

# Optimum Business Trunking and the Fonality Trixbox Pro IP PBX Standard Edition V4.1.2- p13 Configuration Guide

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## 1 Overview

The purpose of this configuration guide is to describe the steps needed to configure the Fonality Trixbox Pro 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:

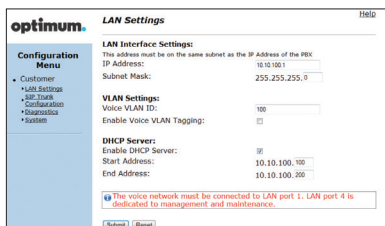
#### Click on the LAN Settings Link

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.

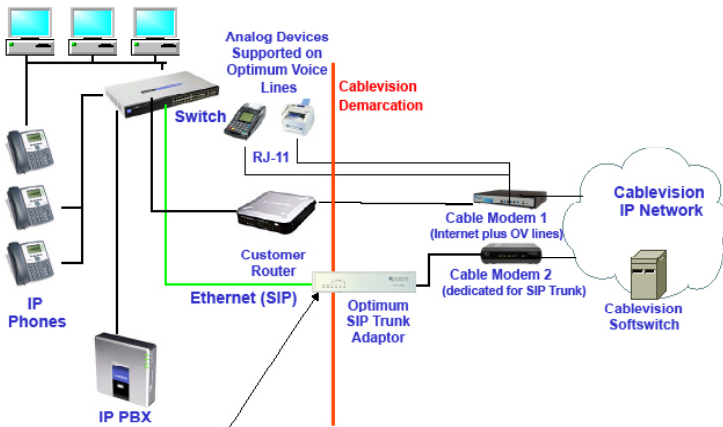
2. Optional: Specify a VLAN for your voice traffic. Click the 'Enable Voice VLAN Tagging' check box. The default VLAN ID is 100.

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



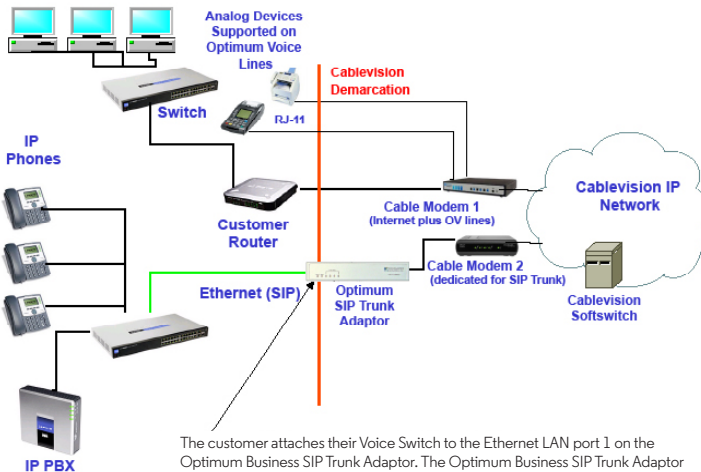
3. Optional: Enable the DHCP server. This will allow the SIP Trunk Adaptor to act as a DHCP server, which will provide IP addresses to the voice network, and create a dedicated voice LAN, as per diagram 2.
4. Click 'Submit'.

**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. Below these are 'Submit' and 'Reset' buttons. A 'Status:' section shows 'Trunk Status:' as 'Not Registered'. At the bottom, there is a 'DID's' list with a scrollable view of numbers: 0164030809, 0164030760, 0164030769, 0164030765, and 0164030841.

## 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 the previous screenshot. The main content area has a title '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 is a section for 'Outbound Call Test:' with a note 'This test will place a call to the provided telephone number and play a series of tones for 30 seconds.' It includes fields for 'Pilot Number:' (0164030809) and 'Telephone Number:', with 'Call' and 'Reset' buttons. The 'Inbound Call Test:' section has a note 'When this test is enabled calls received for the pilot number are diverted to the internal Test UK for 15 minutes, after this elapsed time the test is automatically disabled.' It has radio buttons for 'Enabled' (selected) and 'Disabled', with a 'Submit' button. The 'Ping Test:' section has a field for 'IP Address to Ping:' and 'Ping'/'Reset' buttons. The 'Traceroute Test:' section has a field for 'IP Address to Trace:' 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.

