

Optimum Business
Trunking and the Siemens
OpenScape Office MX IP
PBX vOMX_V2_R1.4.0_001
Configuration Guide

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1 Overview

The purpose of this configuration guide is to describe the steps needed to configure the Siemens 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:

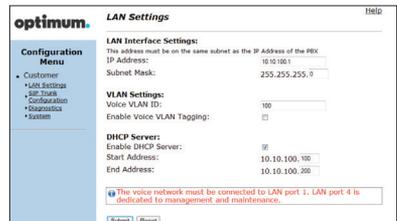
Click on the LAN Settings Link

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.

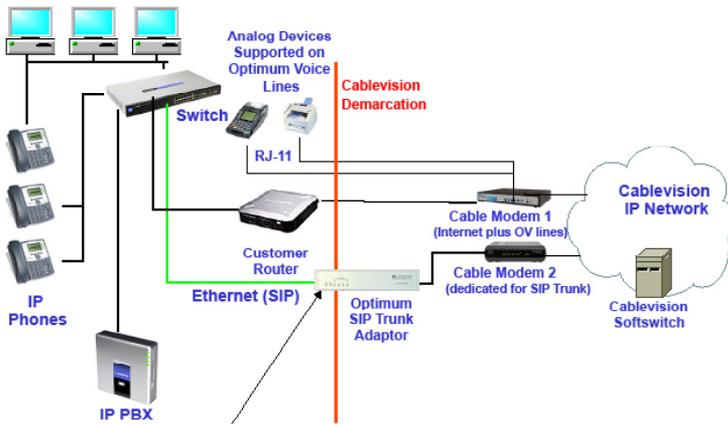
Note: VLAN 200 should not be used. It is dedicated to port 4 for management.



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.

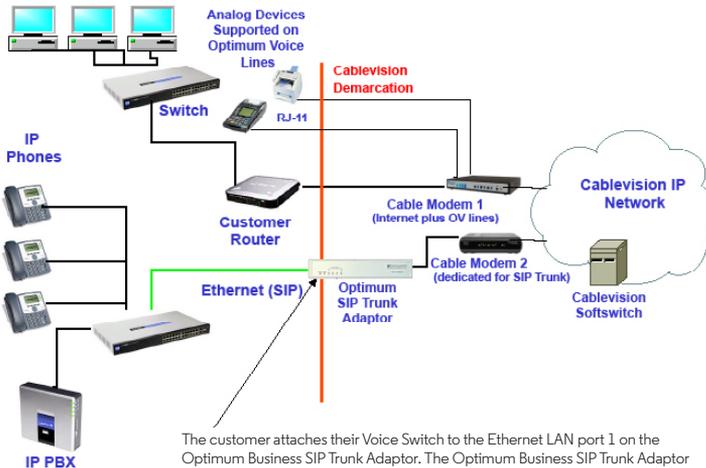
4. Click 'Submit'.

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


System
[Help](#)

Configuration Menu

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

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


Set Password
[Help](#)

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

Configuration Menu

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

Username:

Current Password:

New Password:

Confirm 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

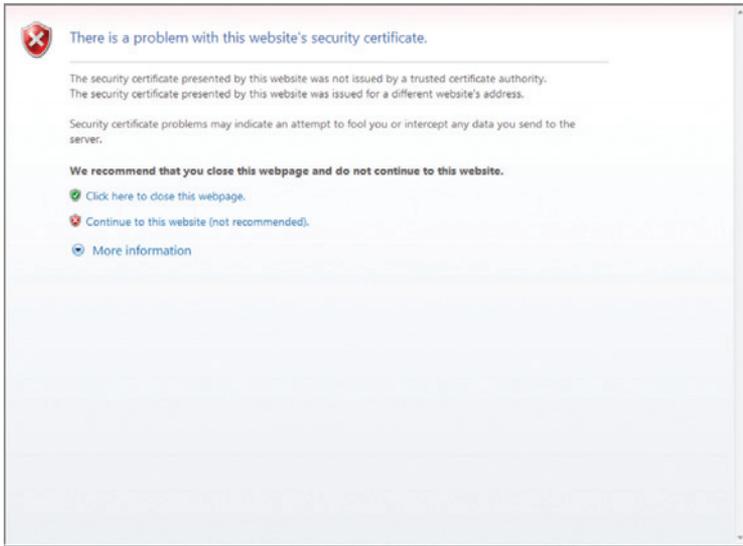
The steps on the next pages describe the minimum configuration required to enable the PBX to use Optimum Business SIP Trunking for inbound and outbound calling. Please refer to the OpenScape Office MX product documentation for more information on advanced PBX features.

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 the Siemens OpenScape Office MX IP PBX vOMX_V2_R1.4.0_001.

The PBX is shipped with a default IP address of 192.168.1.2/24 for the LAN ports. Port 2 and port 3 are disabled. Port 1 (or port 4) should be connected to the phones in the same LAN segment. The PBX's WAN port should be connected to the Optimum Business SIP Trunk Adaptor's LAN port 1 in the same LAN segment. In the lab, the Optimum Business SIP Trunk Adaptor's LAN port 1 and the PBX's WAN port have been assigned with IP addresses of 10.10.100.1/24 and 10.10.100.11/24 respectively.

To configure the PBX, start the browser from a PC and HTTPS to the PBX's LAN or WAN IP address to access the OpenScape Office Assistant.

Click the “**Continue to this website (not recommended)**” link.

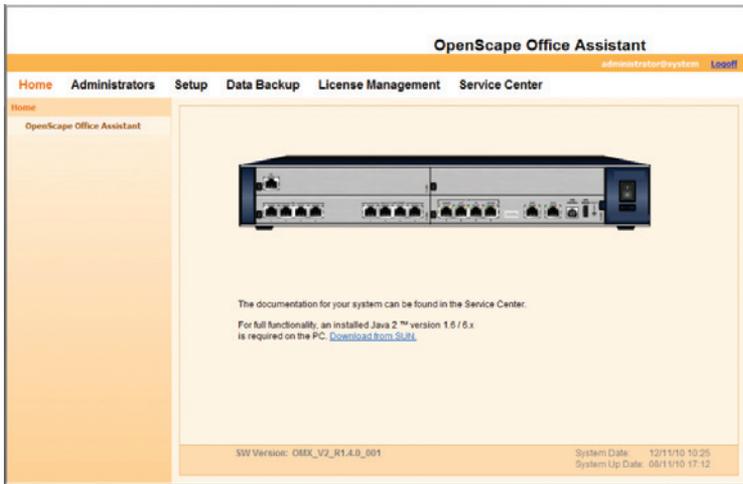


From the login screen, select your choice of language in the “**Language**” field, enter the default username of “**administrator@system**” (or enter “**administrator**” and then press the tab key) in the “**User name**” field, enter the password in the “**Password**” field and then click the “**Login**” button.

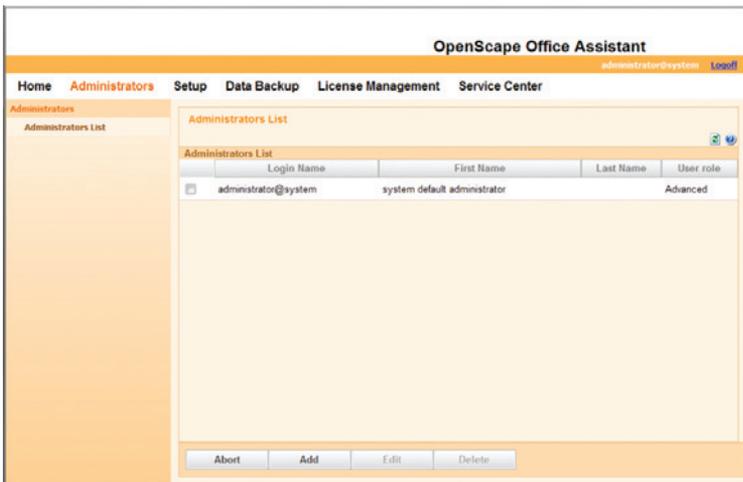
Note: The initial default password is “administrator”, and it must be changed when you log in for the first time.



Click the **“Administrators”** tab to create an administrator to have a user role of **“Expert”**. When logging in as an administrator with Expert user role, the PBX will allow you to make quick configuration changes without going through the lengthy Setup wizards.



Click the **“Add”** button to create an administrator with Expert user role:

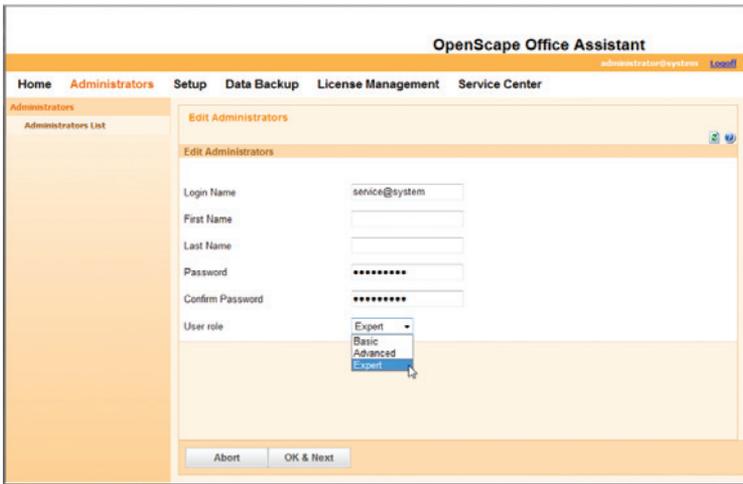


Enter **“service@system”** (or enter **“service”** and hit the tab key) in the **“Login Name”** field.

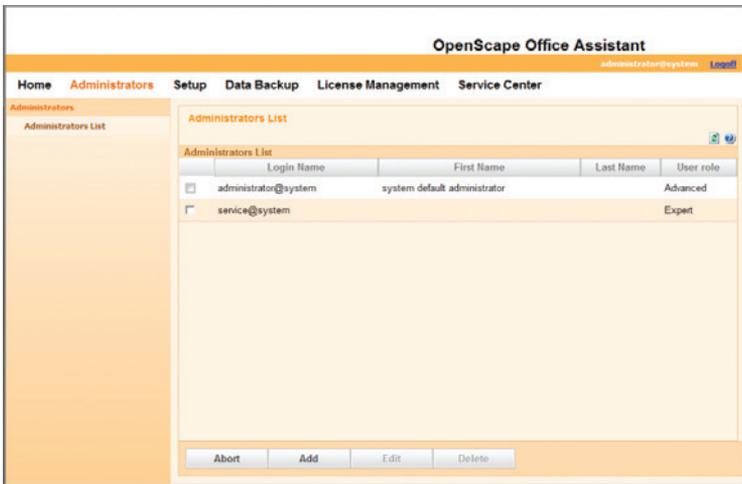
Enter the same password in both the **“Password”** field and the **“Confirm Password”** field.

Select **“Expert”** in the **“User role”** field.

