dedicated to port 4 for management.



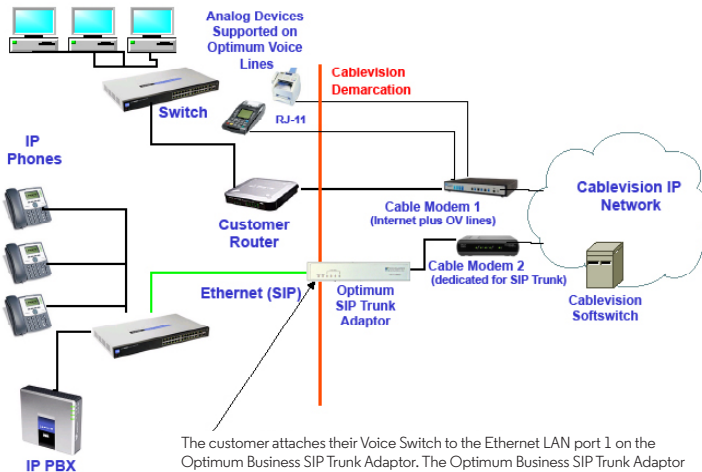
3. Optional: Enable the DHCP server. This will allow the SIP Trunk Adaptor to act as a DHCP server, which will provide IP addresses to the voice network, and create a dedicated voice LAN, as per diagram 2.
4. Click 'Submit'.

**Diagram 1 SIP Trunk Adaptor for IP-PBXs**  
**Example: Single LAN Configuration**



Using a connection from the customer's LAN, the SIP Trunk Adaptor's address can be a statically assigned private IP address. It may not be assigned a Public IP address.

**Diagram 2 SIP Trunk Adaptor for IP-PBXs**  
**Example: Separate Voice and Data Networks Configuration**



The customer attaches their Voice Switch to the Ethernet LAN port 1 on the Optimum Business SIP Trunk Adaptor. The Optimum Business SIP Trunk Adaptor can be enabled as a DHCP server to provide routing for the separate voice network.

## Step 3:

### Click on the SIP Trunk Configuration Link

1. Select your IP PBX make and model from the drop-down menu.
2. Specify how the IP PBX will register to the Optimum Business SIP Trunk Adaptor.
3. The Cablevision network only supports Inband DTMF. Click on the check box next to "Convert Inband DTMF" if you cannot configure your IP PBX to send out Inband DTMF. The DTMF tone duration generated by the phones and/or PBX may need to be increased from their default setting. Some phones and/or PBX have a default setting between 180ms to 200ms. This setting is too low. The recommended setting is 600ms.
4. Click 'Submit'.

The screenshot shows the 'SIP Trunk Configuration' page. On the left is a 'Configuration Menu' with links: Customer, LAN Settings, SIP Trunk Configuration (selected), Diagnostics, and System. The main content area has a 'Select your PBX:' dropdown menu with 'Asterisk' selected. Below this are two radio button options: 'Passive connection using the local, private IP address of the PBX interface' (selected) and 'Active connection using registration'. The 'Active connection' section includes fields for 'User Id:' (set to 'secret') and 'Password:' (masked with asterisks). There is a checkbox for 'Convert Inband DTMF:' which is currently unchecked. At the bottom, there is a 'Status:' section showing 'Trunk Status:' as 'Not Registered' and a list of 'DID's' (0164030809, 0164030760, 0164030769, 0164030765, 0164030841). 'Submit' and 'Reset' buttons are located below the password field.

## Step 4:

### Diagnostics Link

You can make a test call directly from your phone or use the test call application under the Diagnostics link.

The screenshot shows the 'Network Test Tools' page. On the left is the same 'Configuration Menu' as in Step 3. The main content area has a heading 'Network Test Tools' and a brief description: 'A network administrator may use the test tools on this page to verify connectivity of the System and trace the path of data throughout the network.' Below this are three test sections: 'Outbound Call Test:' (with a 'Pilot Number:' field set to '0164030809' and 'Call'/'Reset' buttons), 'Inbound Call Test:' (with a 'Disabled' radio button selected and a 'Submit' button), and 'Ping Test:' (with an 'IP Address to Ping:' field and 'Ping'/'Reset' buttons). At the bottom is the 'Traceroute Test:' (with an 'IP Address to Trace:' field and 'Traceroute'/'Reset' buttons).

