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Optimum Business Trunking and the Zultys MX250 IP PBX Configuration Guide

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Table of Contents

1 Overview

The purpose of this configuration guide is to describe the steps needed to configure the Zultys MX250 IP PBX for proper operation Optimum Business Sip Trunking.

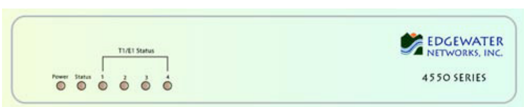
2 SIP Trunk Adaptor Set-up Instructions

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

Step 1:

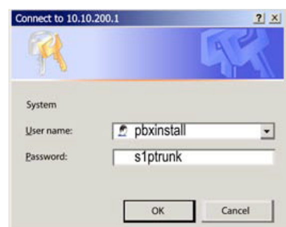
Log on to the Optimum Business SIP Trunk Adaptor

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: s1ptrunk

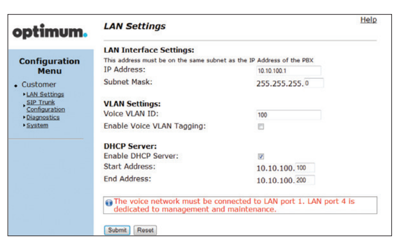


Step 2:

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.



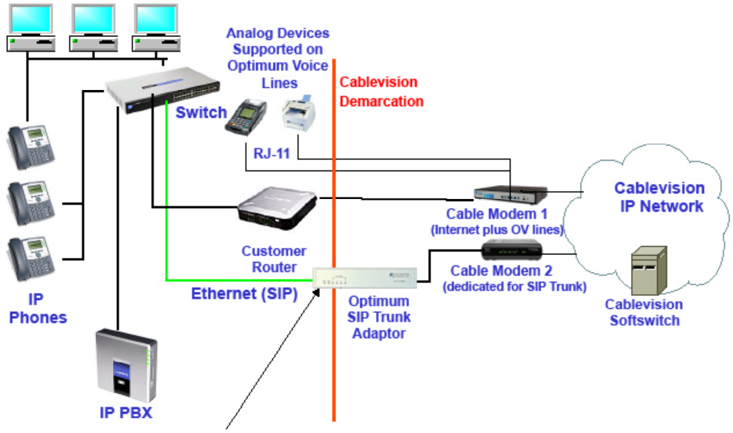
2. Optional: Specify a VLAN for your voice traffic. Click the ‘Enable Voice VLAN Tagging’ check box. The default VLAN ID is 100.

   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.

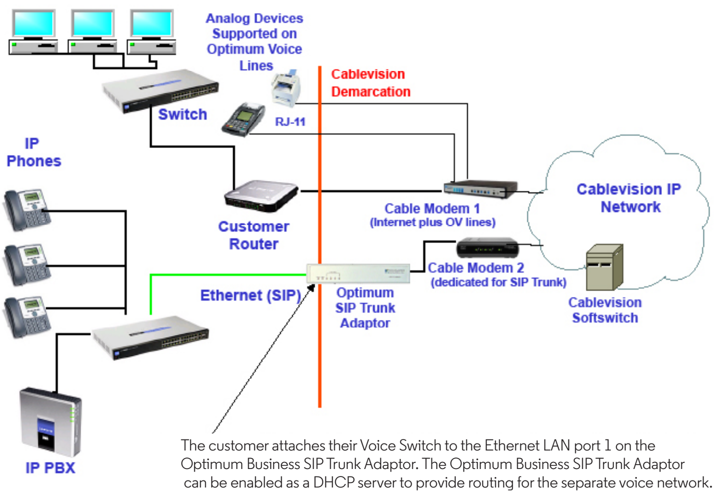
4. Click 'Submit'.

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.

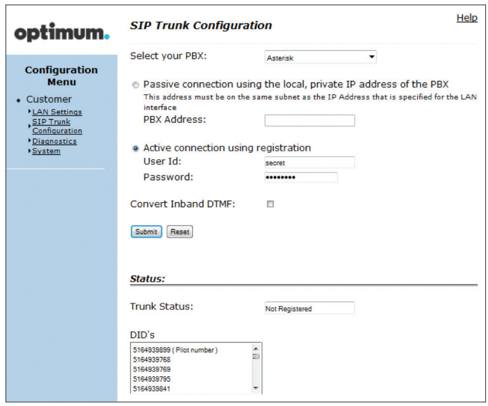
Diagram 2 SIP Trunk Adaptor for IP-PBXs Example: Separate Voice and Data Networks Configuration



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.