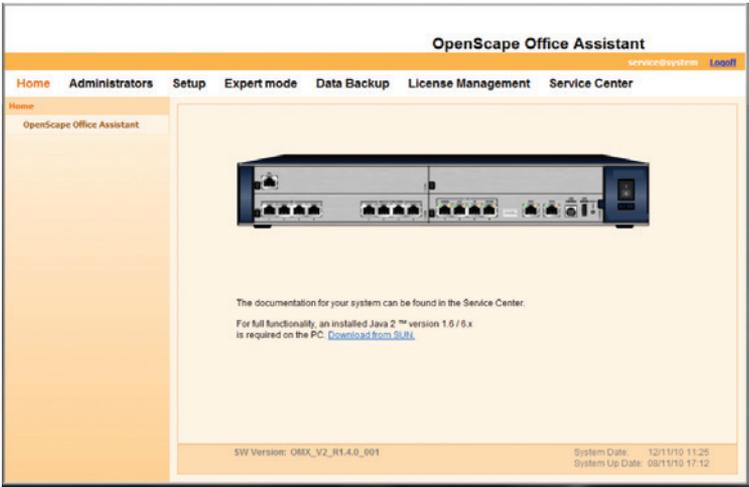
Click the **“OK & Next”** button.



Now click the “**Logoff**” link to log out.

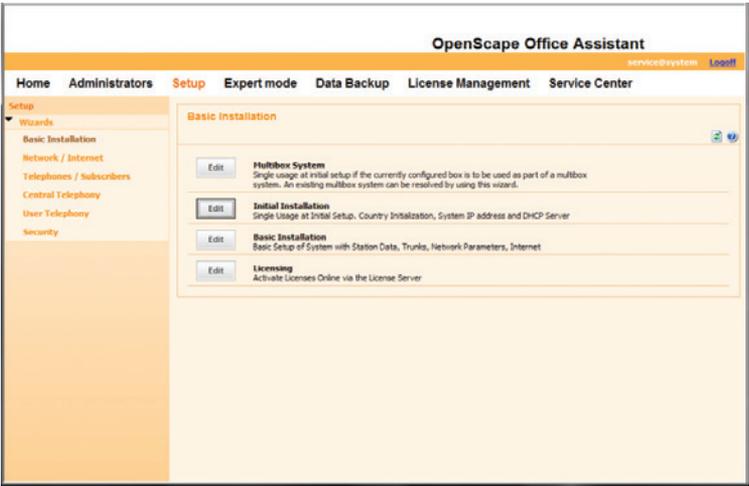


Login again as “**service@system**” and the OpenScope Office Assistant should show an “**Expert mode**” tab. Unlike the Wizards under the Setup tab, Expert mode allows making quick changes to the configurations.

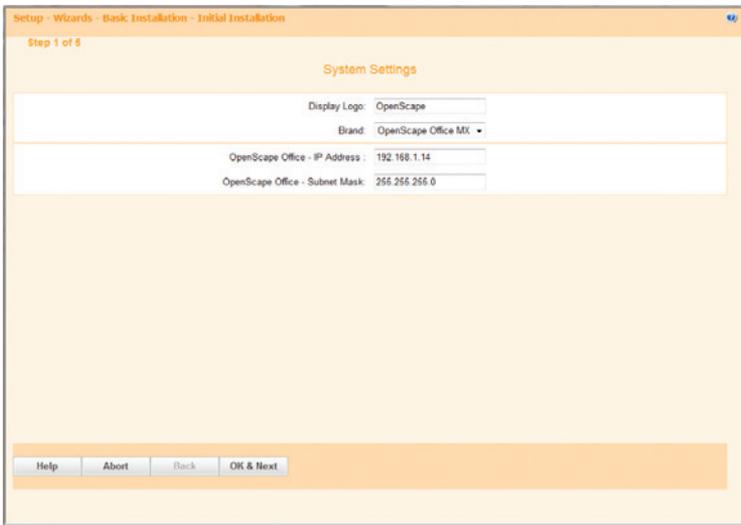


Navigate to “**Setup**”>“**Wizards**”>“**Basic Installation**” to perform Initial Installation setup for the PBX.

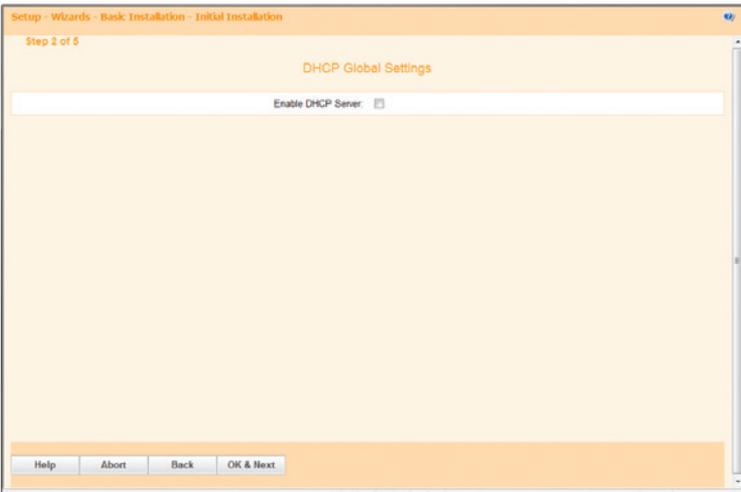
Click the “**Edit**” button next to “**Initial Installation**”.



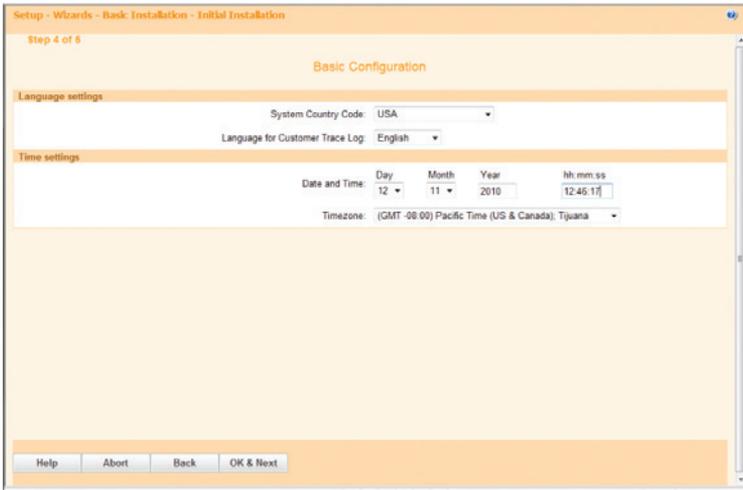
A new window will pop up when the Wizard starts running. Under System Settings, enter “**OpenScope**” in the “**Display Logo**” field, select “**OpenScope Office MX**” in the “**Brand**” field, enter PBX’s LAN IP address (we are using “192.168.1.14” in this example, not the default IP address of “192.168.1.2”) in the “**OpenScope Office – IP Address**” field, enter the subnet mask in the “**OpenScope Office – Subnet Mask**” field and click the “**OK & Next**” button. **Note:** The OptiPoint 420 Standard telephones (default password is “123456”) should be configured to use the PBX’s LAN IP address to access the PBX.



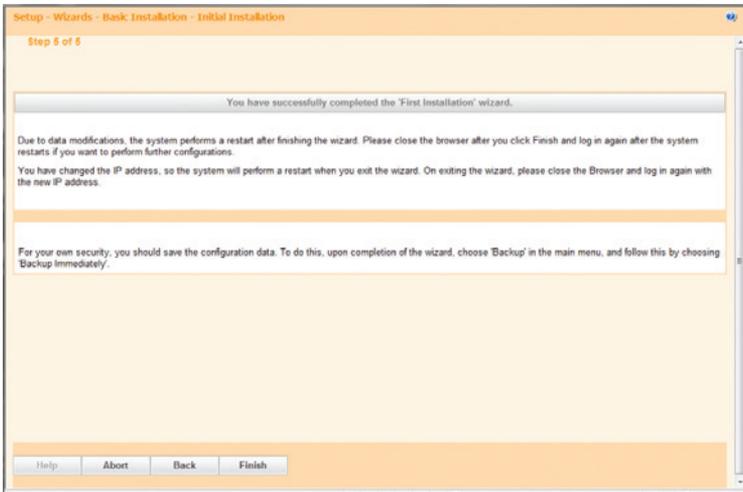
Under DHCP Global Settings, uncheck the **“Enable DHCP Server”** checkbox and click the **“OK & Next”** button. **Note:** In this example, all the IP phones have a pre-assigned IP address and have been configured to communicate with the PBX using H.323 protocol.



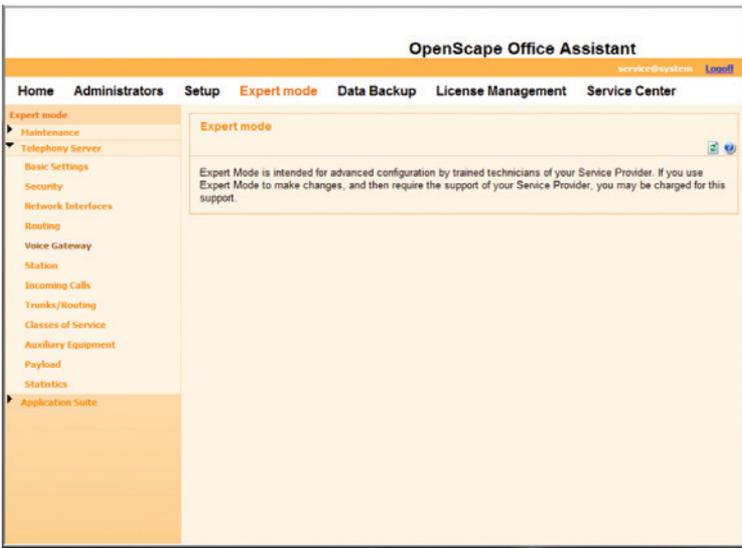
Under Language settings, select country from the drop-down list of the **“System Country Code”** field and select language from the drop-down list of the **“Language for Customer Trace Log”** field. Under Time settings, enter date and time in the **“Date and Time”** fields and select time zone in the **“Timezone”** field and then click the **“OK & Next”** button.



Click the **Finish** button and you are done with Initial Installation setup.



Navigate to **Expert mode** > **Telephony Server** > **Voice Gateway** to configure the PBX to use UDP for SIP protocol.



A new window showing the Voice Gateway screen will pop up.

Select **“SIP Parameters”**. Make sure the **“SIP via UDP”** checkbox is checked.

Note: The “Maximum possible Provider Calls” field shows “0” because the Optimum Business SIP Trunk Adaptor has not been set up as a SIP trunk service provider yet.

Click the **“Apply”** button.