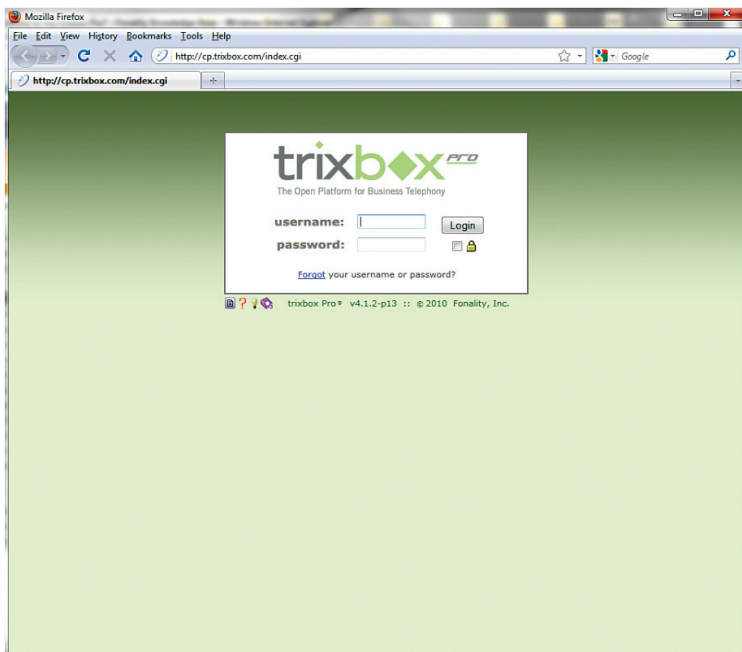


## 5 PBX Configuration

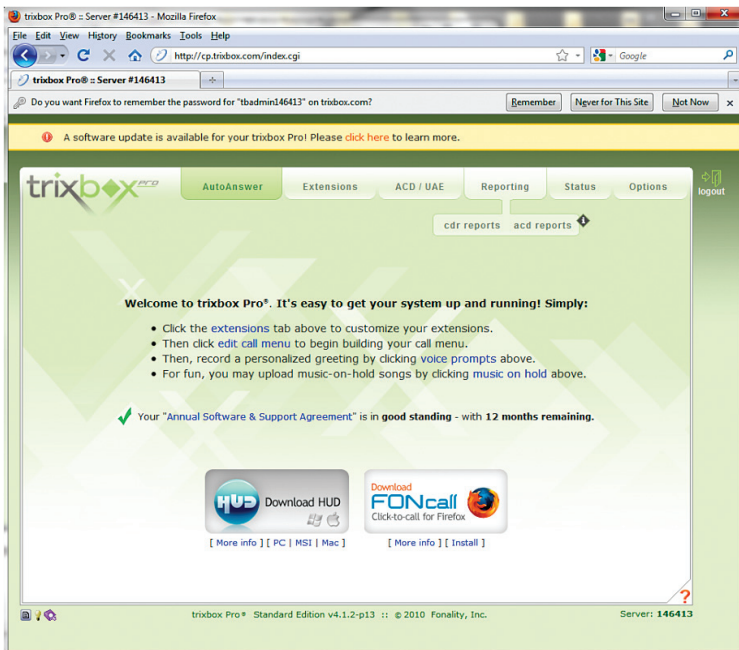
The steps on the next page describe the minimum configuration required to enable the PBX to use Optimum Business SIP Trunking for inbound and outbound calling. Please refer to the Trixbox Pro product documentation for more information on advanced PBX features.

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 Trixbox Pro Standard Edition v4.1.2-p13.