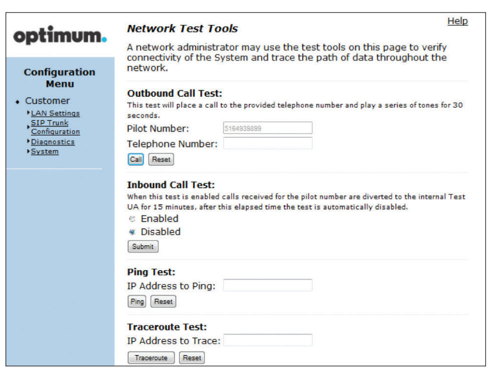
3. The Cablevision network only supports Inband DTMF. Click on the check box next to "Convert Inband DTMF" if you cannot configure your IP PBX to send out Inband DTMF. The DTMF tone duration generated by the phones and/or PBX may need to be increased from their default setting. Some phones and/or PBX 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

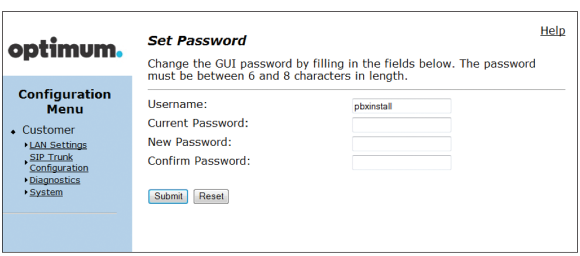
Field	Description
Outbound Call Test TelephoneNumber	Specifies an outside phone number to which an outbound call will be initiated. The pilot telephone number of the SIP Trunk will be prepopulated.
Pilot Number	Displays the provisioned pilot number, which is used for outbound and inbound call tests.
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.

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 Zultys MX250 product documentation for more information on other advanced PBX features.

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

IP PBX Information

Manufacturer:	Zultys
Model:	MX250
Software Version:	8.0.7
Does the PBX send SIP Registration messages (Yes/No)?	Yes

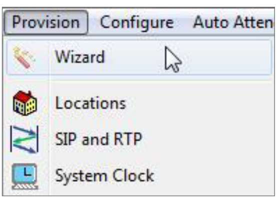
Important:

The Zultys PBX only supports RFC2833 out-of-band DTMF tones. The Cablevision network only supports Inband DTMF tones. In order for the Zultys PBX to operate correctly with the Cablevision network, the Optimum SIP Trunk Adaptor must be configured to convert out-of-band DTMF tones sent by the PBX to inband DTMF tones. To enable this conversion, log into the Optimum Business Sip Trunk Adaptor using the login and password specified in the Optimum Business Sip Trunk 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. This is described in step 3 of the Optimum SIP Trunk Set-up Guide.

The DTMF tones duration generated by the phones and/or PBX may need to be increased from their default settings. Some phones/PBXes have a default setting between 180ms and 200ms. The recommended setting is 600ms.

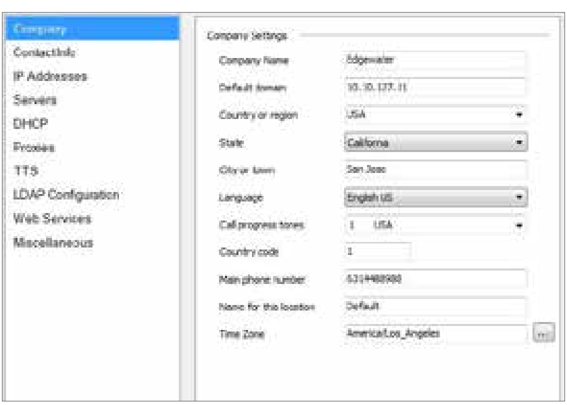
Network Settings

Navigate to **“Provision ▸ Wizard”.**

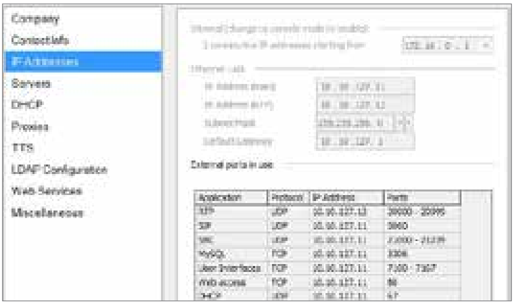


Here is where you configure the initial network settings for the device. The ones you will focus on are: Company, IP Addresses, Servers, DHCP, and Miscellaneous. The rest are relative to your setup.