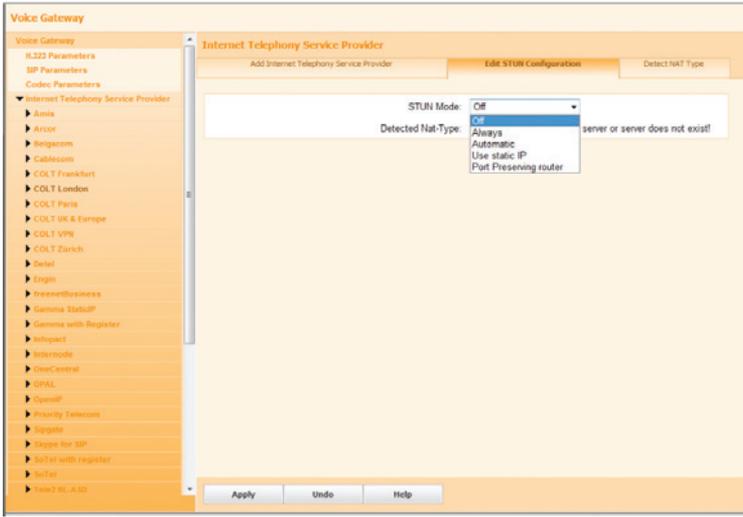


From the Voice Gateway screen, click **“Internet Telephony Service Provider”** to disable any Stun operation.

Click the **“Edit STUN Configuration”** tab, select **“Off”** from the drop-down list of the **“STUN Mode”** field.

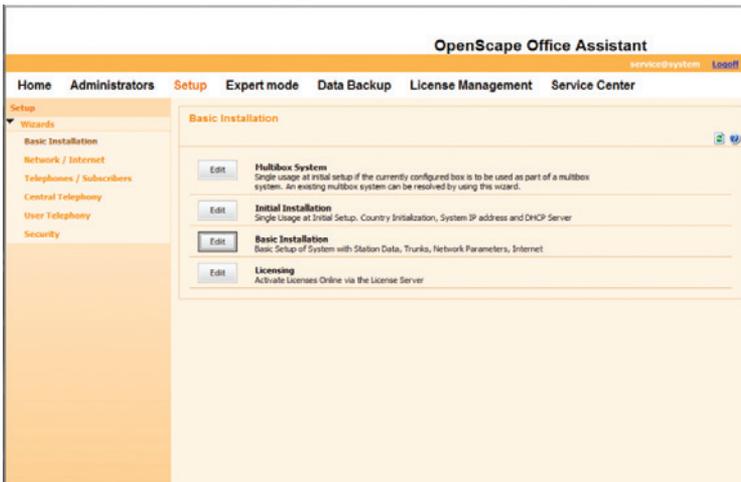
Click the **“Apply”** button.

Click the Windows **“x”** (close) button (above the Voice Gateway screen) to close the window.

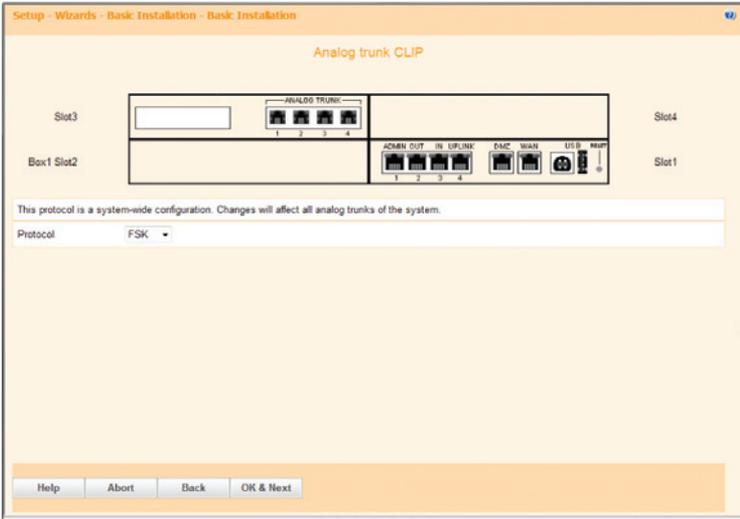


Navigate to **“Setup”** > **“Wizards”** > **“Basic Installation”** to perform Basic Installation setup for the PBX.

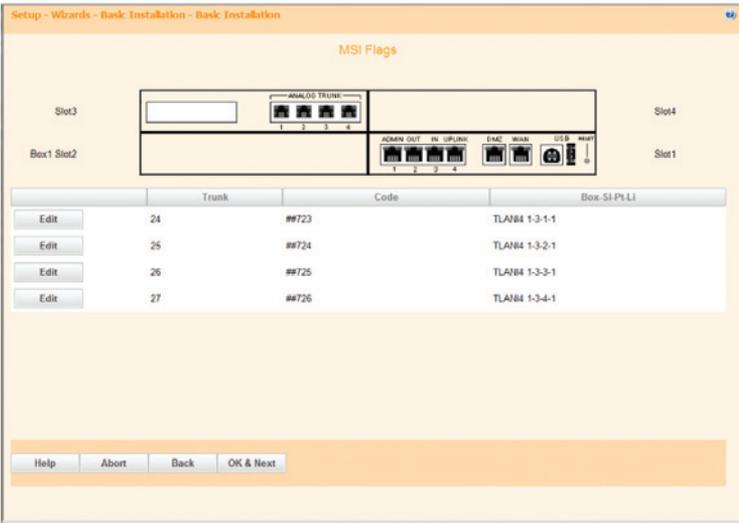
Click the **“Edit”** button next to **“Basic Installation”**.



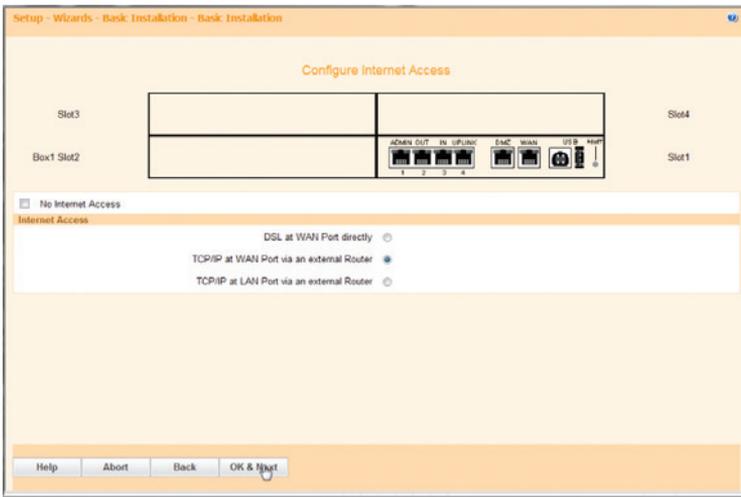
Click the **“OK & Next”** button (we are not using the analog trunks).



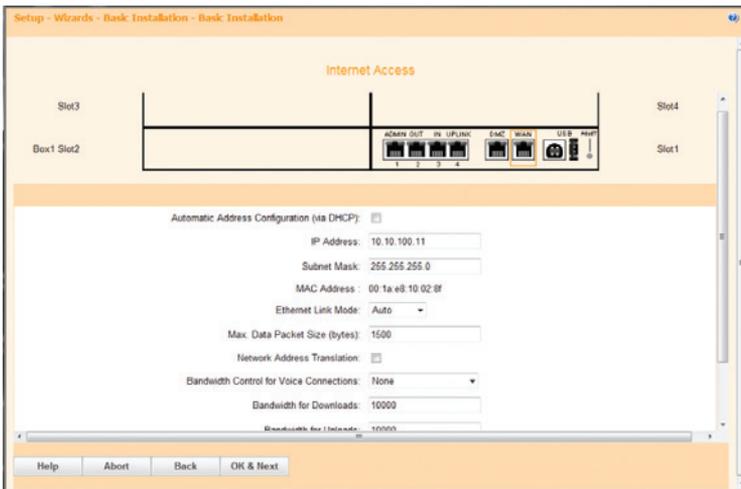
Click the **“OK & Next”** button (we are not using the analog trunks).



Uncheck the **“No Internet Access”** checkbox, select the **“TCP/IP at WAN Port via an external Router”** radio button and then click the **“OK & Next”** button.



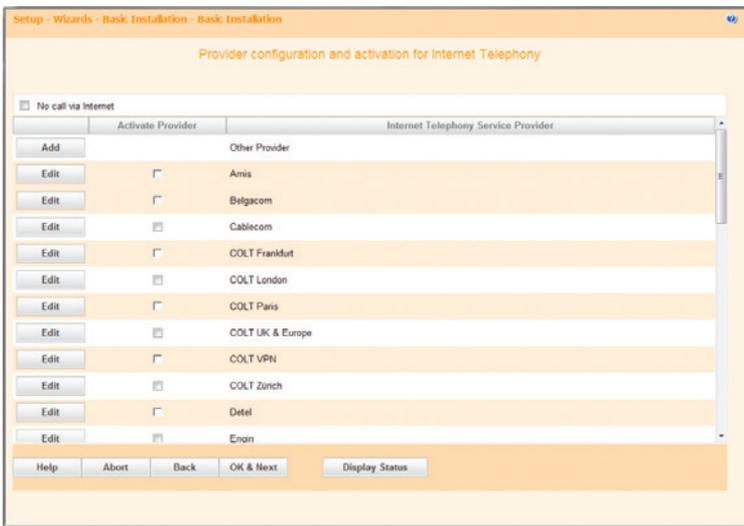
Make sure the “**Automatic Address Configuration (via DHCP)**” checkbox is unchecked, enter PBX’s WAN port IP address in the “**IP Address**” field”, enter the subnet mask in the “**Subnet Mask**” field, leave other fields as default and then click the “**OK & Next**” button.



Enter an IP address for the DNS server in the “**IP Address of DNS Server**” field, enter the Optimum Business SIP Trunk Adaptor’s IP address in the “**IP Address of Default Router**” field and then click the “**OK & Next**” button.



Uncheck the **“No call via Internet”** checkbox and then click the **“Add”** button to add the Optimum Business SIP Trunk Adaptor as an Internet Telephony Service Provider.



Enter a name in the **“Provider Name”** field. **Note:** that “EWN” is used in this example and the PBX will identify it as one of the service providers.

Check the **“Enable Provider”** checkbox.

Enter the Optimum Business SIP Trunk Adaptor’s IP address in the **“Domain Name”** field.

For registration mode: Check the **“Use Registrar”** checkbox.

Enter the Optimum Business SIP Trunk Adaptor’s IP address in the **“IP Address/Host name”** field. Enter **“5060”** in the **“Port”** field.

Enter **“120”** or other preferred numbers in the **“Reregistration Interval at Provider (sec)”** field.

In the Provider Proxy section, enter the Optimum Business SIP Trunk Adaptor’s IP address in the **“IP Address/Host name”** field and enter **“5060”** in the **“Port”** field.

