



Optimum Business Trunking and the Zultys MX250 IP PBX Configuration Guide







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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.
- 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

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





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

optimum.	SIP Trunk Configu	ration Help
Configuration Menu • Customer • <u>LAN Settings</u> <u>SIP Trank</u> <u>Configuration</u> • <u>Diagnostics</u> • <u>System</u>	Select your PBX: Passive connection (This address must be on (interface PBX Address: Active connection us User Id: Password:	America: Ame
	Convert Inband DTMF:	
	Trunk Status: DID's 516433599 (Pior number) 516433768 516433769 516433765 516433841	Not Registered

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.

optimum	Network Test Tools	lp
optimoin.	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	
Configuration Menu • Customer • LAN Settings SIP Trunk SIP Trunk	network. Outboand Call Test: This test will bloc a call to the provided telephone number and play a series of tones for 3 seconds. Plich Number: Seconds.	0
Diagnostics System	Telephone Number:	
	Inbound Call Test: When this test is enabled calls received for the pilot number are diverted to the internal Te UA for 13 minusci, after this algoand time the test is automatically disabled. © Enabled Summ:	st
	Ping Test: IP Address to Ping: Ping Reset	
	Traceroute Test: IP Address to Trace:	



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.	System Help
Configuration Menu	Software Version: Version 11.6.14.1 Fri Jan 4 17:49:28 PST 2013
Customer LAN Settings SIP Trunk	Hostname: 5164939899
Configuration Diagnostics System	Model: EdgeMarc 4552
	Vendor: Cablevision
	LAN Interface MAC Address: A8:70:A5:00:D8:18
	Registration Status: The ALG feature is registered. View <u>license kev</u> .
	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.	Set Password Change the GUI password by filling	ng in the fields below. The password	<u>Help</u>
Configuration Menu • Customer • LAN Sattings SIP Trunk Configuration • Diagnostics • System	User be between 6 and 8 charact Username: Current Password: New Password: Confirm Password: Submit Reset	pixinstall	



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.

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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.

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Network Settings

Navigate to "Provision > Wizard".

Prov	ision Configure	Auto Atten
\$	Wizard 🕞	
	Locations	
2	SIP and RTP	
	System Clock	

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"**.

Company	Company Settings		
ContactInle	Conpany Name	Edgewater	
IP Addresses	Cofedt Annan	10.10.177.H	
Servers	Conceptual and and	154	12
DHCP	courreyor regor		
Provies	State	California	*
TTS	Oly in Linn	San Jose	
LDAP Configuration	Language	English US	•
Web Services	Cal propess tores	1 USA	
Miscellaneous	Country code	1	
	Main phone number	6314488988	
	Name for this location	Default	
	Time Zone	America/Los_Angeles	1
	1 1 1 1 1 1 1 1 1 1 1 1 1 1 1 1 1 1 1		



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.

Company Contectivits	Steed darges		ula (r. m.d.s.) Labora francisco	02.010.1
#FARMERS .	There are a			
Servers	in Address from		THE THE LOSS	11 11
DHCP	10 Address \$17	2	48. 10 127	
Provina	subcertrait.		120.205.200	us left
TIS	18 Million 1	TT.	18.38.129.	B
	Externel pints in a			
LDre-Comgutation				
Web Services	-	Perfort	Pattent	2wm
Univ Congutton Web Services Micelarecus	Applexiten	Protocti JDP	P Address 92. (A. 127.13	Parm
LDN+ Computers Web Services Miscellareout	Applexter 325 32	Protocol JOP JOP	P-Address 00.10.127.13 00.00.127.11	Parts 30000 20995 3000
LDre Doniguration Web Services Mocellaneous	Acolector 315 59 36	JOP JOP JOP	P Arthest 98, 99, 127, 13 98, 99, 127, 11 98, 99, 177, 11	Parts 30000 20995 3040 7.000 - 21239
LDre Conguiston Yueb Services Mocellaneous	Acolestan 115 59 96: 96:	Archaean JOP JOP JOP	P Attest 30.90.127.13 30.90.127.11 30.90.117.11 30.90.117.11	Parts 30000 - 20995 9840 7.000 - 21229 3306
LUNe Computer Web Services Miscelereous	Acalenter 105 59 106 106 106 106 106 106 106 106 106 106	JOP JOP JOP IOP TOP	P Address 22 97 427 43 25 97 427 43 26 97 427 41 25 97 427 41 26 97 427 41 27 97 427 41	Parts 30000 20095 5000 2000 - 21229 3208 7300 - 7357
Lune Computer Web Services Miscelareous	Applexton 305 38 39 39 39 39 39 39 39 39 39 39 39 39 39	Protects 109 109 109 109 109	P-Address 32.99.127.13 35.99.127.13 35.99.127.11 35.99.127.11 35.99.127.11 36.91.127.11	Parts 20000 20096 5060 2.000 - 21236 2000 7.000 - 7157 56