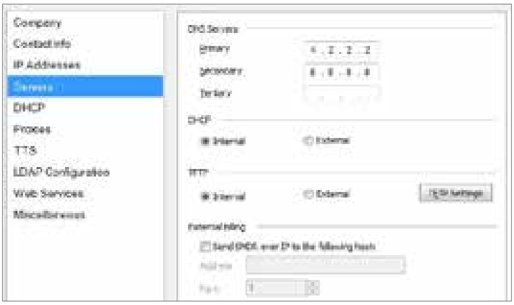
Starting with **“Company”**, enter your company name next to **“Company Name”.** Enter the IP address of the MX device next to **“Default domain”.** Enter your Pilot DID next to **“Main phone number”** and **“Default”** next to **“Name for this location”.**



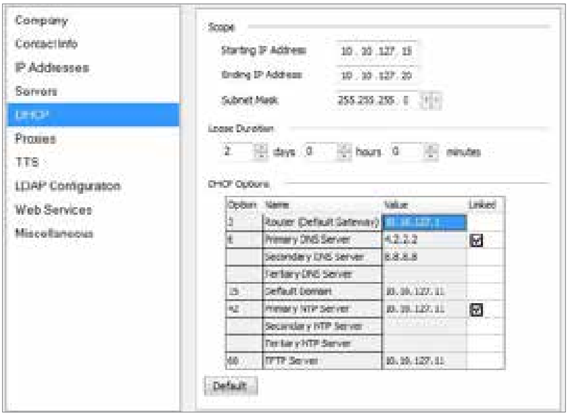
Now go to **"IP Addresses".** Remember you need to use the console port on the device when configuring the IP addresses and System Settings unless the box is being configured for the first time. When in normal mode, the console port and the port on the left are just used as Ethernet 1 and 2 interfaces. Choose an address next to **"2 consecutive IP addresses starting from".** This is required. **"172.16.0.1"** was chosen for our example. Right under is where you will need to enter the addresses you're using. 10.10.127.11 /24 was entered for our (main) field which essentially is the IP of the MX. 10.10.127.12/24 was entered for (RTP) and 10.10.127.1/24 for the Default Gateway which is the IP address of the Optimum Business SIP Trunk Adaptor.



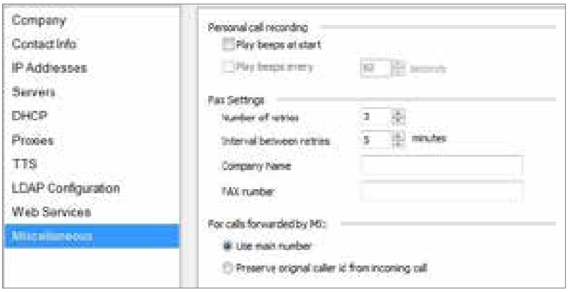
Now go to **"Servers"** and enter your DNS addresses. In the examples, you will be using DHCP for your phones, click the radio button of **"Internal"** under **"DHCP"** and likewise do the same for our firmware and click the radio button of **"Internal"** under **"TFTP".**



Now go to **“DHCP”** and here you will need to enter a starting and ending range for your phones. In our example the range 10.10.127.15 through 20 is used. The mask will remain/24. Make sure on the bottom under **”DHCP Options”** you have 10.10.127.1 next to Option 3 and 10.10.127.11 next to Options 15,42, and 66. The boxes next to Option 6 and 42 should also be checked under **“Linked”.**



Now go to **“Miscellaneous”** and under **“For calls forwarded by MX”** check the radio button of **“Use main number”.** When finished, apply all changes.



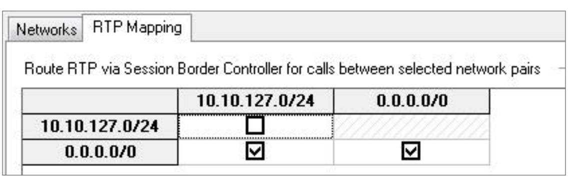
Navigate to **“Provision ▸ SBC”.**

From the top enter RTP port range 2100–2139. Select the radio button of **“Authenticate all traffic from untrusted networks”.** On the bottom right-click anywhere in the empty space and select **Add.**

Enter 10.10.127.0 for the Network Address, 24 under NML which is the network mask length, and 255.255.255.0 under Mask. Check the **“Trusted Network”** box and leave the **“Port Mapping”** box empty. Again right-click anywhere in the empty space and select **“Add”.** Enter 0.0.0.0 under **“Network Address”,** 0 under **“NML”,** and 0.0.0.0 under **“Mask”.** Leave both **“Trusted network”** and **“Port Mapping”** empty.

When done click **Apply**.

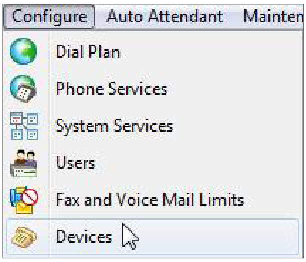
Then, go to the **“RTP Mapping”** tab at the top and check as displayed:

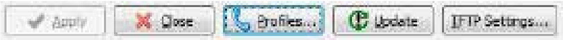


When done click **Apply**.