Setup - Wizards - Basic Installation - Basic Installation

Internet Telephony Service Provider

Provider Name: EWN
Enable Provider:
Domain Name: 10.10.100.1

Provider Registrar

Use Registrar:
IP Address / Host name: 10.10.100.1
Port: 5060
Reregistration Interval at Provider (sec): 120

Provider Proxy

IP Address / Host name: 10.10.100.1
Port: 5060

Provider Outbound Proxy

Use Outbound Proxy:
IP Address / Host name: 0.0.0.0
Port:

Provider STUN

Help Abort Back **OK & Next** Delete Data

If the Optimum Business SIP Trunk Adaptor has been configured for static IP mode, instead of registration mode, uncheck the **“Use Registrar”** checkbox.

Setup - Wizards - Basic Installation - Basic Installation

Internet Telephony Service Provider

Provider Name: EWN
Enable Provider:
Domain Name: 10.10.100.1

Provider Registrar

Use Registrar:
IP Address / Host name: 10.10.100.1
Port: 5060
Reregistration Interval at Provider (sec): 120

Provider Proxy

IP Address / Host name: 10.10.100.1
Port: 5060

Provider Outbound Proxy

Use Outbound Proxy:
IP Address / Host name: 0.0.0.0
Port:

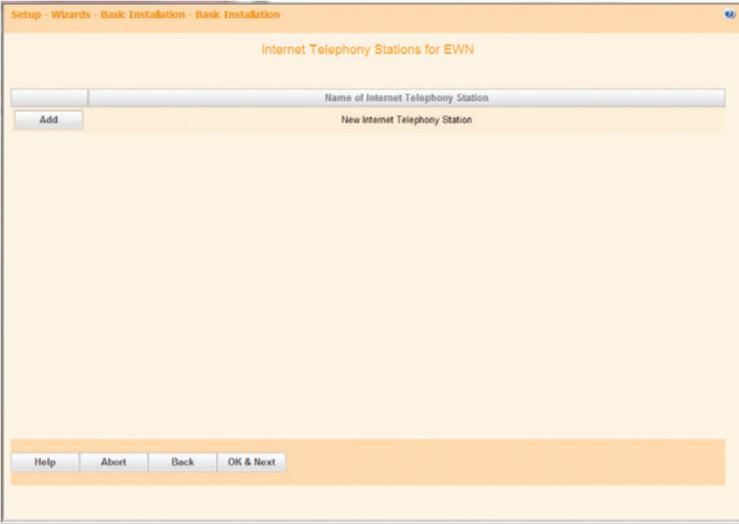
Provider STUN

Help Abort Back **OK & Next** Delete Data

Done Internet | Protected Mode Off 100%

Leave other fields as default and then click the **“OK & Next”** button.

Click the **“Add”** button to add the pilot DID for the Internet Telephony Station.



Add one Internet Telephony Station and a phone number for the EWN provider for now. We will add all the stations and phone numbers we need for the voice gateway later in Export mode.

In the **“Internet Telephony Station for EWN”** section, enter the username (the pilot DID of 6316769659 in this example to match the Optimum Business SIP Trunk Adaptor configuration) in the **“Internet telephony station”** field.

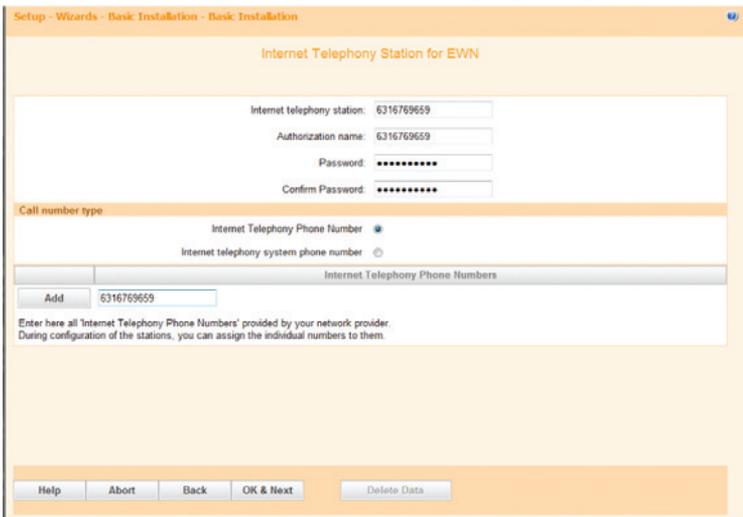
Enter an authorization name (if needed) in the **“Authorization name”** field.

Enter the same password in both the **“Password”** field and the **“Confirm Password”** field. **Note:** The authentication info is not needed for static IP mode.

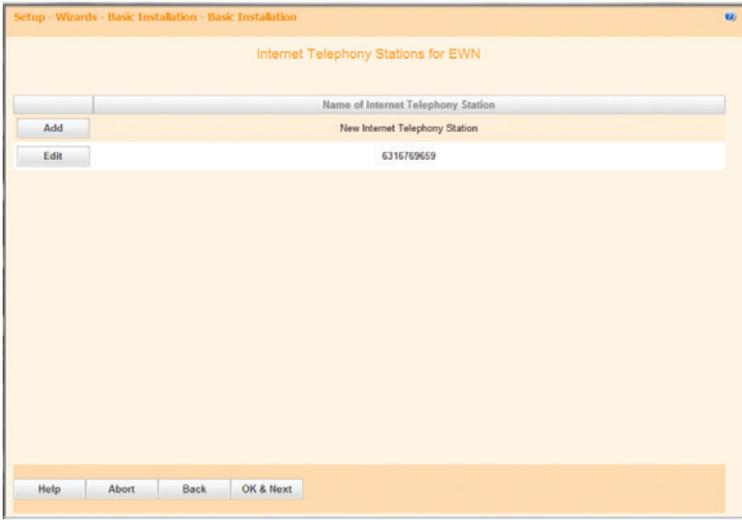
In the **“Call number type”** section, select the **“Internet Telephony Phone Number”** radio button.

In the **“Internet Telephony Phone Numbers”** section, enter the pilot DID and click the **“Add”** button.

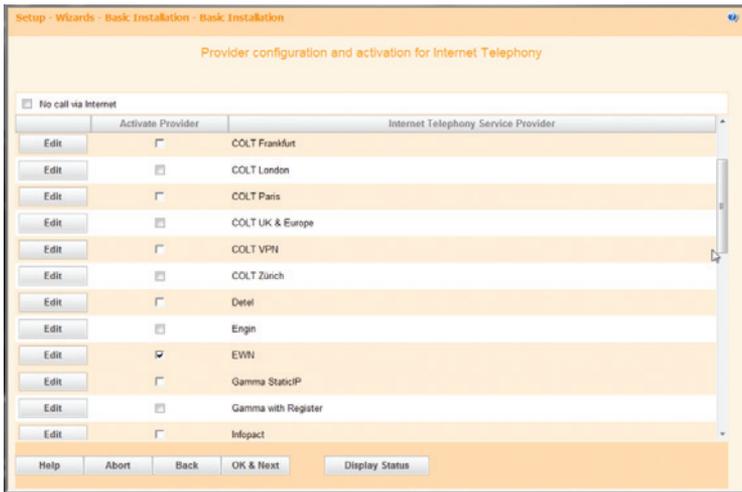
Click the **“OK & Next”** button.



Click the **“OK & Next”** button.

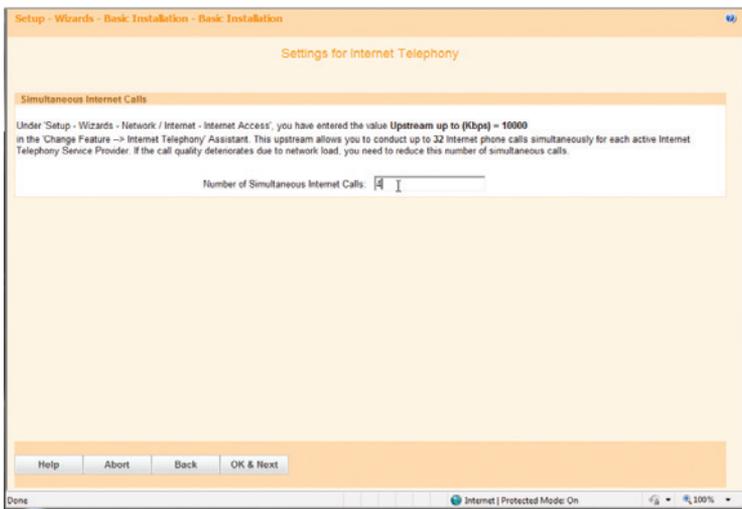


Make sure the **“Activate Provider”** checkbox next to the newly created Internet Telephony Service Provider of **“EWN”** is checked and then click the **“OK & Next”** button.

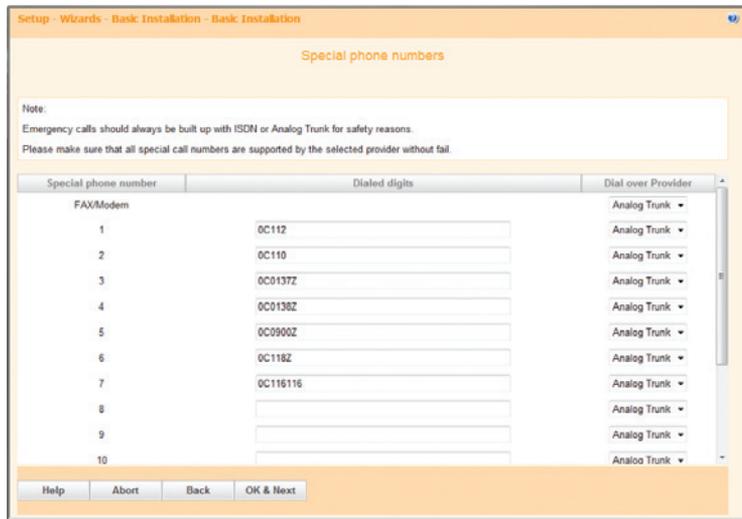


Enter the maximum number of simultaneous calls for the SIP trunk service in the **“Number of Simultaneous Internet Calls”** field, **“4”** in the example.

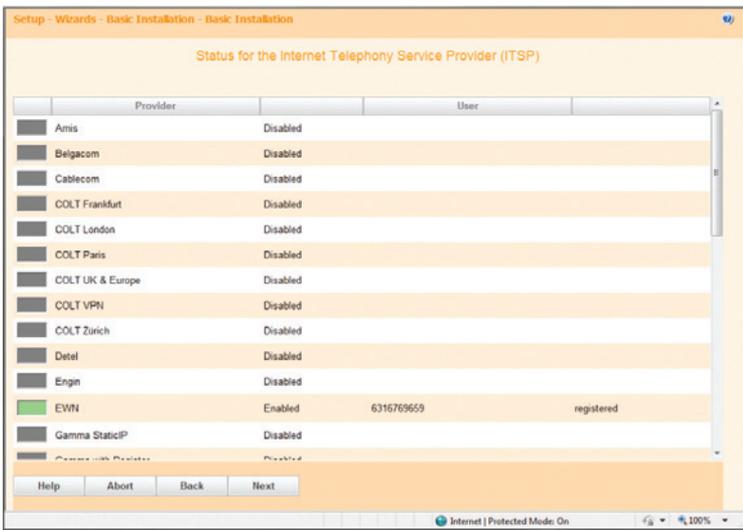
Note: The PBX does not require SIP trunk license and the default for this field is 32.