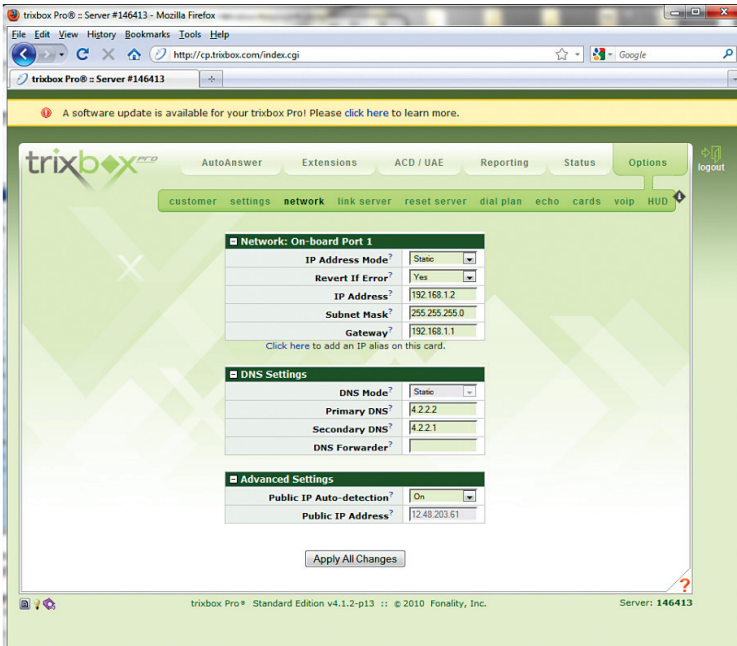
1. The PBX, the phones and the Optimum SIP Trunk Adaptor should be on the same LAN segment. The PBX should be configured with a static IP address (i.e.: 192.168.1.2/24) and its gateway set to the Optimum Business SIP Trunk Adaptor's port 1 LAN IP address (i.e.: 192.168.1.1/24). All the Polycom phones are shipped with default set to obtaining an IP address via DHCP. All the Polycom phones also support Auto Discovery that is strongly recommended for the PBX. You should enable the Optimum Business SIP Trunk Adaptor's DHCP server option when connected to the LAN, so the phones can get an assigned IP address and register with the PBX, once the PBX is configured for adding the phones.
2. To configure the PBX, launch the browser and access <http://cp.trixbox.com> from an Internet-ready PC.



3. Enter the **username** and **password** for the PBX and hit the “**Login**” button.



4. Select **Options>network** to verify or change the network configuration on the PBX. Hit the **Apply All Changes** button if you make any changes.



5. Select “**Options>voip**” to create a VoIP account for communicating with the Optimum Business SIP Trunk Service.
  - a. Enter a descriptive name in the “**Route Name**” field.
  - b. Select “Other” and “SIP” from the 2 drop-down menus for the “**Provider**” fields.
  - c. Select “yes” in the “**Register**” field if the Optimum Business SIP Trunk Adaptor is configured to receive SIP registration from the PBX; select “no” if Optimum Business SIP Trunk Adaptor is configured to communicate with the PBX via the PBX’s static IP address.
  - d. If PBX registration is required, enter the username in the “**Username**” field and password in the “**Password**” field. Note that the username and password must match what are configured on the Optimum Business SIP Trunk Adaptor. In this example, the pilot DID, 6316769604, is used for both the username and the password.
  - e. In the Advanced section, select “no” in the “**NAT**” field.
  - f. If PBX registration is required, make sure the SIP registration string (see example from screen shot below) entered in the “**Register String**” field is in the form of “username:password@server/username”.
  - g. Leave other fields as default and hit the “**Add VoIP Account**” button.

trixbox Pro® - Server #146413 - Mozilla Firefox

File Edit View History Bookmarks Tools Help

http://cp.trixbox.com/index.cgi

trixbox Pro® - Server #146413

A software update is available for your trixbox Pro! Please [click here](#) to learn more.

trixbox PRO

AutoAnswer Extensions ACD / UAE Reporting Status Options

customer settings network link server reset server dial plan echo cards voip HUD

Your trixbox Pro network may not be configured to work with SIP or IAX2 VoIP providers. [\[learn more\]](#)

Watch a video demonstration of adding a VoIP trunk!