## Step 4 continued

Field	Description
Outbound Call Test TelephoneNumber	Specifies an outside phone number to which an outbound call will be initiated. The pilot telephone number of the SIP Trunk will be prepopulated.
Pilot Number	Displays the provisioned pilot number, which is used for outbound and inbound call tests.
Call	Initiates a call outbound to a telephone number entered or inbound to the pilot number displayed.
Inbound Call Test (radio button)	Indicates whether inbound test call will be enabled or disabled. If inbound test calls are enabled, calls made to the pilot number will be redirected to the test UA for fifteen minutes. When the pilot number is dialed, you will hear a test message play.
Submit	Enables or disables the inbound call test.
IP Address to Ping	Verifies basic connectivity to a networking device. Successful ping test results indicate that both physical and virtual path connections exist between the system and the test IP address.
Ping Button	Sends a ping to the IP address specified in the field "IP Address to Ping".
IP Address to Trace	Tracks the progress of a packet through the network. The packet can be tracked through the WAN or LAN interfaces of the adaptor.
Interface (radio button)	Indicates whether a packet will be tracked through the LAN or the WAN.
Traceroute Button	Initiates a traceroute to the specified IP address on either the LAN or the WAN.
Reset	Clears all fields and selections and allows you to enter new information. Reset applies to outbound call test, ping and traceroute.

# 3 Additional Set-up Information Systems

**optimum.**

Configuration Menu

- Customer
  - LAN Settings
  - SIP Trunk Configuration
  - Diagnostics
  - System

System

Help

Software Version:  
Version 11.6.14.1 -- Fri Jan 4 17:49:28 PST 2013

Hostname:  
5164939899

Model:  
EdgeMarc 4552

Vendor:  
Cablevision

LAN Interface MAC Address:  
A8:70:A5:00:D8:18

Registration Status:  
The ALG feature is registered. View [license key](#).

System Date:  
02/29/2016 15:03:40 UTC

Change Password:

- [pbxinstall](#)

Field	Description
Pbxinstall Link	Select to change the default password for the pbxinstall login ID. Only the password can be changed. The login ID cannot be changed.

## Password

**optimum.**

Configuration Menu

- Customer
  - LAN Settings
  - SIP Trunk Configuration
  - Diagnostics
  - System

Set Password

Help

Change the GUI password by filling in the fields below. The password must be between 6 and 8 characters in length.

Username:  
Current Password:  
New Password:  
Confirm Password:

pbxinstall

Submit

Reset

7

Field	Description
Username	Specifies the username for which the password can be changed.
Current Password	Specifies the current password.
New Password	Specifies the new password.
Confirm Password	Confirms the new password.
Submit	Applies the settings configured on this page.
Reset	Clears all fields and selections and allows you to enter new information.

## 4 International Calling

Optimum Voice Business Trunking offers an optional International Calling Service for direct-dialed calls made from the Customer's business or from any phone via the Optimum Voice International Calling remote access number to destinations outside of the United States, Puerto Rico, Canada and the U.S. Virgin Islands at per minute rates. The Customer must login to the Optimum Business Account Center and activate the service on the Optimum Business Trunk Pilot telephone number to activate the service and manage the monthly International spending limit for the account.

Activating International calling on the Pilot TN will enable International calling for all Direct Inward Dial (DIDs) telephone numbers as well. Blocking International calling for one or more DIDs is managed by the customer directly from the PBX phone system configuration. To minimum the exposure to fraudulent calling, It is recommended to limit International calling capability to those DID's that require it and set up an account spending limit that reflects what is necessary to run the business.

It is the Customer (and/or the Customer Agent's) responsibility to properly secure the customer's PBX to prevent the PBX from being compromised and fraudulent calls from being made by unauthorized (internal or external) users. If fraudulent calls are detected, Cablevision reserves the right to disable International Calling until the PBX is properly secured by the customer.



## 5 PBX Configuration

The steps below describe the minimum configuration required to enable the PBX to use Optimum Business SIP Trunking for inbound and outbound calling. Please refer to the Digium product documentation for more information on other advanced PBX features.