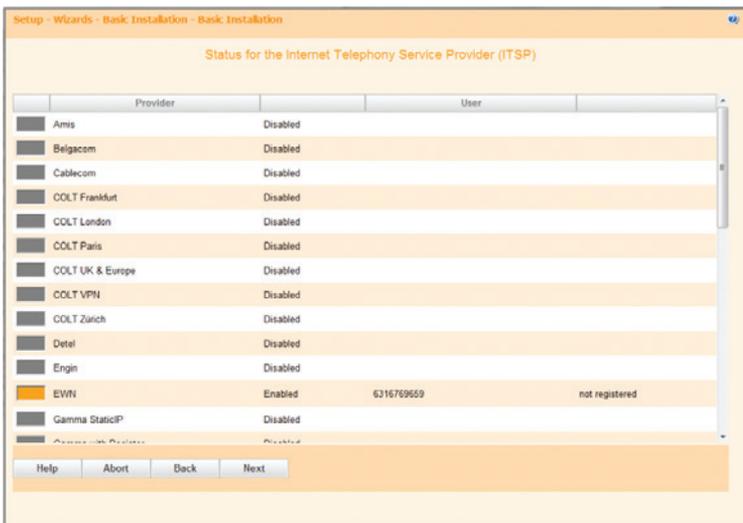
Click the **“OK & Next”** button (we are not using the ISDN/analog trunks).



If the PBX has been set up for static IP mode, you will see a green light for the EWN provider entry. The DID would also be considered “registered” (trusted) although there has been no SIP registration packet sent to the Optimum Business SIP Trunk Adaptor. Click the **“Next”** button to continue.



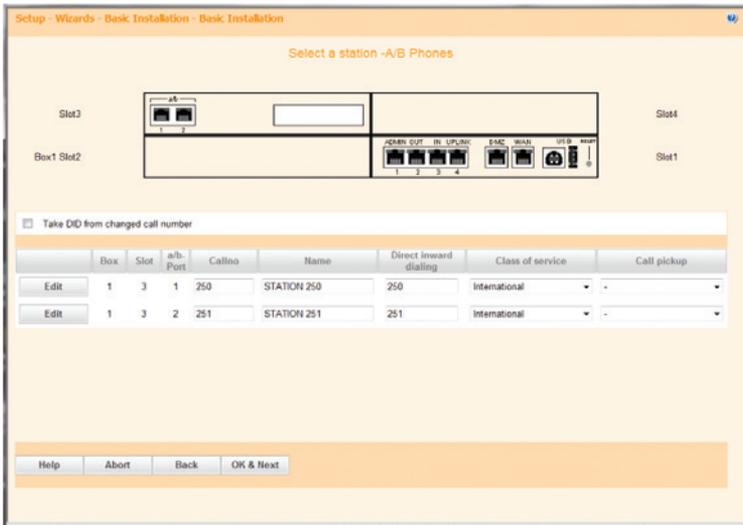
If the PBX has been set up for SIP registration mode, you will see an orange light for the EWN provider entry. The DID would also be marked as “not registered”. Click the **“Next”** button to continue.



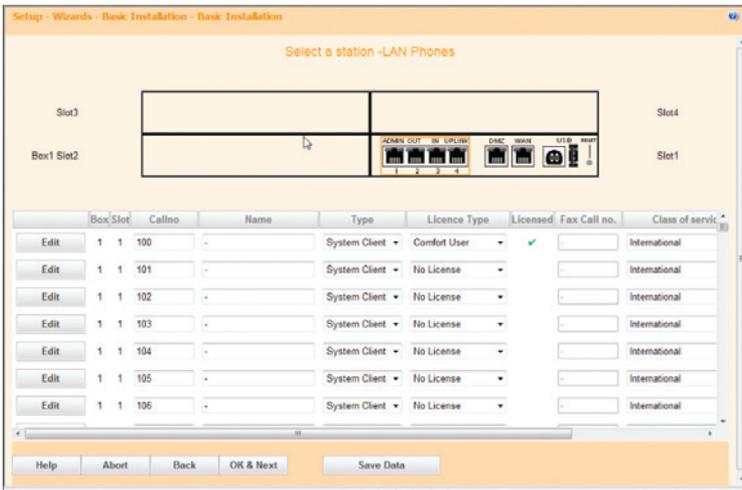
Click the **“OK & Next”** button.



Click the **“OK & Next”** button (we are not using analog phones).



For each DID assigned for SIP trunk service, map it to an internal extension (under the **“Callno”** column) by clicking the **“Edit”** button next to the extension.

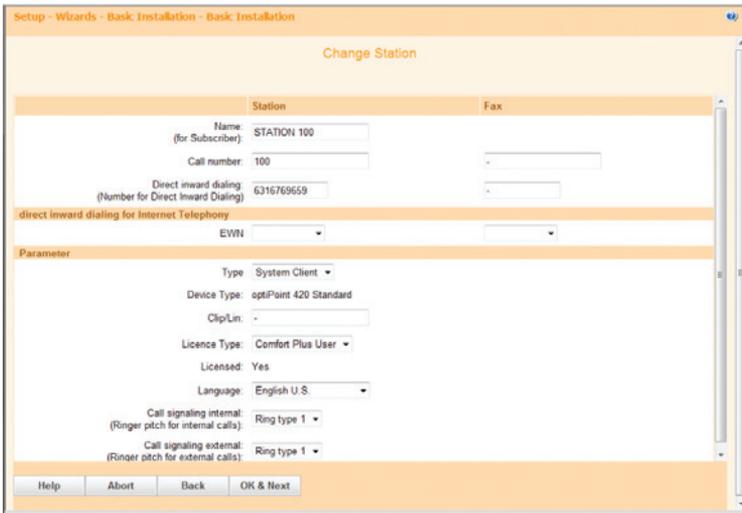


Enter a name in the **“Name”** field.

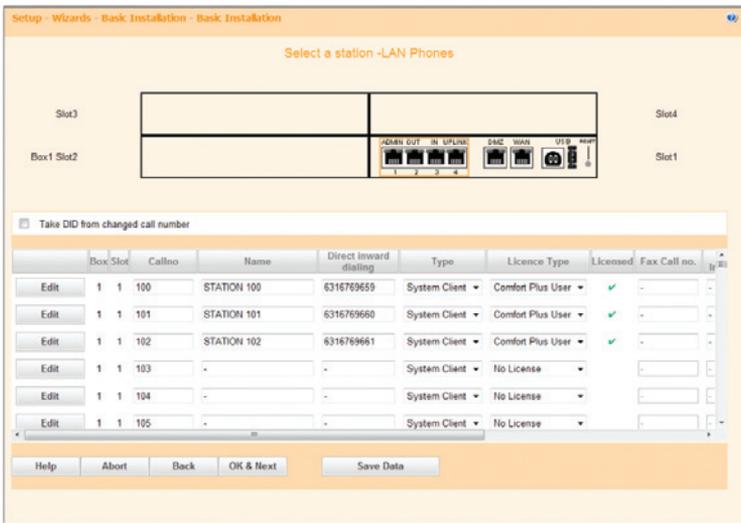
Enter the DID assigned for this extension in the **“Direct inward dialing”** field.

Select **“Comfort Plus User”** from the drop-down list of the **“Licence Type”** field.

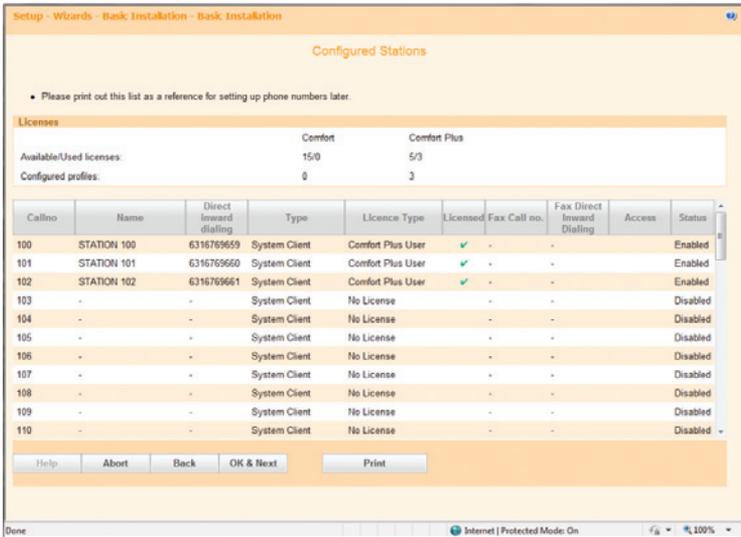
Leave other fields as default and click the **“OK & Next”** button.



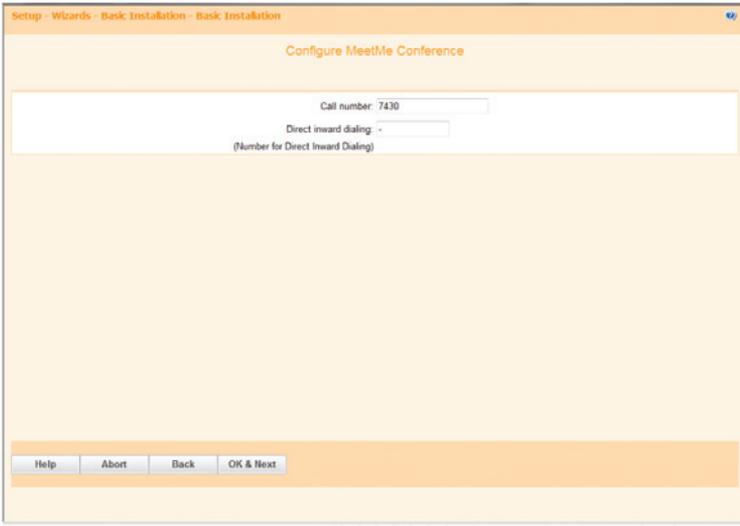
Click the **“OK & Next”** button after you are done with assigning DIDs to the internal extensions.



Click the “**OK & Next**” button.