Welcome to your VoIP Accounts area. From this screen you will be able to see your existing VoIP Accounts as well as add new VoIP Accounts - now of type SIP or IAX2!

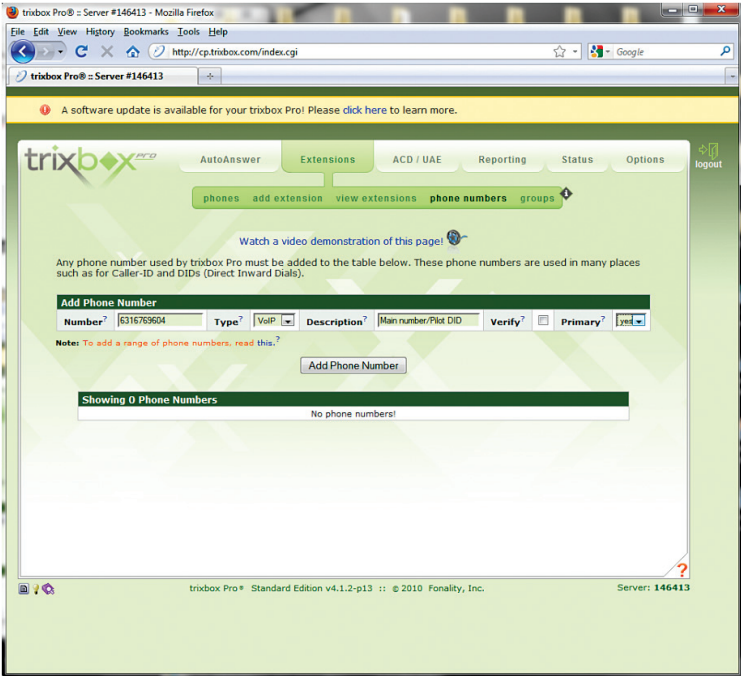
**Update VoIP Account**

|             |            |           |             |     |
|-------------|------------|-----------|-------------|-----|
| Route Name* | EM4552     | Provider* | Other       | SIP |
| Username*   | 6316769604 | Register* | no          |     |
| Password*   | *****      | Server*   | 192.168.1.1 |     |

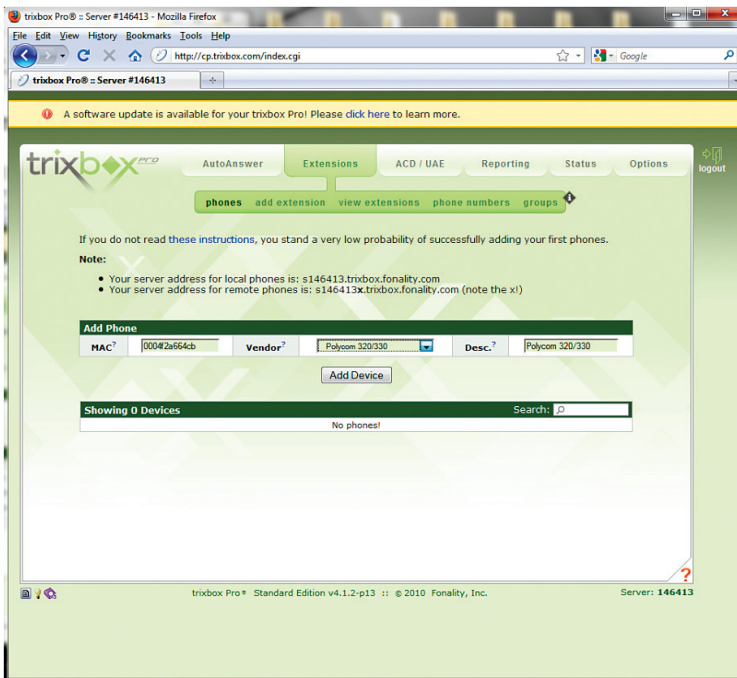
☐ Advanced

|                 |             |                 |                |
|-----------------|-------------|-----------------|----------------|
| From User*      |             | Direction*      | both - (frand) |
| From Domain*    | 192.168.1.1 | TOS*            | none           |
| Allowed IPs*    |             | Authentication* | plaintext      |
| Disallowed IPs* |             | Transfer*       | yes            |
| Outbound Proxy* |             | Reinvite*       | no             |
| Codec 1*        | ulaw        | NAT*            | no             |
| Codec 2*        | alaw        | Insecure*       | very           |
| Codec 3*        | gsm         | DTMF Mode*      | auto           |