The configuration described here assumes that the PBX is already configured and operational with station side phones using assigned extensions or DIDs. This configuration is based on Digium IP PBX version 5.11.1. In this document the address of the Digium is 10.10.108.11 /24 and the Optimum Business SIP Trunk Adaptor (EdgeMarc 4552) is 10.10.108.1 /24.

**Table 1 – PBX Information**

<b>Manufacturer:</b>	Digium
<b>Model:</b>	Digium Switchvox
<b>Version:</b>	5.11.1
<b>Does the PBX send SIP Registration messages (Yes/No)?</b>	Yes
<b>Vendor Contact:</b>	<a href="http://www.digium.com">www.digium.com</a>

## 5.1 SIP Trunking

To configure SIP, navigate to **Setup→Call Routing→VoIP Providers** and click **Create SIP Provider**. Under **SIP Provider Information** enter the relevant information required between the PBX and the Optimum Business SIP Trunk Adaptor. Enter a descriptive name next to **SIP Provider Name** (EM-4552 used for this test). Enter the Pilot DID next to **Your Account ID** along with the password next to **Your Password**.

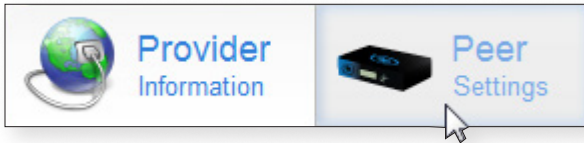
This must match the User ID and Password configured on Optimum Business SIP Trunk Adaptor. This is step 3 of the Optimum Business Set-up Guide.

Next to **Hostname/IP Address** enter the Optimum Business SIP Trunk Adaptor's IP address. The IP address was assigned in step 2 of the Optimum Business SIP Trunk Set-up Guide.

Enter **800** in the **Callback Extension** field which essentially will be the extension for the Auto Attendant. Select **Inband** in the **DTMF Mode** field. **Important:** Only Inband DTFM is supported with Optimum Business Sip Trunking.

SIP Provider Name	<input type="text" value="EM-4552"/>
Your Account ID	<input type="text" value="4085551234"/>
Your Password	<input type="password"/>
Leave blank to keep current password.	
Hostname/IP Address	<input type="text" value="10.10.108.1"/>
Callback Extension	<input type="text" value="Auto Attendant (800)"/>
Default Fax Extension	<input type="text" value=""/>
DTMF Mode	<div> <div>Inband</div> <div>RFC2833</div> <div>Info</div> <div>✓ Inband</div> <div>Auto</div> </div>
<input type="button" value="Save SIP Provider"/>	

Next, from above click on **Peer Settings**.



**Provider** needs to be selected next to **Host Type** and **YES** should be selected next to **Apply Incoming Call Rules to Provider**.

Peer Settings [?](#)

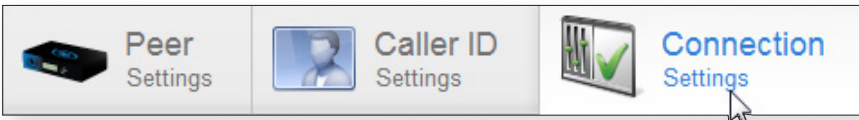
Host Type

Provider

Apply Incoming Call Rules to Provider

YES

Under **Connection Settings** enter **5060** next to **SIP Port** and **3600** next to **SIP Expiry**. The Optimum Business Sip Trunk Adaptor's IP address should be entered next to **Proxy Host** and the Pilot DID should again be entered next to **Authentication User**. Select **YES** next to **Always Trust this Provider**.



Connection Settings

SIP Port

5060

SIP Expiry (in seconds)

3600

Proxy Host

10.10.108.1

Authentication User

4085551234

Always Trust this Provider

YES

Qualify Hosts

NO

Include user=phone in SIP

NO

Use Local Address in From Header

NO

When done click **Save SIP Provider**.

To configure Static Mode, under **Peer Settings** select **Peer** instead of **Provider**.

Host Type

Peer

Provider

✓ Peer

## 5.2 Extensions/DID

Navigate to **Setup→Call Routing→Outgoing Calls** and select **Caller ID**. Click on **Create Single Rule** and provide it a name. Under **Single Caller ID Rule Routing** select **Extension**. Under **Extension Settings** the desired extension needs to be entered. In this case extension “101” was entered. Enter a name next to **caller id name to** and its corresponding DID next to **caller id number to**. When done click **Save Single Caller ID Rule** below.