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"**.

Company Contact info IP Addresses	Chill Services Barney Decondery	1.1.1.1 1.1.1.1	
DHCP Proces	Terlery DHP		
TTS LDAP Configuration	I Stand	C Esternal	1
Web Services Miscellareaut	Ribbernal External taking	() Ddenai	15 p receive
	E Stand (PCA. en	er D' to the following fronts	
	Part 8	10	





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"**.

Company Contact Info IP Addresses Servers	Scope	P Address 10.20 P Address 10.30 Aust 255.255	127, 15 127, 20 235, 4 (11)	
Proves TTS	Lose Davi	ten E days 0 [5] hau	n <u>0 10</u> e	intes
the conganon	Centres	Marra .	Value	(mint)
Web Services	1	Router Default Satenan	COLUMN A	
Miscollaneous	4	Premary DNS Server	42.2.2	M
		Secondary DNS Server	8.8.8.8	-
		Fertary DNS Server		
	25	Sentary DNC Server Sentaut Doman	10.10.127.11	
	3	Serlary DNS Server Seflaut Lonan meany NTP Server	83, 59, 127, 11 83, 59, 127, 11	0
	13 92	Fentary DRG Server Sefault borrant menary NTP Server Secandary NTP Server	10.10.127.11 10.19.127.11	0
	3	Ferfany (INC Server Serfault Lomant Himany NTP Server Secundary INTP Server Ferfany NTP Server	49, 19, 127, 11 49, 19, 127, 11	0

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

Company Contact Info	Personal call recording Play beeps at start	
IP Addresses	2Rey beeps every	10 Internet
Servers DHCP	Pas Settings Number of vehics	1.5
Proxies	Internal between retries	s 🗄 misster
TTS	Company Name	
LEAP Configuration	TAX runber	
Web Services	For calls forwarded by MD: Use main number D Preserve original caller is	for incoming call





Navigate to **"Provision ► SBC"**.

Prov	ision Configure Auto Atter
\$	Wizard
۲	Locations
R	SIP and RTP
	System Clock
2	System Settings
4	SIP Servers and ITSPs
++	Codecs
8	Bandwidth Management
8	Analog (FXS)
由	Analog (FXO)
an.	PCM
	Firewall and NAT
VPH	VPN Configuration
4	BRI Interfaces
1	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**.

Add	D
Delete	
-	





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.

Verworks RTP	Mapping							
Session Border	Controller RTI	Poort range						
Start port 21	000	Endpoit 21223	9					
Networks								
Authenticate	al tallic hor	untrialed networks						
C Block all hal	lic from unitial	ted networks						
D Con	gured SIP Se	410000	and based on	To shad an are				
11-22		rvers and 115Ps are all	ethic search of	Housed House	-			
Network Addre	n NML	Mask	Trusted network	Pot Maping	Public IP	External SIP Pod	Esternal RTP Port Range Starting	External RTP Port Range Ending
Network Addre	12 NML	Mark 255 255 255 0	Tructed network	Pot Mapping	Public IP	External SIP Post	External RTP Port Range Starting	External RTP Pot Range Ending

When done click **Apply**.

