# Optimum Business Trunking and the ESI-100 PBX Configuration Guide

ESI-100 PBX

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## Table of Contents

# 1 Overview

The purpose of this configuration guide is to describe the steps needed to configure the ESI-100 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.

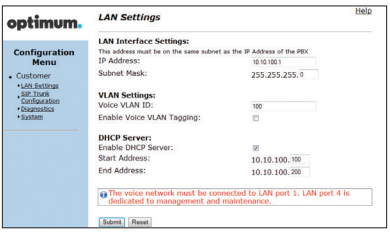
3. Enter login and password and click 'OK'.
   Login: pbxinstall
   Password: s1ptrunk


## 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.



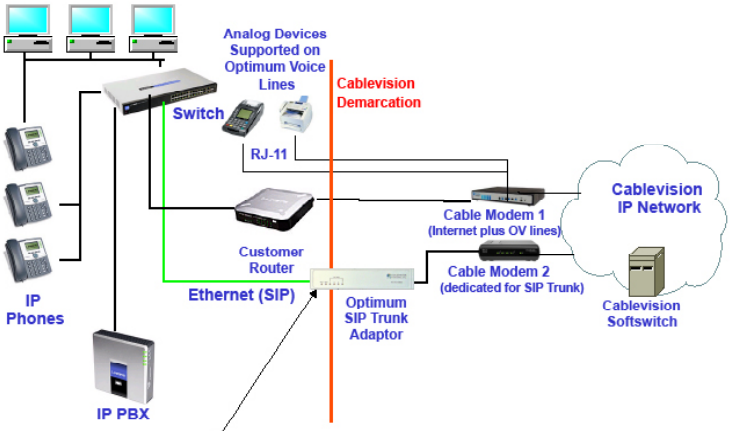

2. Optional: Specify a VLAN for your voice traffic. Click the 'Enable Voice VLAN Tagging' check box. The default VLAN ID is 100.

   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.

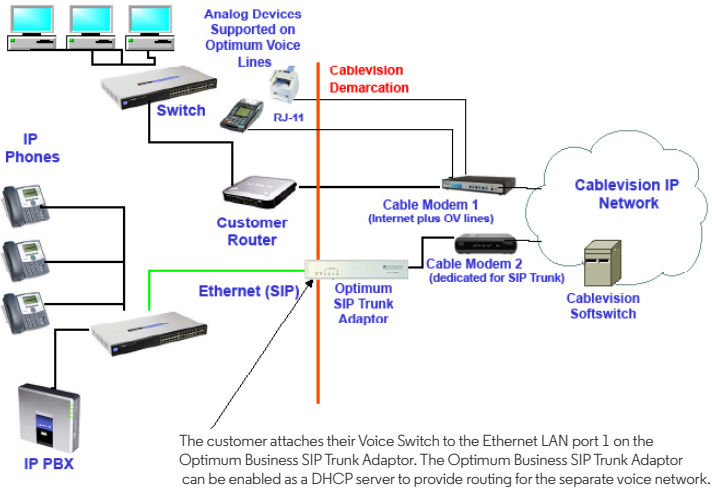
4. Click 'Submit'.

## 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.

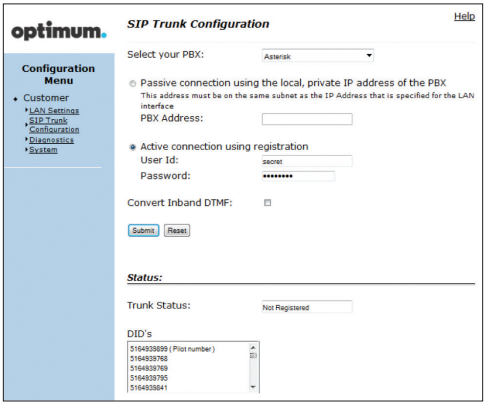
## Diagram 2 SIP Trunk Adaptor for IP-PBXs Example: Separate Voice and Data Networks Configuration




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.

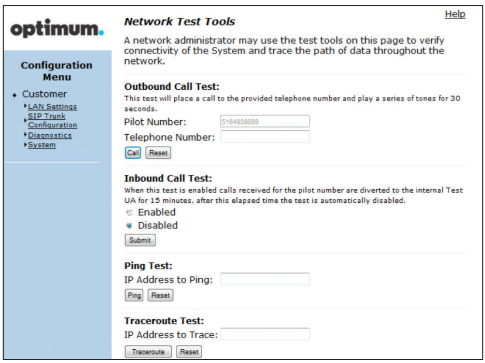
3. The Cablevision network only supports Inband DTMF. Click on the check box next to "Convert Inband DTMF" if you cannot configure your IP PBX to send out Inband DTMF. The DTMF tone duration generated by the phones and/or PBX may need to be increased from their default setting. Some phones and/or PBX have a default setting between 180ms to 200ms. This setting is too low. The recommended setting is 600ms.