Single Caller ID Rule Settings ⓘ

Rule Name

Note

Single Caller ID Rule Routing ⓘ

When the following group or extension:

✕

Extension Settings

✕

🔍

is making an outgoing call through rule  ▶ change their

caller id name to  caller id number to

Save Single Caller ID Rule ✓

\*The Internal Rule and all Peered Switchvox rules are excluded.

Next, navigate to **Setup→Call Routing→Incoming Calls** and click **Create Single DID Route**. In a similar way enter a rule name and next to **Incoming DID to Match** enter a DID to be mapped to this specific extension. The extension can be selected below next to **Extension to Route Call** which in this case is “101” again. Enter the Provider name created earlier for SIP next to **Incoming Provider**. **Incoming Call Type** should be set to **Voice Calls**. When done click **Save Single DID Route**.

Single DID Route Settings ?

Rule Name

Note

Incoming DID to Match

Incoming Provider

▶
i

Incoming Call Type

▶

Extension to Route Call

✖
🔍

Process All Calls as a Fax

☐ NO

Save Single DID Route ✓

## 5.3 Dial Plan

To configure Dial Plan navigate to **Setup→Call Routing→Outgoing Calls** and under **Outgoing Call Rules** click on **Create Outgoing Call Rule**. Here an example rule was created allowing 10 digit calls to be prepended by 9. Any name may be given. **Pattern To Match** effectively defines the behavior of this rule. The number “9” was entered next to **Number begins with the digits** to prepend before the dialed call. To strictly allow only 10 digits to be dialed, enter “10” followed by “10” in the second line below to essentially define the digit range. Entering “1” next to **Before connecting the call, trim** allows the 9 to be trimmed. The next field was simply left empty. Under **Call Through** the Provider’s name which is the EM-4552 in this case needs to be selected and under **Rule Final** select **YES**. Other rules may be configured accordingly. When done click **Save Outgoing Call Rule**.

### Rule Name & Note

Rule Name

Note

### Pattern To Match

Number begins with the digits

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

Primary Call Through Provider  ⓘ

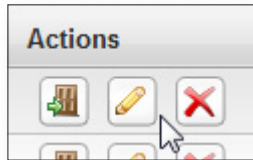
[Add Failover Call Through Provider](#) ✓

### Rule Final

Is this rule final? ☒ YES

[Save Outgoing Call Rule](#) ✓

To decide which of these rules each phone can utilize, navigate to **Setup→Extensions→Manage**. Click the Modify icon next to an extension under **Actions**.



Under **Outgoing Call Rules** is where a specific rule can be allowed or denied to the user using the buttons displayed on the right.

Profile Information
 Phone Settings
 Permissions
 **Outgoing Call Rules**

Outgoing Call Rules ?

Type to Search

	Priority	Name	Description	
✓	1	International	Begins with 9011 and the remainder is 7 to 13 digits in length.	✗
✗	2	1-900 Numbers	Begins with 91(900 976) and the remainder is 7 digits in length.	✓
✓	3	Toll Free	Begins with 91(800 888 877 866 855 844) and the remainder is 7 to 13 digits in length.	✗
✓	4	911	Number exactly matches 911.	
✓	5	Local	Begins with 9 and the remainder is 7 digits in length.	
✓	6	Internal	Any local extension.	
✓	7	Long Distance 91	Begins with 91 and the remainder is 10 digits in length.	
✓	8	Long distance 9	Begins with 9 and the remainder is 10 digits in length.	
✓	9	Operator	Number exactly matches 0.	

Save SIP Extension ✓

When done click **Save SIP Extension** below.

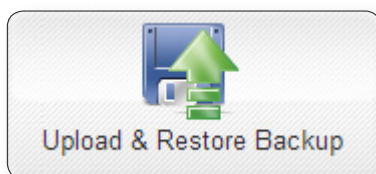


## 5.4 Backup/Restore

To backup the configuration navigate to **Server→Backups**. Here a backup file may be created by selecting **Create Backup**.



To restore select **Upload & Restore Backup**.



After the file has been chosen click **Upload and Restore Backup** to restore the file.