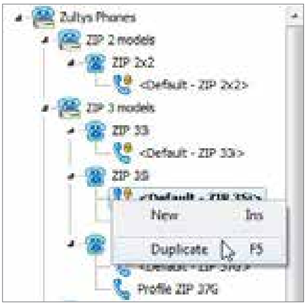
Phone Registration and Assignment

Navigate to **“Configure ▸ Devices”** and when there click on **“Profiles”.**

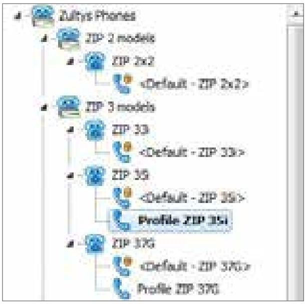


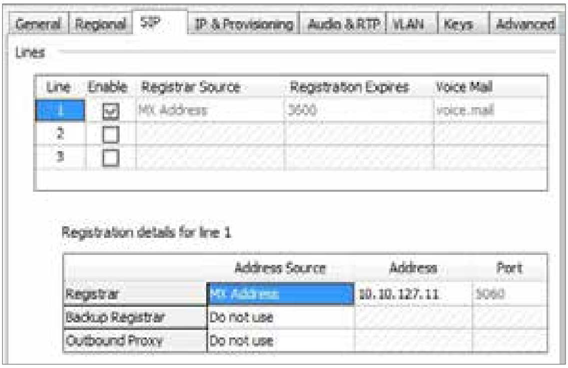


Depending on the phones you have the selection will differ but for this test one 35i phone and one 37G phone were used. For the 35i phone just open first **“ZIP 3 models”** and under **“ZIP 35i”** right-click **“<Default-ZIP 35i>”** and select **“Duplicate”.**



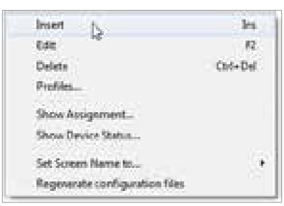
The duplicate was named **"Profile ZIP 35i".** When it is highlighted, on the right you should see several tabs. Go to the **"SIP"** tab and make sure the Registrar on the bottom under **"Registration details for line 1"** is set to **"MX Address"** and that the address is the corresponding MX IP. Make sure to enable Line 1 next to **"Lines"** above.





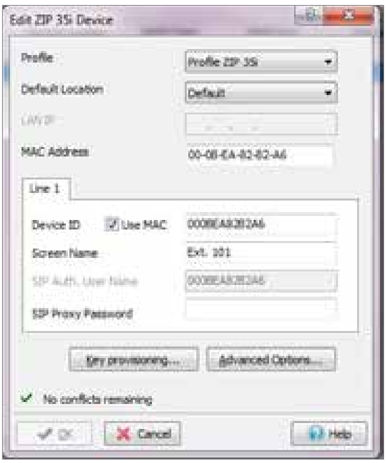
For our purposes mentioned DHCP on the MX is enabled. Under **“IP & Provisioning”** tab this is what is displayed:

Under **“Provisioning”** you will see the protocol as **“DHCP (option 66)”** and **“MX”** as the Server. The other tabs above are relative to your setup. When done click **OK** then **Apply** and repeat this process for your remaining phones. When back on **“Configure ▸ Devices”** right-click anywhere in the empty space and select **“Insert”**.

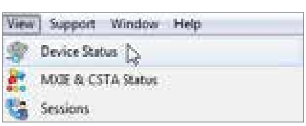


Choose your device and select **“Add single device”** if you have one phone from this particular model or select **“Add multiple devices”** if you have more from this particular model. Click **Next**.

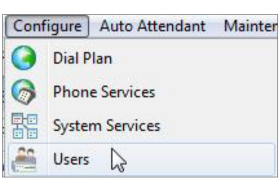
A screen will appear with various fields. Next to **“Profile”** select the name of the duplicate created earlier. Leave **“Default Location”** as Default. Enter the MAC address of the phone. Check **“Use MAC”** next to **“Device ID”** and reenter the MAC address. Next to **“Screen Name”** you may enter the extension of the phone. When done click **OK**.