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

letworks RTP Mapping			
Route RTP via Session	Border Controller for calls b	etween selected netwo	ork pair:
	10 10 127 0/24	0 0 0 0/0	1
	10.10.121.0724	0.0.0.070	
10.10.127.0/24		0.0.0/0	

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.



seneral	Regional	SIP	JP & Provisio	ning Aude	ARTP	VLAN	Keys	Advanc
ines								
Line	e Enable	Registrar Source		Registration Expires		res	Voice Mal	
- 1		MX Add	ress	3600			voice.ma	si
2				8 8802				
3								
1	Registration	details	for line 1		_			
1	Registration	details	for line 1 Addre	ess Source		Addres	8	Port
[Registration Registrar	detais	for line 1 Addre	ess Source	10, 1	Addres 0.127.1	8 1 5	Port 060
	Registration Registrar Backup Reg	details i	for line 1 Addin Do not us	ess Source	10.1	Addres 0. 127. 1	8 1 9	Port 060





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

.H				
200				
2010/100	20120325.8	-34-		
Salad Lines	41.0.0101			
arved .		_		
Millionet	10.00.121.11			
(14) (16)				
Income and				
Patent	UNIT DOM: NO			
there a	0.00			
	47.10.107.11			
	C'real .			
			1.1	344
100				
the fame				

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".

Insert 5	les.
Edit ¹⁶	12
Delete	Chd+Del
Profiles	
Show Assignment_	
Show Device Status	
Set Screen Name to	6
Regenerate configuration files	

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**.

New Derivs		U.M. MCGad
Deletion	2238	•
iii kat onge droor ⊖ kat onlije in o		
		100





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**.

rfault Location	Defailt .
NR: ()	
AC Address	00-08-EA-82-82-A6
ine 1	
Device ID 😥 Use MAC	000864828246
Screen Nane	Ext. 201
SP Auth, User Nime	000BEA8/H2A6
SIP Proxy Password	
Par an income	Advanced Centerer
But to prove deriv	Buarden official

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"**.

View	Support Window Help	User Agent	Ac
1	Device Status	Tullys 277-278	τu
2	MOTE & CSTA Status	Zuitys ZIP 37	Tru
-	Sessions		

Now navigate to "Configure > Users".







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

New User	Ins
Delete User	Del
Edit User	
Reset 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.

sign

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
9	Tim	Steven	101
5	mike	anderson	100

	ID	Profile
핥	0008EA82A70C	Profile ZIP 37G
8	000BEA82B2A6	Profile ZIP 35i

œ⊜ Assign

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.

0000738121256	Intert	Inc.	Denume IP Celluit
	Edt	. 12	
	Delete	Ctrf=Del	
	Profiles		
	Show Assignment		
	Show Device Statut	_	
	Set Screen Name to	•	Device Id
	Regenerate configuration files		User Extension
			User Full Name User Last Name = Est.
			User Full Name + Ext.

When done click **Apply**.

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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.

Routin	Routing Outside Call Restriction					
	Name	Source	Pattern	Destination	Transformation	Restricted
1	Internal extensions	Internal	100	Extensions	2000	
2	100g1	Internal	1000000000	SIP Server : NevSIP	000000000	
3	19ka	Internal	13000000000X	SIP Server : NewSIP	1000000000	
4	Sha_Strefx	Internal	910000000000	SIP Server : NewSIP	D10000000000	
5	200gt_9Ptefx	Internal	91000000000	SIP Server : NewSIP	210000000000	

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"**.

Prov	ision Configure Auto Attend
5	Wizard
	Locations
N	SIP and RTP

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"**.