4. Click 'Submit'.

## Step 4:

### Diagnostics Link

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

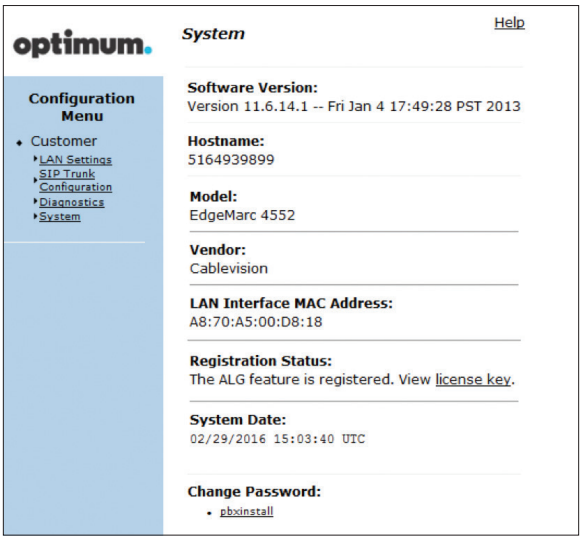


Step 4 continued

| 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. |

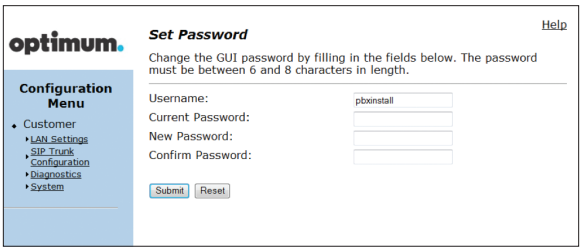
# 3 Additional Set-up Information

## Systems




| 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



| 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 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 ESI-100 version 12.5.25.0.

This configuration guide provides the configuration steps for both PBX registration and Static IP or non-registration modes of PBX operation.

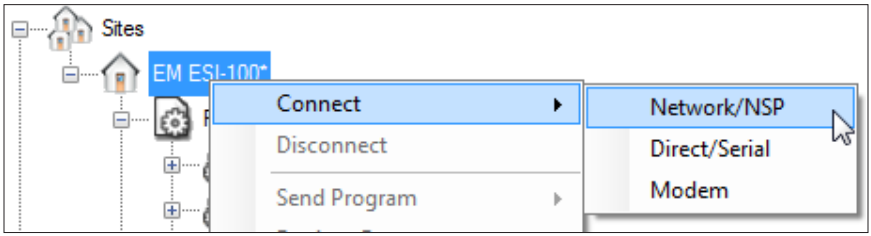
The PBX used in the lab comprises of the following:

Table 1 – PBX Information

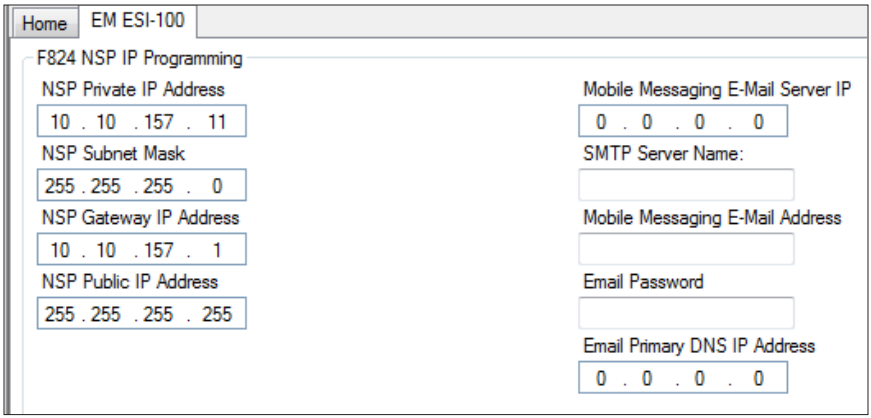
| | |
|---|---|
| **Manufacturer:** | ESI |
| **Model:** | ESI-100 |
| **Software Version:** | 12.5.25.0 |
| **Does the PBX send SIP Registration messages (Yes/No)?** | Yes |
| **Vendor Contact:** | www.esi.com |

## 5.1 Network Settings

The ESI-100 system includes a management, SIP, and RTP port. They are 10.10.157.11, 10.10.157.12, & 10.10.157.13 respectively and all utilized the /24 netmask. To modify network settings first right click the site name and connect to **Network/NSP**.




Navigate to **Program→F8 IP Programming→F82 Local Programming→F824 NSP IP Programming** and enter the address of the management port under **NSP Private IP Address** and the address of the Optimum Business Sip Trunk Adaptor under **NSP Gateway IP Address**.