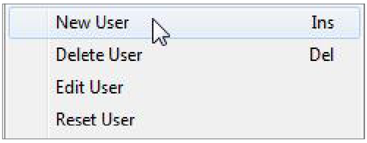
To verify if your device registered correctly or not, navigate to **“View ▸ Device Satus”** and under **“Active”** it should display **“True”.** Otherwise it would say **“False”.**



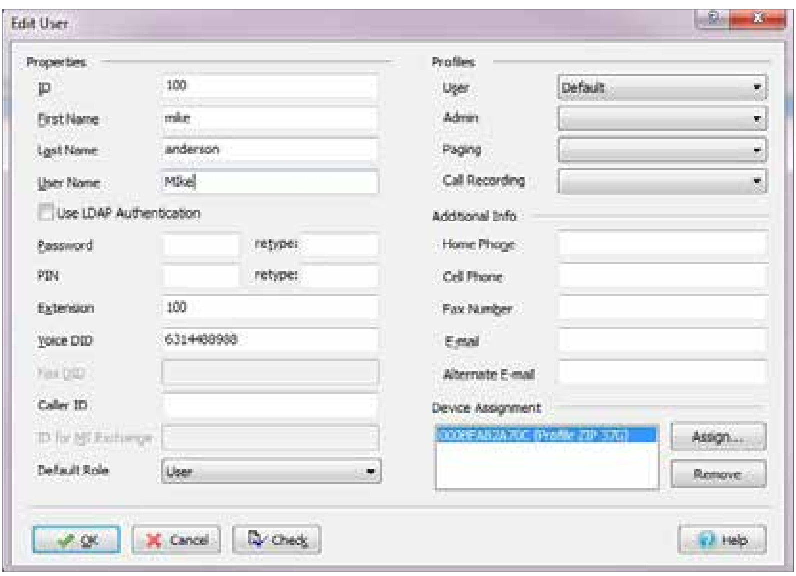
Now navigate to **“Configure ▸ Users”.**



Right-click anywhere in the empty space and select **"New User"**.



This is where you will enter your DIDs for your phones. Next to **"ID"** just enter the extension. The First and Last Name can be anything. You will then enter a User Name for your phone. Enter the extension number next to **"Extension"**. Enter your DID next to **"Voice DID"**. Make sure the Default Role is set to **"User"**. Next to **"User"** above leave as **"Default"**. Where it says **"Device Assignment"** below make sure to select the corresponding profile, again this will be the duplicate that was created earlier. Repeat this process for remaining phones.



Click **OK**, then **Apply**.

Now you will need to manually associate each phone with the appropriate extension. Navigate to **“Configure ▸ Assignment”.** Highlight the extension under **“Users”** from the left and highlight its corresponding phone under **“Devices”** on the left and then click **Assign** on the bottom. Repeat this process for remaining phones.

	First Name	Last Name	Extension
	Tim	Steven	101
	mike	anderson	100

	ID	Profile
	000BEA82A70C	Profile ZIP 37G
	000BEA82B2A6	Profile ZIP 35i

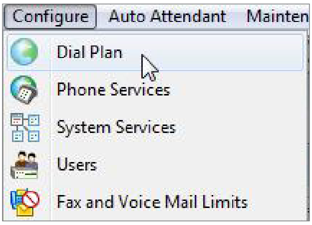
When done click **Apply**.

Now back to **“Configure ▸ Device”** right-click each phone and go to **“Select Screen name to”** then select **“User Extension”.** Click **Update** and simply repeat this process for remaining phones.

When done click **Apply**.

Dial Plan

Navigate to **“Configure ▸ Dial Plan”.**



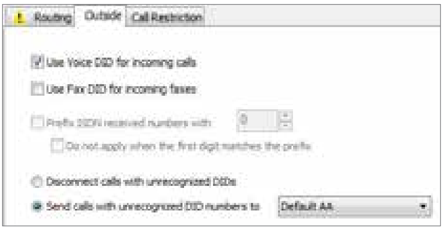
Here is where you will configure what numbers can be dialed. For purposes of this guide, you created a pattern for our internal extensions as well as 10-digit numbers. Right-click anywhere in the empty space and select **“New Pattern”.** Enter the name of your pattern, for **“Source”** select **“Internal”.** For our first pattern you entered **“XXX”** under **“Pattern”** where each X is representative of any digit. You entered three because both of our extensions 100 and 101 consist of 3 digits. Under **“Destination”** choose **“Extensions”** and just reenter the pattern under **“Transformation”.** You then created a 10-digit pattern but this time under **“Destination”** you select **“SIP Server: NewSIP”** to direct call to SIP Server. For outbound calls, you need to choose this option. You can also create patterns for prefixes like 9, 1, and even 91 as displayed below. For the 9-digit, you will need to substitute it for **“D”** under **“Transformation”** which will mean delete. As for the 1 prefix, under **“Transformation”** just substitute it for an **“X”** as shown. You may configure your setup accordingly.