6. Select “**Extensions>phone numbers**” to add the DIDs/phone numbers assigned for the SIP trunk services.
  - a. Enter the phone number in the “**Number**” field.
  - b. Select “VoIP” in the “**Type**” field.
  - c. Enter a description in the “**Description**” field.
  - d. If the DID is a pilot DID (main number), select “yes” for the “**Primary**” field.
  - e. Hit the “**Add Phone Number**” button.



7. Select “**Extensions>phones**” to create/add device for phones.
  - a. Enter the MAC address of the phone in the “**MAC**” field.
  - b. Select the phone vendor in the “**Vendor**” field.
  - c. Enter a description in the “**Desc.**” field.
  - d. Hit the “**Add Device**” button. If you have trouble getting the phone to register with the PBX, click the “**these instructions**” link for more information.



8. Select **"Extensions>add extension"** to create/add extension and associate the extension with the phone and the assigned DID.
  - a. Enter an extension number in the **"Extension Number"** field.
  - b. Enter a description in the **"Description"** field.
  - c. Enter the first name and last name of the phone user in the **"First / Last Name"** field.
  - d. Enter a username in the **"Web Username"** field.
  - e. Enter a password in the **"Web Password"** field.
  - f. In the **"Inbound Phone No."** field, select the DID for routing incoming calls to this extension. Note that in this example, we use the pilot DID to associate with the first extension number.
  - g. In the **"Outbound Caller-ID"** field, select the DID/username as the caller ID. Note that for PBX registration mode, the Optimum Business SIP Trunk Adaptor will always relay the outbound call with the pilot DID as the caller ID. Also, the caller ID that is used by the PBX for any outbound call must be the same username that is successfully registered with the Optimum Business SIP Trunk Adaptor. For static IP mode, the outbound caller ID can be the same as the DID phone number.
  - h. In the **"Phones / Devices"** field, select the phone/device for this extension. Note that the detected phone/device will show up in the available list even when it can not successfully register with the PBX.
  - i. Enter a password in the **"Voicemail Password"** field.
  - j. Select "no" in the **"NAT"** field.
  - k. Leave all other fields as default and hit the **"Add Extension"** button.

**Add Extension: >> step one <<**

Extension Number? 111  
 Description? Extension 111  
 First / Last Name? FN111 LN111  
 Web Username? FN111  
 Web Password? \*\*\*\*\*  
 Employee Email?  
 Instant Messenger?  
 Employee Mobile?  
 Mobile SMS Email? --- Select ---  
 Inbound Phone No.? 631-676-9604  
 Outbound Caller-ID? 631-676-9604  
 Phones / Devices? polycom 331-SIP0004F2A664CB (x114)

**Add Extension: >> step two <<**

Ring Type? Ringing  
 Ring Seconds? 20 seconds  
 Call Forwarding? disabled  
 Department? -- None --  
 Queue Auto-logout? disabled  
 Queue Ignore-if-busy? yes no  
 Queue Pwd Required? yes no  
 In Name Directory? yes no  
 In Company Directory? yes no  
 In Blast Group? yes no  
 In HUD? yes no  
 Voicemail Enabled? yes no

**Voicemail Settings**

Voicemail Box? New Mailbox  
 Voicemail Password? \*\*\* \*numbers only  
 Voicemail Email?  
 Voicemail Pager Email? --- this feature disabled ---  
 Email Attachments? yes no  
 Delete When Emailed? yes no  
 Enable CallReturn? yes no  
 Enable CallOut? yes no

**Advanced Settings**

NAT? no  
 DTMF Mode? f62833 (default)  
 Quality? yes no  
 Queue Press-to-accept? yes no  
 Multiple Queue Calls? yes no  
 Private Extension? yes no  
 Visual-only Extension? yes no

[Add Extension](#)

9. Select **“Extensions>view extensions”** to verify that all phones are properly registered with the PBX. A green icon should be shown next to each extension. Mouse-over the “status” header to see what other status you may get. Note that you may need to set the **“NAT”** field for the extension to “yes” in order to get the green icon to appear.