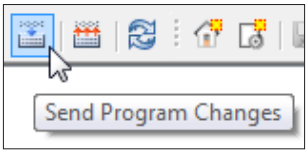

Next navigate to **Program→F8 IP Programming→F84 SIP Card Configuration Programming**. Under **Card Type** what should be selected is **SIGNALING**. The **Primary Processor IP** field is essentially the SIP port which will be used for signaling. The IP address of the Optimum Business Sip Trunk Adaptor should be entered under **Gateway IP**. The **Primary Processor** (SIP IP) address can be entered in the **Public IP field**. The **Secondary Processor IP** field will contain the IP address of the RTP port.

**Note**: The Card number should correlate to the slot being used. In this case it was slot 2.

F84 SIP Card Configuration Programming

| | Card | Support Password | Card Type | Primary Processor IP | Subnet Mask | Gateway IP | NAT Traversal | Public IP | Secondary Processor IP | Primary DNS IP | Secondary DNS IP |
|---|---|---|---|---|---|---|---|---|---|---|---|
| ▶ | 2 | 73829164 | SIGNALING | 10.10.157.12 | 255.255.255.0 | 10.10.157.1 | ☐ | 10.10.157.12 | 10.10.157.13 | 4.2.2.2 | 8.8.8.8 |

When done click Save followed by the **Send Program Changes** button above.




## 5.2 SIP Programming

To program SIP the SIP port must initially be given an IP address. Navigate to **Program→F8 IP Programming→F84 SIP Card Configuration Programming** and assign an IP address to the SIP port under **Primary Processor IP** which in this case is 10.10.157.12 /24.

| Card Type | Primary Processor IP | Subnet Mask |
|---|---|---|
| SIGNALING | 10.10.157.12 | 255.255.255.0 |

Next navigate to **Program→F2 CO Lines→F21 CO Lines→F214 SIP Line Programming→F2142 SIP Account Programming**. The **Prov. Num**. field should be 1. The **ITSP Name** here was "ESI". The selected vendor was **COX CABLE –BD** and the **Signaling Card** was **2**. **Note**: Due to the limited number of options under Vendor to select, the COX CABLE –BD option was the one selected for registration mode. VOXITAS –FS in turn was selected for Static Mode.

The Pilot DID, credentials, and SIP information should be entered in the fields that follow. The address of the Optimum Business Sip Trunk Adaptor should be entered under **ITSP URI** and **Outbound Proxy**. The RTP port range was "10000-11000" and the codec **711**.

F2142 SIP Account Programming

| | Prov. Num | ITSP Name | Vendor | Signaling Card | Primary Number | Acct. Name | Acct. Password | Local SIP Port | Registration Expiration | ITSP URI | Outbound Proxy | Min. Rtp. Port | Max. Rtp. Port | Codec | AssertIdHeader |
|---|---|---|---|---|---|---|---|---|---|---|---|---|---|---|---|
| | 1 | ESI | COX CABLE -BD | 2 | 4085555555 | 4085555555 | | 5060 | 3600 | 10.10.157.1 | 10.10.157.1 | 10000 | 11000 | 711 | |

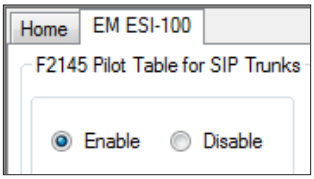
Now navigate to **Program→F2 CO Lines→F21 CO Lines→F214 SIP Line Programming→F2141 SIP Trunk Programming**. The ITSP Name which in this case was "ESI" should be included for SIP Trunks 24-31 as shown. **ID 1** under **Ring 1 (Day)** represents the Auto Attendant.

F2141 SIP Trunk Programming

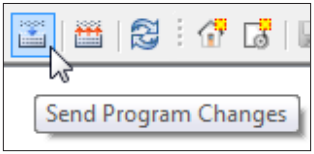
Day Night

| | Number | Name | Outbound Group | Ring Tone | ITSP Name | Ring 1 (Day) |
|---|---|---|---|---|---|---|
| ▶ | 24 | | 9 | 0 | ESI | ID 1 |
| | 25 | | 9 | 0 | ESI | ID 1 |
| | 26 | | 9 | 0 | ESI | ID 1 |
| | 27 | | 9 | 0 | ESI | ID 1 |
| | 28 | | 9 | 0 | ESI | ID 1 |
| | 29 | | 9 | 0 | ESI | ID 1 |
| | 30 | | 9 | 0 | ESI | ID 1 |
| | 31 | | 9 | 0 | ESI | ID 1 |

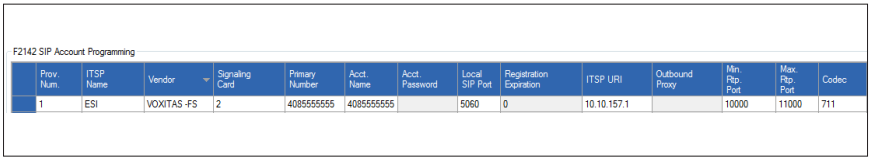
Finally select **Enable** under **F2145 Pilot Table for SIP Trunks** as displayed below.