Routing | Outside | Call Restriction

	Name	Source	Pattern	Destination	Transformation	Restricted
1	Internal extensions	Internal	XXX	Extensions	XXX	☐
2	10Digit	Internal	XXXXXXXXXX	SIP Server : NewSIP	XXXXXXXXXX	☐
3	1Plus	Internal	1XXXXXXXXXX	SIP Server : NewSIP	XXXXXXXXXXX	☐
4	1Plus_9Prefix	Internal	91XXXXXXXXXX	SIP Server : NewSIP	DXXXXXXXXXXX	☐
5	10Digit_9Prefix	Internal	9XXXXXXXXXX	SIP Server : NewSIP	DXXXXXXXXXX	☐

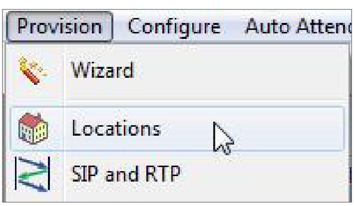
When done click **Apply**.

Under the **“Outside”** tab you checked **“Use Voice DID for incoming calls”** and selected the radio button of **“Send calls with unrecognized DID numbers to”** and have chosen **“Default AA”.**



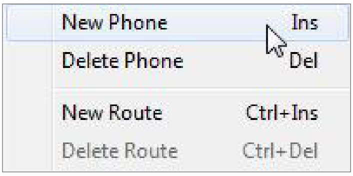
When done click **Apply**.

As for 911 Emergency calls, this will have to be configured separately. Navigate to **“Provision ▸ Locations”.**



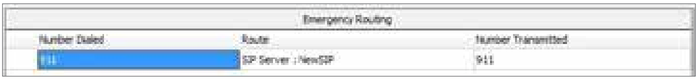
On the left under **“Location Name”** right-click anywhere in the empty space and select **“New”.**

Where it says **"Emergency Routing"** right-click the empty space and select **"New Phone".** Enter 911 under **"Number Dialed",** select **"SIP Server: NewSIP"** under **"Route"** and re-enter 911 under **"Number Transmitted".**

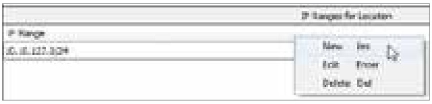


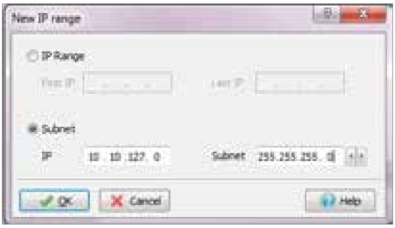
Enter **"Default"** for Location Name and then enter your appropriate time zone.

When done click **Apply**.



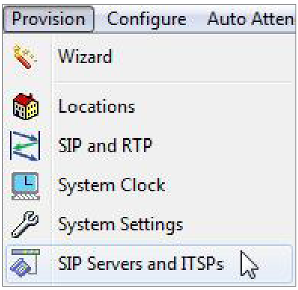
Just below, right-click **"IP Range"** and select **"New".** Select the radio button of **"Subnet"** and enter 10.10.127.0/24 and click **OK**.



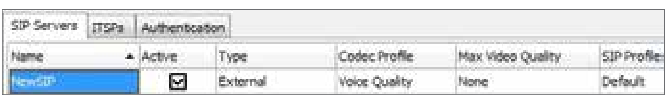


SIP Registration

Navigate to **"Provision ▸ SIP Servers and ITSPs"** as shown.

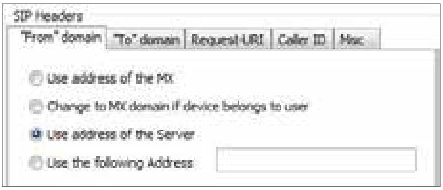


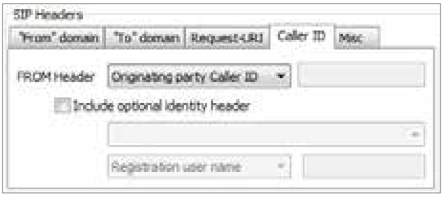
Right-click anywhere in the empty space and select **"Add".** Under name, enter **"NewSIP"** and check **"Active".** Under **"Type"** choose **"External"** and under **"Codec Profile"** choose **"Voice Quality".** Leave **"SIP Profile"** as **"Default".**



On the right under **"Use the following servers"** enter the address of the Optimum Sip Trunk Adaptor followed by port 5060. The address of the Optimum Business SIP Trunk Adaptor was assigned in step 2 of the Optimum Business Set-up Guide. Just under, make sure to check **"Register"** and enter your username followed by the timeout.