You are using an unsupported browser with known compatibility issues. Supported browsers are Internet Explorer 7.0 and Firefox 3.0.

Remote phones and remote HUD users will not be able to connect to your trixbox Pro. [\[learn more\]](#)

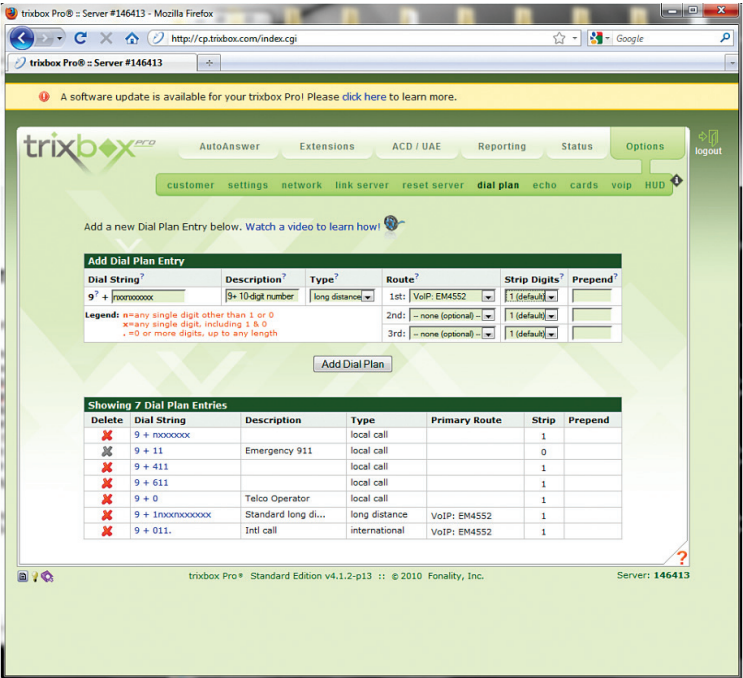
You have 1 license left out of 4. [\[buy more\]](#)

| Del | Ext | Status | User's Name  | Description   | Department | Direct Dial  | Blast | HUD | Fwd. | Login                 |
|-----|-----|--------|--------------|---------------|------------|--------------|-------|-----|------|-----------------------|
| X   | 111 | ●      | LN111, FN111 | Extension 111 |            | 631-676-9604 |       | ✓   |      | <a href="#">LOGIN</a> |
| X   | 112 | ●      | LN112, FN112 | Extension 112 |            | 631-676-9605 |       | ✓   |      | <a href="#">LOGIN</a> |
| X   | 113 | ●      | LN113, FN113 | Extension 113 |            | 631-676-9606 |       | ✓   |      | <a href="#">LOGIN</a> |

Showing 22 System Extensions / Key Codes

trixbox Pro® Standard Edition v4.1.2-p13 :: © 2010 Fonality, Inc. Server: 146413

10. Select “**Options>dial plan**” to add dial plan for making outbound calls with “9+ 10-digit number”.
  - a. Enter “nxxxxxxxxx” in the “**Dial String**” field.
  - b. Enter a description in the “**Description**” field.
  - c. Select “long distance” in the “**Type**” field.
  - d. Select the Route name for the Optimum Business SIP Trunk Adaptor in the “**Route**” field.
  - e. Leave other fields as default and hit the “**Add Dial Plan**” button.





11. To configure the Trixbox PBX with a specific DTMF mode, follow the steps below.

Go to the “Options” tab, then select the “voip” link. In the drop-down box labeled “DTMF Mode”, there are four options: RFC2833, Auto, Info, and Inband. Select “Auto”. This will allow both Inband and RFC2833 protocols.