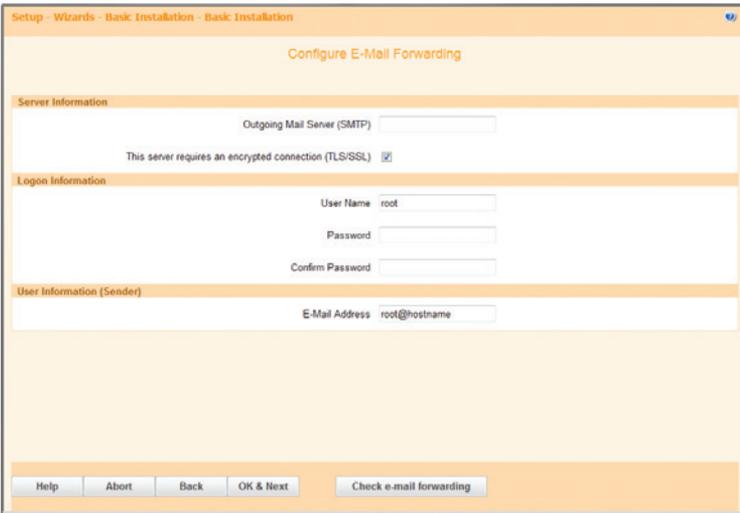


Leave all fields as default and click the “OK & Next” button.



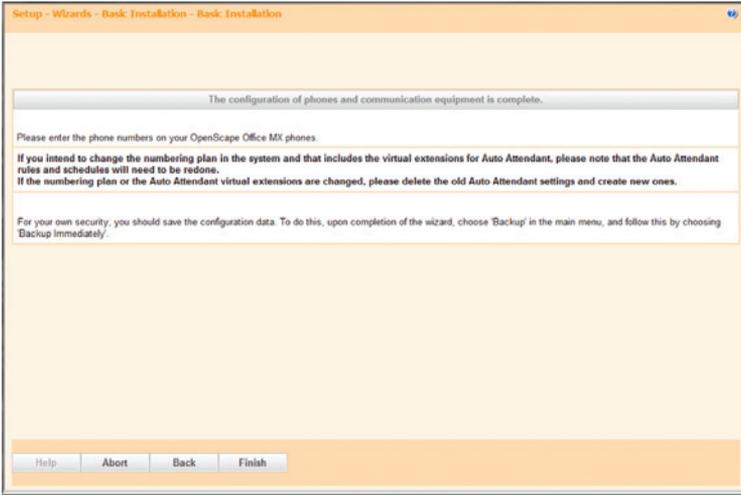
The screenshot shows a web-based configuration wizard titled "Configure MeetMe Conference". The breadcrumb trail at the top reads "Setup - Wizards - Bask Installation - Bask Installation". The main content area contains two input fields: "Call number: 7430" and "Direct inward dialing: -". Below the second field is the text "(Number for Direct Inward Dialing)". At the bottom of the screen, there is a navigation bar with four buttons: "Help", "Abort", "Back", and "OK & Next".

Make sure the “**This server requires an encrypted connection (TLS/SSL)**” checkbox is checked, leave other fields as default and then click the “OK & Next” button.



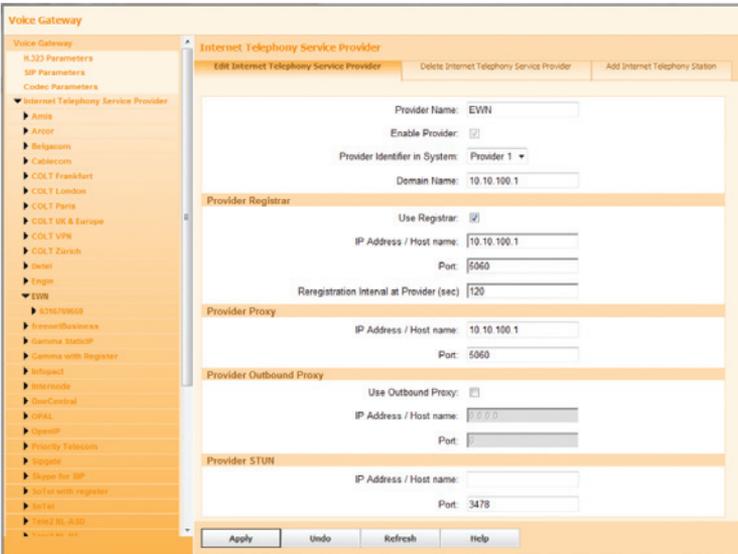
The screenshot shows a web-based configuration wizard titled "Configure E-Mail Forwarding". The breadcrumb trail at the top reads "Setup - Wizards - Bask Installation - Bask Installation". The screen is divided into three sections: "Server Information", "Logon Information", and "User Information (Sender)".
- In the "Server Information" section, there is an "Outgoing Mail Server (SMTP)" input field and a checked checkbox labeled "This server requires an encrypted connection (TLS/SSL)".
- In the "Logon Information" section, there are three input fields: "User Name" (containing "root"), "Password", and "Confirm Password".
- In the "User Information (Sender)" section, there is an "E-Mail Address" input field (containing "root@hostname").
At the bottom of the screen, there is a navigation bar with five buttons: "Help", "Abort", "Back", "OK & Next", and "Check e-mail forwarding".

Click the **“Finish”** button.



Navigate to **“Expert mode”** > **“Telephony Server”** > **“Voice Gateway”** to add Internet Telephony Stations and phone numbers for service provider EWN. A new window showing the Voice Gateway screen will pop up. Click **“Internet Telephony Service Provider”** and then click **“EWN”**. **Important:** The configuration in this section may look redundant, but it is required for getting the SIP trunk service to work.

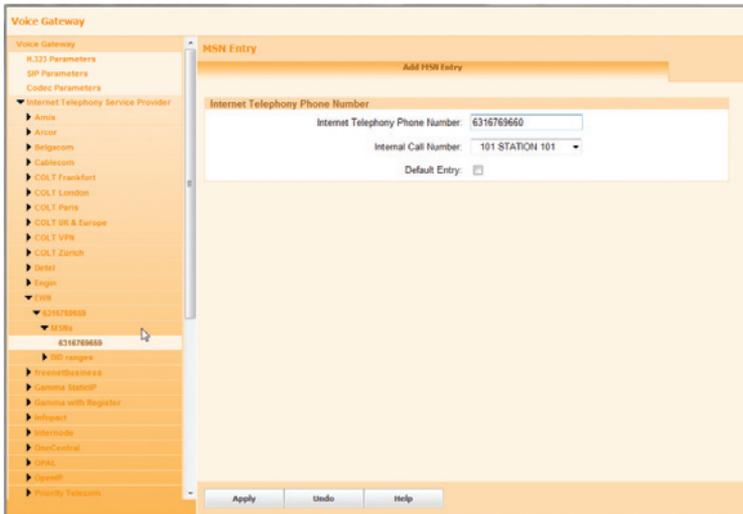
To configure the provider EWN for SIP registration mode, follow these steps:



Click the **Internet Telephony Station** under EWN (pilot DID of “6316769659” in this example) and then click **“MSNs”** to add another DID and map the DID to an extension. Enter a DID (“6316769660” in this example) in the **“Internet Telephony Phone Number”** field. **Note:** The phone number “6316769659” has already been added by the Wizard.

Select next available internal extension from the drop-down list of the **“Internal Call Number”** field.

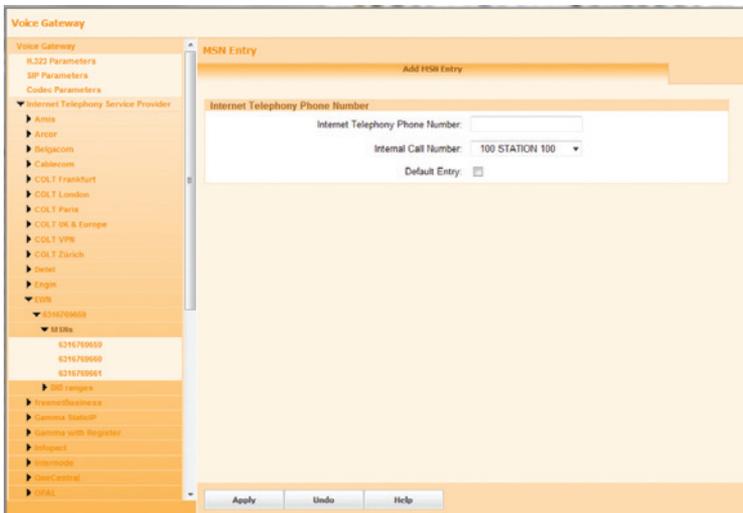
Leave other fields as default and then click the **“Apply”** button.



Repeat step “1” above to add all the DIDs and then click the **“Apply”** button.

Note: Each DID in this configuration will use the pilot DID as its caller ID when placing outbound calls to the Optimum Business SIP Trunk Adaptor.

Click the Windows **“x”** (close) button (above the Voice Gateway screen) to close the window. **Note:** You may need to restart the PBX (“Service Center”>“Restart/Reload” > “Restart”) in order to get the PBX to send SIP registration packet to the Optimum Business SIP Trunk Adaptor.



To configure the provider EWN for static IP, follow the steps below:

Voice Gateway

Internet Telephony Service Provider

Edit Internet Telephony Service Provider Delete Internet Telephony Service Provider Add Internet Telephony Station

Provider Name:

Enable Provider:

Provider Identifier in System:

Domain Name:

Provider Registrar

Use Registrar:

IP Address / Host name:

Port:

Registration Interval at Provider (sec):

Provider Proxy

IP Address / Host name:

Port:

Provider Outbound Proxy

Use Outbound Proxy:

IP Address / Host name:

Port:

Provider STUN

Click the **“Add Internet Telephony Station”** button to another station under EWN. Enter another DID (“6316769660” in this example).

Leave other fields as default and then click the **“Apply”** button.

Voice Gateway

Internet Telephony Service Provider

Edit Internet Telephony Service Provider Delete Internet Telephony Service Provider **Add Internet Telephony Station**

Internet telephony station:

Authorization name:

Password:

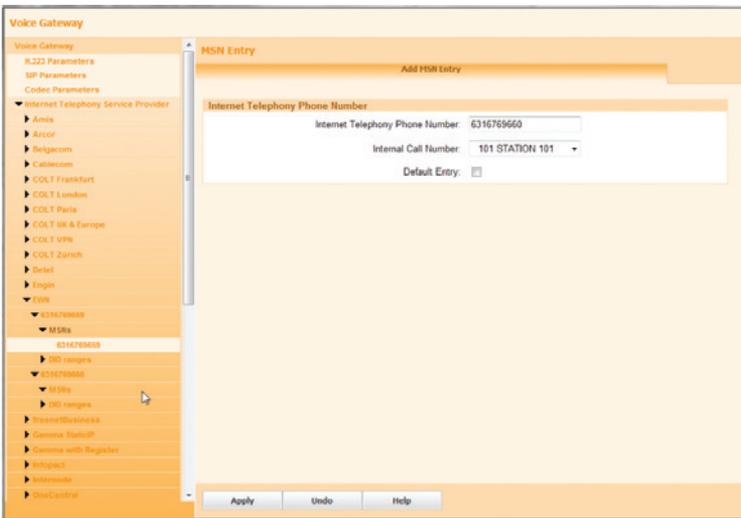
Confirm Password:

Click the **second Internet Telephony Station** and then click **“MSNs”** to add a phone number for it. **Note:** The phone number of “6316769659” has already been added for the first Internet Telephony Station.