Just below where it says **“SIP Headers”** select the **“‘From’ domain”** tab and select the radio button of **“Use address of the Server”.** Then, click on the **“Caller ID”** tab and next to **“FROM Header”** scroll to and select **“Originating party Caller ID”.**





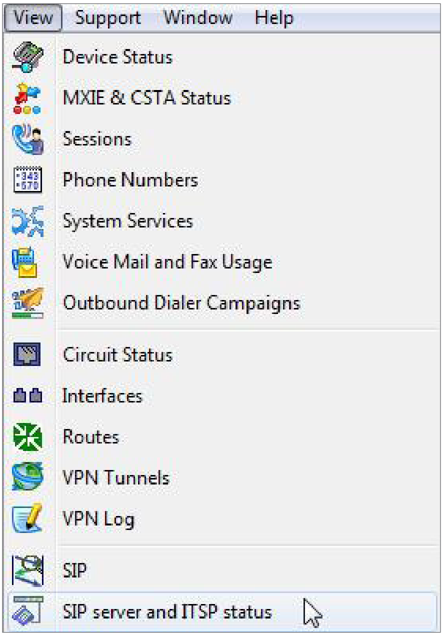
When done click **Apply**.

Now navigate to **“Authentication”.**

Right-click the empty space and select **“Add”.** Here you will enter your SIP credentials with the Optimum Business Sip Trunk Adaptor. Next to **“Realm Domain”** simply enter the address of the Optimum Business Sip Trunk Adaptor. This was the address that was assigned in step 2 of the Optimum Business Sip Trunk Set-up Guide. The user name and password must match what was configured in the Optimum Business Sip Trunk Adaptor in step 3 of the Set-up Guide.

When done, click **OK** then **Apply**.

To verify your registration, navigate to **"View ▸ SIP Server and ITSP status"**.



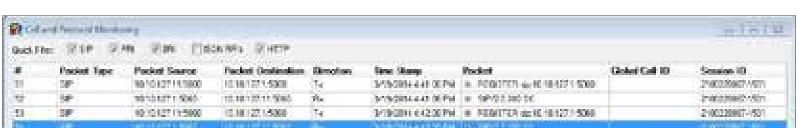
If registered it will show you as displayed below.

Name	SIP Servers	Username	Registered	Outbound Proxy
NewSIP	10.10.127.1	6314488996	Yes	Yes

For Static mode, simply specify the Optimum Business Sip trunk Adaptor's address as the server and uncheck the **"Register"** box as shown:

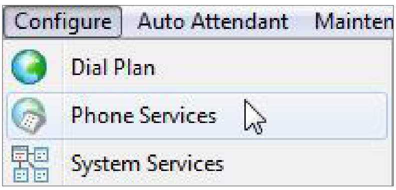
If you desire, you can even monitor the SIP signalling for troubleshooting by navigating to **"View ▸ Call and Protocol Monitor".**

When here simply click **Start** and you should see SIP signaling start flowing.



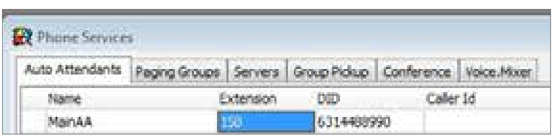
Auto-Attendant

Auto-Attendant requires three steps to configure. First navigate to **“Configure ▸ Phone Services”.**



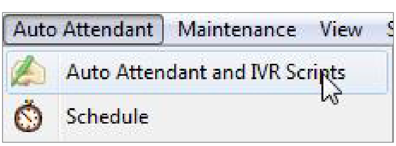
Right-click anywhere in the empty space and select **“New”.**

Here you will be prompted to enter a name for your AA, extension and DID. It was named **“MainAA”,** assigned it to extension 150 and DID 6314488990.



When done click **Apply**.

Now navigate to **“Auto Attendant ▸ Auto Attendant and IVR Scripts”.**

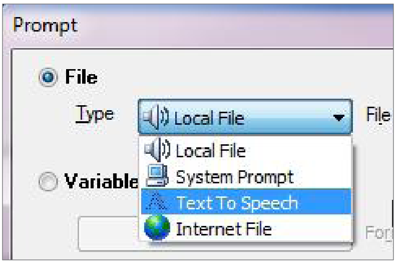


Right-click anywhere in the empty space and select **"New Project".** Enter the name you desire. In this example **"DayAA"** was entered. When done click **OK**.

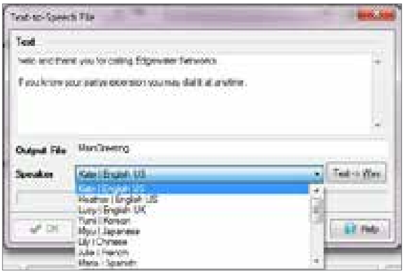
Double-click on the project you created and you will see a new screen.