New Phone	Ins
Delete Phone	Del
New Route	Ctrl+Ins
Delete Route	Ctrl+Del

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

ocation Name	Time Zone	Caller ID
Default	America/Los_Angel	
Default	America/Los_Angel	

When done click **Apply**.

Emergency Routing				
Number Dialed	Route	Number Transmitted		
14.5	SP Server : NexSP	911		

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**.

The second s	2º large fir Leuten	
P Rege 8. il. 127.3(24	New Ins Dy Rolt Proser Debate Del	

New IP range	8. X
C P Rarge	140 P
# Submet IP 15 15 127 0	Submet 255,255,255, dj [4]4
V CK	Rep





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"**.

SIP Servers ITSPs	Authentica	noon			
Name	Active	Type	Codec Profile	Max Video Quality	SJP Profile
New50 ⁵		External	Voice Quality	None	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.

Serversitier				
Chepetin	with the			
# Los te his	ming servers			
	Address		Fort	
100	1210.021		500	t
4				
legisted as				
2. Feglile				
UnrMarie	6210468308	Trienk	2683	
And a local division of	faire-ent true prin		Auto	





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"**.

From [*] domain	"To" domain Request URI Calir	r ID Misc
🗇 Use addres	is of the MX	
Change to	MX domain if device belongs to use	σ.
() Use addres	is of the Server	
diama di Al	Invino Address	

"From" domain	*To* doman Request-URI	Caller ID Misc
FROM Header	Originating party Caller ID	•
[[]] Indu	de optional identity header	
	Registration user name	

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.



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Realm			and be
RealmDown	TOTOTAL		
User Name	6314468996		
Password		Cirilien Painword	•••••
1 CK.	X Cance		R Heb

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:

NexCIP Servers List	Ê.		
Regular	using DNS_SRV from		
i Use the	following servers		
	Address	Part	
-A Die	27.10.127.1	5060	6
1.4			
\$ 10mm			
Registration	10		
Register			
Q: Internet	m: 611+45895	mout 3600	H
Page St	water har have as	Auto	
1000	And the second second second	Martin Int	101 -

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"**.

Conf	figure	Auto Atte	ndant	Mainten
0	Dial P	lan		
6	Phon	e Services	23	
	Syste	m 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.

5	Phone Service	1				
ſ	Auto Attendants	Paging Groups	Servers	Group Pidup	Conference	Voice.Mixer
	Name	1	Extension	DID	Caler	Id
	MainAA		150	63144889	90	

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**.

Auto Attendant & IVR Scripts	0 8 8
× (2 3 4 ×	
A Scourt Jinformation A Scourt Jinformation Default Default Default ScopeSumple	
R Enter Project Name	
Project name	
DayAA	
	_
 - project was successfully complete and uploaded - project was not complete and can not be used 	
- system preinstalled project	
X Dose	Help

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.

User input	Actors	
The Monith	Uport from t	
Nagel Jage. #	Angenation second	
5	Dailey Neren, Steerch by Advisert, Attempts-3	
120	Transfer Destination from [Sout_desig]	

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"**.

New	1000			
目	Schedule type # Loops of visual 	State of week 12 Society 12 Noveles 12 Noveles 12 Noveles	iil ruesiae iil ruese iil ruese iil totories	
i do	Rocateres	-	- inees 1	
1	sear (2005A	•)		

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"**.

Maintenance		View
	Backup	
8	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.

Backup	Specify backup items	
Scheduler ITP Accounts Restore Status and Log	Configuration Auto Attractant Scripts Auto Attractant Scripts Auto Attractant Scripts Configuration Configuration and Call Groups Control Call Call Groups Control Call Call Reports Control Call Call Call Reports	
tus idle		

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



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Restore		
Backup Scheduler FTP Accounts	Select Backup source	
Restore	Restore badup from	
Status and Log	@ Local PC	
	OFF	
	Test#TF — Rps(/100.10.10.10.10)	•
Status idle	Citab. Indo-	
X Qose		12 Heb

Simply select your file and proceed with the restoration process.