When done click **Save** followed by the **Send Program Changes** button above.



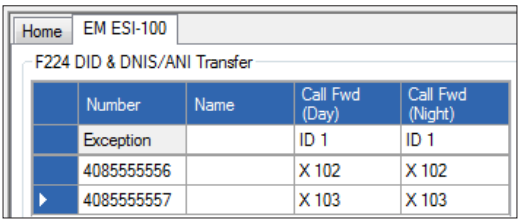
For Static mode exclude the credential section under **F2142 SIP Account Programming** and confirm the device's SIP port address remains specified under **F84 SIP Card Configuration Programming**. The vendor option can be one that excludes the registration fields. VOXITAS -FS in this example was chosen.


F2142 SIP Account Programming

| Prov. Num. | ITSP Name | Vendor | Signaling Card | Primary Number | Acct. Name | Acct. Password | Local SIP Port | Registration Expiration | ITSP URI | Outbound Proxy | Min. Rtp. Port | Max. Rtp. Port | Codec |
|---|---|---|---|---|---|---|---|---|---|---|---|---|---|
| 1 | ESI | VOXITAS -FS | 2 | 4085555555 | 4085555555 | | 5060 | 0 | 10.10.157.1 | | 10000 | 11000 | 711 |

## 5.3 Extensions/DID

To configure extensions and DIDs navigate to **Program→F2 CO Lines→F22 CO Access Deny Tables→F224 DID & DNIS/ANI Transfer**. Click **Add** from below then click the empty field under **Number**. Here is where the DID will be entered. Under **Call Fwd (Day)** and **Call Fwd (Night)** simply right click and select an available extension and this will be mapped to the entered DID.

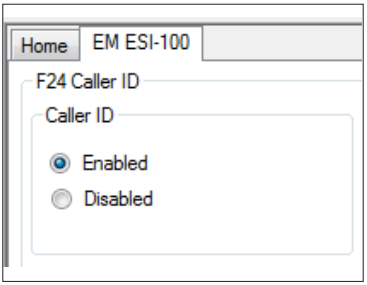

Home | EM ESI-100

F224 DID & DNIS/ANI Transfer

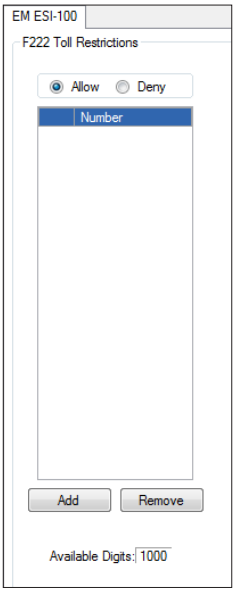
| Number | Name | Call Fwd (Day) | Call Fwd (Night) |
|---|---|---|---|
| Exception | | ID 1 | ID 1 |
| 4085555556 | | X 102 | X 102 |
| 4085555557 | | X 103 | X 103 |

The **Exception** field defines the default route, in this case it is the Auto Attendant depicted as **ID 1**.

Navigate to **Program→F2 CO Lines→F24 Caller ID** and select **Enabled** under **Caller ID**.



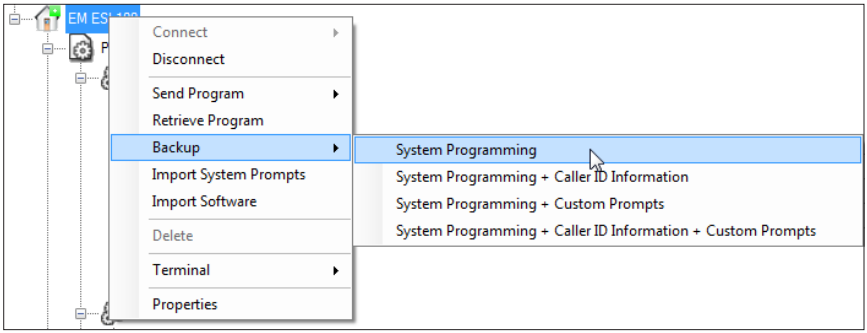
To control the behavior of particular outbound numbers navigate to **Program→F2 CO Lines→F22 CO Access Deny Tables→F222 Toll Restrictions** and here numbers can be allowed or denied.




**Note**: The Cablevision network supports Inband DTMF only. ESI uses Inband DTMFby default and this is not an configurable option.

## 5.4 Backup/Restore

To backup the device file simply right click the site and navigate to **Backup** followed by **System Programming**.




Once a backup has been performed, a backup folder will appear under **Program**. Right click this folder to restore.