On the top right of the screen click **"Add".**

Next to the radio button of **"File"**, where it says **"Type"**, scroll to and select **"Text to Speech"**.



Click **"New"** and you will be presented with an empty screen in which you must type in the text you would like the AA to repeat when being called. Then you must name the file next to **"Output File"**. Also you have the option of selecting different AA Speakers from the bottom. When you have added your text, named the file and selected your speaker, click **"Text ▸ Wav"**. Finally click **OK**.

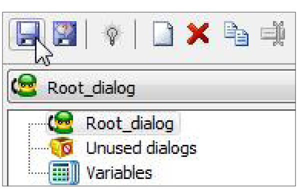


Now, just under **"Actions"** right-click and select **"Insert Action"**. The **"User input"** corresponds to the **"Actions"** field and essentially is telling the AA **"if a user enters x then respond with y"**. If you would like to allow the caller to enter an extension number then you must enter a **"?"** for each dialed digit. Since in our example there are extensions 100 and 101 that are 3 digits, you will need to enter 3 of these as displayed. Once you enter this, you will need to select the Action the AA will take by double-clicking **"Actions"** then selecting from the list your corresponding action from the left.

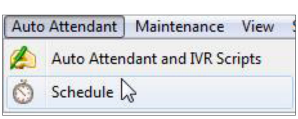
For the extensions, first select **"Transfer"**, then under **"Destination"** select the radio button of **"Variable or Dialog"**. From here scroll to and select **"Root_dialog"**, which should be there by default. Click **OK**.

Possible inputs have been added with their corresponding actions. Below reveals what has been configured in our example. Configure your setup accordingly to meet the needs of your AA.

Finally, when you have configured both the prompt and actions on this page, click the save icon button on top for you, configuration to take effect.



As for the last step, navigate to **"Auto Attendant ▸ Schedule"**.



Click **“New”** on the bottom of the screen. Here is where you implement the schedule of your AA. **“DayAA”** was entered for name and configured our desired schedule. Next to **“Script”** you must enter the Project you created earlier, in our case **“DayAA”.**

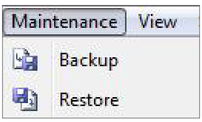
When done click **OK**.

IMPORTANT: The Cablevision network only supports Inband DTMF tones. The Zultys PBX only supports RFC2833 out-of-band DTMF tones. In order for the Zultys PBX to operate correctly with the Cablevision network, the Optimum SIP Trunk Adaptor must be configured to convert out-of-band DTMF tones sent by the PBX to inband DTMF tones. To enable this conversion, log into the Optimum Business Sip Trunk Adaptor using the login and password specified in the Optimum Business 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. This is described in step 3 of the Optimum SIP Trunk Set-up Guide.

The DTMF tones duration generated by the phones and or PBX may need to be increased from their default settings. Some phones/PBXes have a default setting between 180ms and 200ms. The recommended setting is 600ms.

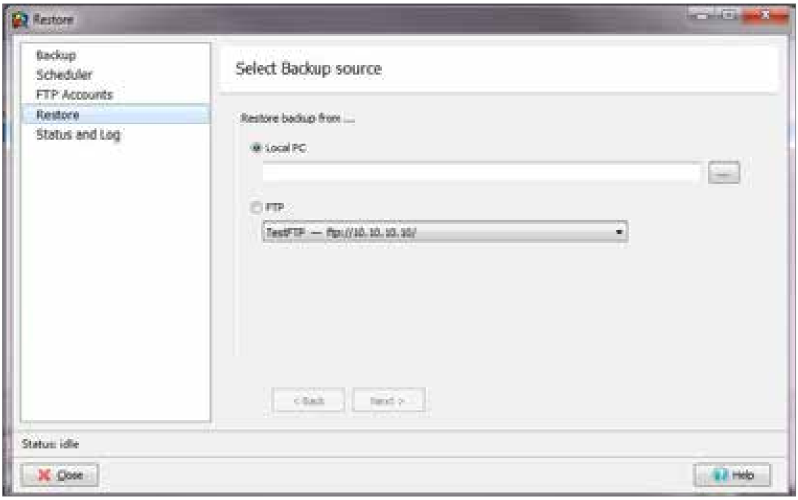
Backup/Restore

To backup or restore the device, navigate to **“Maintenance ▸ Backup”** or **“Maintenance ▸ Restore”.**



If for example you selected Backup, you will then have a window come up giving you the option of specifying the items you would like to back up. Check as desired then click on **“<Select Desitnation...>”** or **“Backup Now”** depending on whether or not you previously performed a backup.

Specify your destination folder and the files will be backed up. To restore, select **“Maintenance ▸ Restore”** and you will see this window:



Simply select your file and proceed with the restoration process.