Enter a DID (“6316769660” in this example) in the **“Internet Telephony Phone Number”** field.

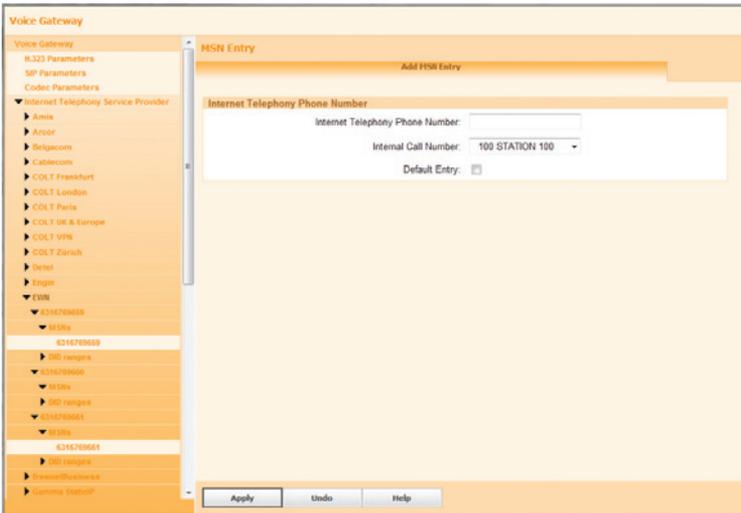
Select next available internal extension from the drop-down list of the **“Internal Call Number”** field.

Leave other fields as default and then click the **“Apply”** button.



Repeat step “1” and “2” above to add all the stations (one for each DID) and then click the “Apply” button. **Note:** Each DID in this configuration will be used as caller ID when placing outbound calls to the Optimum Business SIP Trunk Adaptor.

Click the Windows “x” (close) button (above the Voice Gateway screen) to close the window.



Navigate to “**Expert mode**”>“**Telephony Server**”>“**Voice Gateway**” to configure codec parameters.

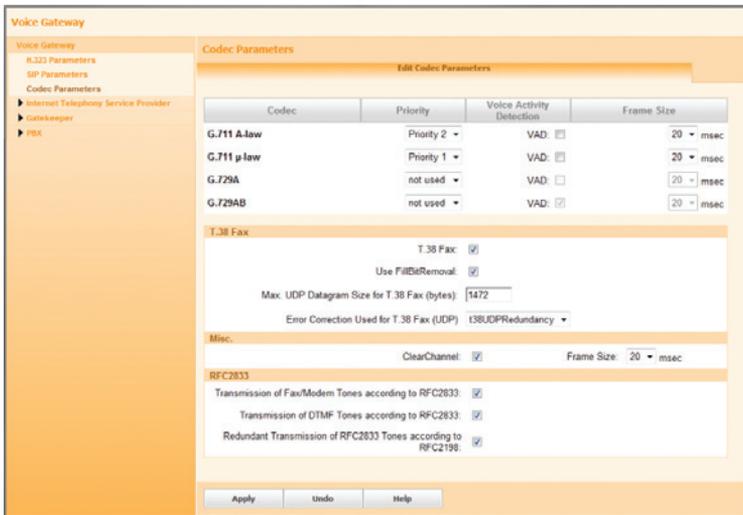
A new window showing the Voice Gateway screen will pop up. Select “**Codec Parameters**”.

For the **G.711 Mu-law** codec, select “**Priority 1**” in the “**Priority**” field, uncheck the “**VAD**” checkbox and select “**20**” msec in the “**Frame Size**” field.

For the **G.711 A-law** codec, select “**Priority 2**” in the “**Priority**” field, uncheck the “**VAD**” checkbox and select “**20**” msec in the “**Frame Size**” field.

For the **G.729A** and **G.729AB** codec, select “**not used**” in the “**Priority**” field. Check the “**Transmission of DTMF Tones according to RFC-2833**” checkbox. Leave other fields as default and then click the “**Apply**” button.

Click the Windows “**x**” (close) button (above the Voice Gateway screen) to close the window.



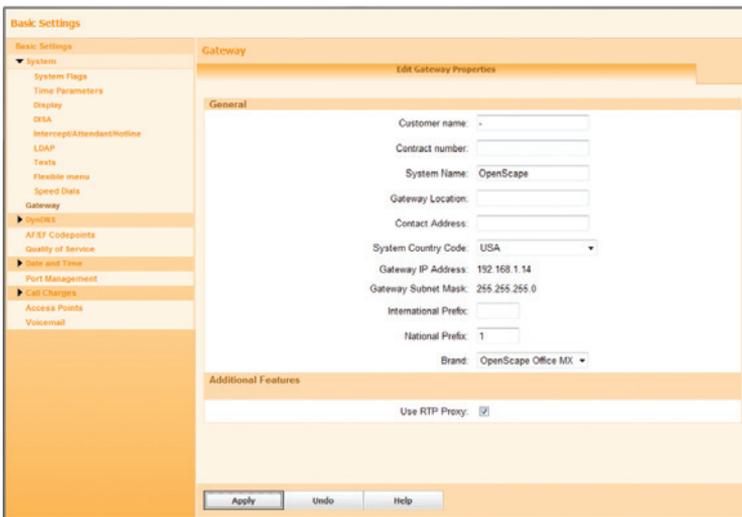
Navigate to “**Expert mode**”>“**Telephony Server**”>“**Basic Settings**” to configure gateway parameters.

A new window showing the Basic Settings screen will pop up. Click “**System**” and then click “**Gateway**”.

Make sure the “**Use RTP Proxy**” checkbox is checked.

Leave other fields as default and click the “**Apply**” button.

Click the Windows “**x**” (close) button (above the Basic Settings screen) to close the window.



Navigate to **“Expert mode”>“Telephony Server”>“Routing”** to find out the dial-out access code for SIP trunk (via “EWN” service provider). The dial-out access code is “80” and it is automatically set up by the PBX when “EWN” was added as a SIP Trunk service provider.

A new window showing the Routing screen will pop up. Select **“LCR”** (Least Cost Routing) and then select **“Routing table”**. We can find out that the PBX has used Routing Table 4 (according to this example) to set up the dial rules for “EWN”.



Select **“Dial Plan”**. We can find out that Dial Plan 19 and 20 are set up for Routing Table 4 with dial-out access code of “80”. For example, to call (408) 351-7255, you can dial “80 408 351 7255” (Dial Plan 19) or “80 1 408 351 7255” (Dial Plan 20).

Navigate to “**Expert mode**”>“**Station**” to create an Auto-Attendant. A new window showing a Station screen will pop up. Click “**Station**”, click “**Application-Suite**”, click “**Auto-Attendant**” and click on the “**815 7410**” entry (“815 7410” is the first available entry in the example) to configure an Auto-Attendant.

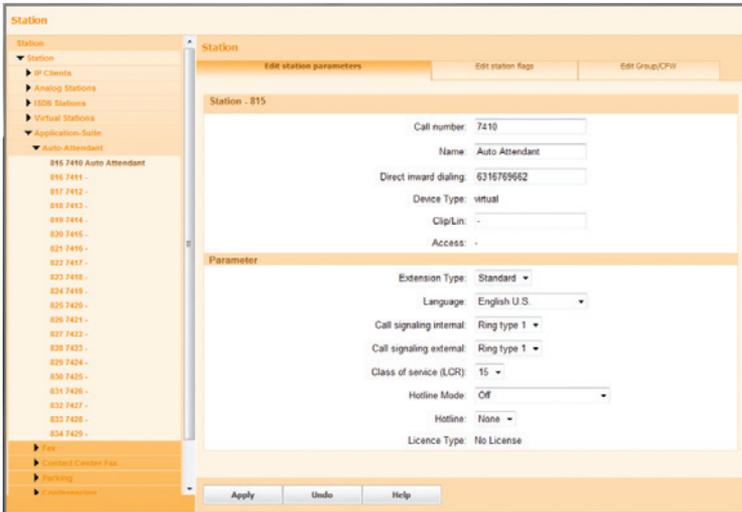
Note: The Auto-Attendant supports only RFC-2833 DTMF.

Enter a name in the “**Auto Attendant**” field.

Enter the DID assigned for the Auto-Attendant in the “**Direct inward dialing**” field.

Leave other fields as default and then click the “**Apply**” button. Once the Auto-Attendant is configured, it will answer external calls to the DID (“631-676-9662” in this example) or internal calls to “7410”.

Click the Windows “**x**” (close) button (above the Basic Settings screen) to close the window.



Navigate to “**Expert mode**”>“**Telephony Server**”>“**Basic Settings**” to configure the Auto-Attendant to route calls for operator to an extension. A new window showing the Basic Settings screen will pop up. Select “**System**” and then select “**Intercept/Attendant/Hotline**”.

Select an extension by name from the drop-down list of the “**Day**” field and the “**Night**” field (extension 100 in this example).

Check the “**on RNA**” checkbox.

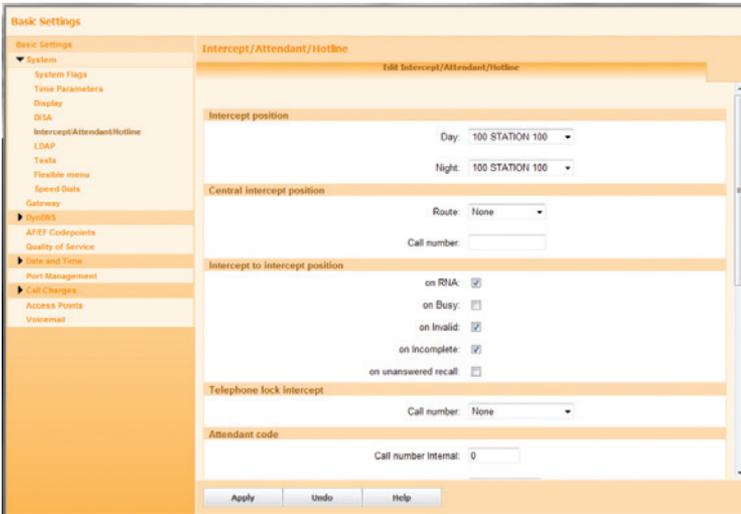
Check the “**on Invalid**” checkbox.

Check the “**on Incomplete**” checkbox.

Enter “**0**” in the “**Call number Internal**” field.

Leave other fields as default and click the “**Apply**” button.

Click the Windows “**x**” (close) button (above the Basic Settings screen) to close the window.



Navigate to **“Expert mode”>“Telephony Server”>“Voice Gateway”** to add the DID to the SIP trunk service.

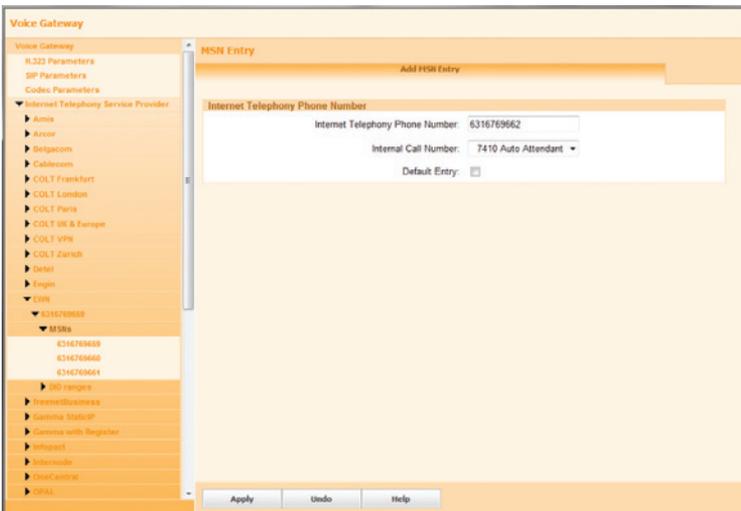
A new window showing the Voice Gateway screen will pop up. For SIP registration now, follow the previous example and navigate to **“Internet Telephony Service Provider”>“EWN”>“6316769659”>“MSNs”**.

Enter the DID assigned to the Auto-Attendant in the **“Internet Telephony Phone Number”** field.

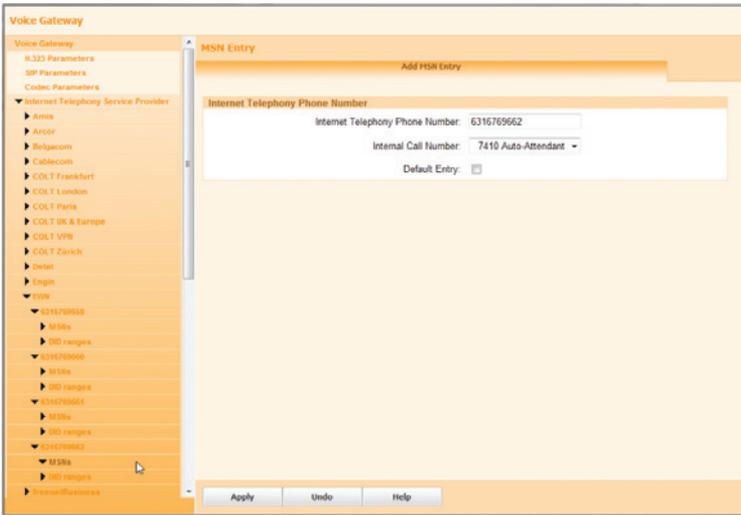
Select **“7410 Auto Attendant”** from the drop-down list for the **“Internal Call Number”** field.

Leave other fields as default and then click the **“Apply”** button.

Click the Windows **“x”** (close) button (above the Basic Settings screen) to close the window.



If the PBX has been set up for static IP mode, follow the previous example and navigate to **“Internet Telephony Service Provider”>“EWN”>“Add Internet Telephony Station”** to add another Internet Telephony Station and a phone number for the Auto-Attendant.



Navigate to **“Expert mode”>“Telephony Server”>“Basic Settings”** to enable the system flags needed for supporting conference.

A new window showing the Basic Settings screen will pop up.

Click **“System Flags”**.

Check the **“Through-connection for external FWD on”** checkbox.

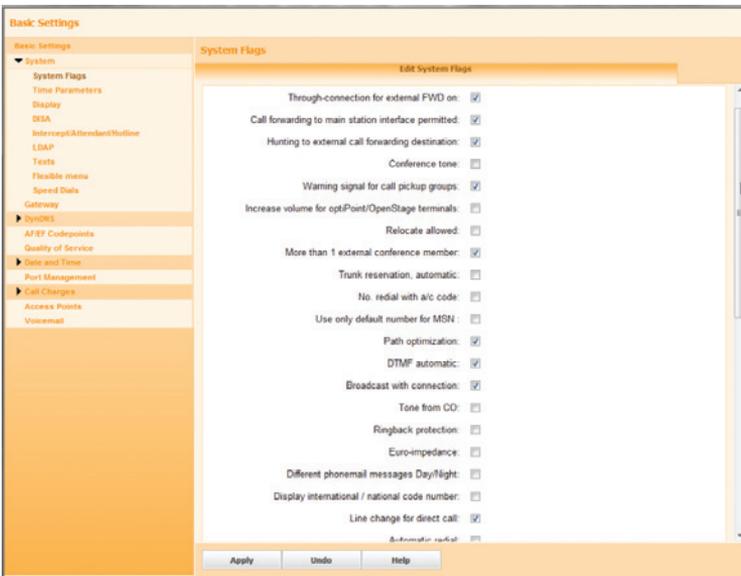
Check the **“Hunting to external call forwarding destination”** checkbox.

Check the **“SIP Prov. to SIP Prov. Transit”** checkbox.

Check the **“External traffic transit”** checkbox.

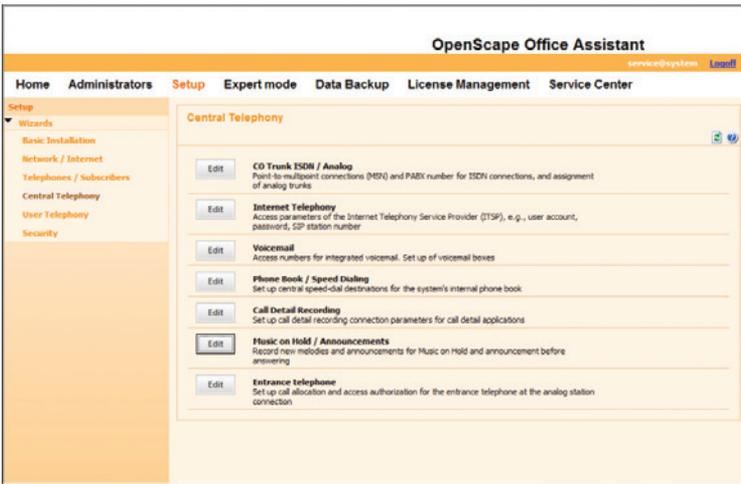
Leave other fields as default and click the **“Apply”** button.

Click the Windows **“x”** (close) button (above the Basic Settings screen) to close the window.

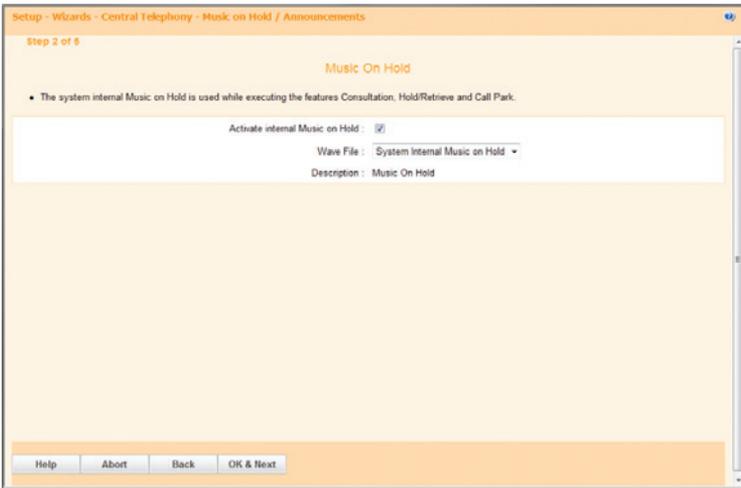


Navigate to “**Setup**”>“**Wizards**”>“**Central Telephony**” to configure MOH (music on hold).

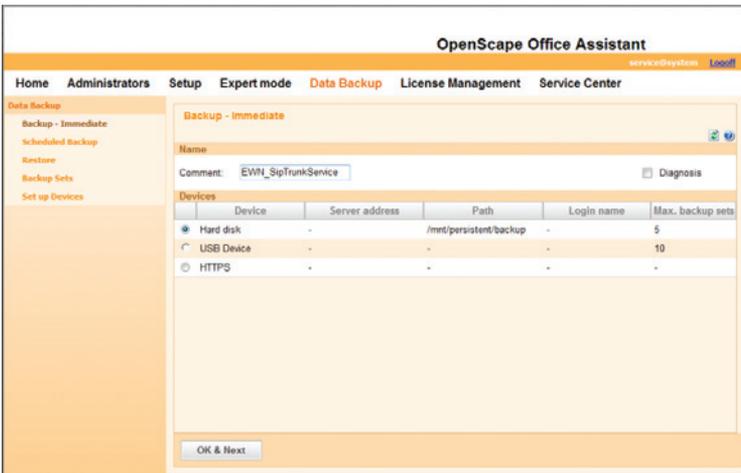
Click the “**Edit**” button next to “**Music on Hold/Announcements**”.



Click the “**OK & Next**” button if the next screen is not about Music On Hold. On the screen with a “**Music On Hold**” section, check the “**Activate internal Music on Hold**” checkbox, select “**System Internal Music on Hold**” from the drop-down list of the “**Wave File**” field and then click the “**OK & Next**” button. Click the “**Finish**” button on the last screen.



To back up the current configuration, navigate to **“Data Backup”**>**“Backup - Immediate”**. Enter a comment in the **“Comment”** field, select the **radio button for the device** and then click the **“OK & Next”** button.



Most of the configurations do not require a reset to take. When in doubt, back up the current configuration to a file and restart the PBX.

IMPORTANT: The Cablevision network only supports inband DTMF tones. Some functions of the Siemens OpenScope Office MX PBX only supports out-of-band DTMF tones. In order for the Siemens OpenScope Office MX PBX to operate correctly with the Cablevision network, the Optimum Business SIP Trunk Adaptor must be configured to convert out-of-band DTMF tones sent by the Siemens OpenScope Office MX PBX to inband DTMF tones and vice versa. To enable this conversion, log into the Optimum Business SIP Trunk Adaptor using the login and password specified in the Optimum Business SIP Trunk Adaptor Set-up Guide. On the **SIP Trunk Configuration** page, you **must** check the **Convert Inband DTMF** checkbox, and click the **Submit** button to update this setting.

The screenshot shows the Optimum Business SIP Trunk Configuration page. On the left is a 'Configuration Menu' with links for Customer, LAN Settings, SIP Trunk Configuration, Diagnostics, and System. The main content area is titled 'SIP Trunk Configuration' and includes a 'Help' link. A dropdown menu for 'Select your PBX:' is set to 'Siemens OpenScope Office MX'. Two radio buttons are present: 'Passive connection using the local, private IP address of the PBX' (selected) and 'Active connection using registration'. The 'Passive' option has a 'PBX Address' field with '192.168.1.10'. The 'Active' option has 'User Id' and 'Password' fields. A 'Convert Inband DTMF:' checkbox is checked. At the bottom of the configuration section are 'Submit' and 'Reset' buttons. Below this is a 'Status:' section with a 'Trunk Status:' field showing 'Registered' and a 'DID's' list box containing '6316769522 (Pilot number)', '6316769523', '6316769524', and '6316769525'.