



Optimum Business Trunking and the CudaTel 2.6.004 IPPBX Configuration Guide





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1 Overview

The purpose of this configuration guide is to describe the steps needed to configure the Cudatel 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 Configur	ation	Help
Configuration	Select your PBX:	Asterisk	
Customer <u>LAN Settings</u> <u>SID Trunk</u> <u>Configuration</u>	 Passive connection u This address must be on th interface PBX Address: 	sing the local, private IP address of the PB he same subnet as the IP Address that is specified for	X the LAN
 Diagnostics System 	 Active connection using the set of the set	ng registration	
	Password:	secret	
	Convert Inband DTMF:		
	Submit Reset		
	Status:		
	Trunk Status:	Not Registered	
	DID's		
	5164939899 (Pilot number) 5164939768 5164939769 5164939795 5164939841	20 7	

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.

ontimum	Network Test Tools
optimoni	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 Senfiguration • Diagnostics	network. Outbound Call Test: This tast will place a call to the provided telephone number and play a series of tanes for 30 seconds. Pilot Number: Telephone Number:
, <u>System</u>	Baset Inbound Call Test: When this test is enabled calls received for the pilot number are diverted to the internal Test When this test is enabled calls received for the pilot number are diverted to the internal Test When this test is enabled.
	© Enabled ≪ Disabled Submit
	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 in the fields below. The password muct be between 6 and 8 characters in length					
Configuration Menu • Customer • LAN Sattings <u>SIP Truck</u> <u>Configuration</u> • Diagnostics • System	Username: Current Password: New Password: New Password: Confirm Password: Submit Reset	poxinatal				



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.



Important:

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

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

CudaTel

optimum.

5 PBX Configuration

To access the PBX configuration GUI point the browser to 192.168.200.200 (unless changed from the default IP previously). The below login page will appear.

riease L	og In							
Username:								
Password:	Log In	6	all C	ontro	I Clis	Int		

Figure-1

The default username and password login is:

Username: admin

Password: admin

A screen similar to the below image will be displayed upon successfully logging into the PBX.





Confirm the firmware is 2.6.004.

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Select the **Configuration** button in the upper panel on the far right, select the **Network** link on the left panel.

🖌 CUDATEL	Destitionerd	entoteoura	Latersians	Konsters	Aquata Config	atos
Configuration Addator Herwork Security System Proces Mail Directory Services/LDAP	Network LAN interface			(P Address Subnet Maak Gateway	10 - 10 - 123 - 11 256 - 256 - 256 - 0 10 - 10 - 123 - 1 Une Luk pathway as Definit Gateway Behind HAT Fourter 0
Rackop, 0 Front Log Updatos Extensions Saunda and Monic Traublastica ling	WAN starface				P Address Saboet Mask Gateway External & Address	WAII Interface Enabled

Figure-3

Modify the following settings:

LAN Interface: This section will configure the static IP address of the PBX. IP Address: Enter in the IP address to be assigned to the PBX.

Subnet Mask: Enter in the subnet mask for the network assigned to the PBX.

Gateway: Enter in the Gateway IP address to the network. By default, this is the LAN side IP address of the upstream Optimum Business SIP Trunk Adaptor.

Use LAN Gateway as Default Gateway: Uncheck this box.

Behind NAT Router: Uncheck this box.

WAN Interface Enabled: Uncheck this box. No other fields should be populated.

Scroll to the bottom of the page, click on the **Apply Changes** button.

With this Network configuration, the WAN port on the PBX will no longer be in use. The LAN port will be plugged into the Optimum Business SIP Trunk Adaptor's LAN port or the switch that plugs into the Optimum Business SIP Trunk Adaptor's LAN port.

On the left panel, click the **Phones** link. The page below will appear.





Set Local Area Code	or www.authound.dollar	
Area Code: 831 Ringback Tone	concerning of the second second	
Ringback Tone	United States	
Ringback Tone (on Transfer)	United States	
Operator Extension When a caller dials "0", transfer them to: Automatic Provisioning If you are using the Phane Server on a network provisioned to hetsp://10.10.123.11/provi. Automatic Provisioning Off • Codecs For locoming Calls 1/21 selected + For Outgoing Calls 1/21 selected +	k with other PBX devices, you may wish t ston/. See your quick-start guide or man	o turn off as uait for provi
	Set Local Area Code Set your local area code to allow seven-digit () Free Code: 831 Ringback Tone Ringback Tone Ringback Tone Ringback Tone (on Transfer) Operator Extension When a caller dials "0", transfer them to: Automatic Provisioning If you are using the Phone Server on a networ provisioned to http://10.10.123.11/provi Automatic Provisioning Off Codecs For incoming Calls 1/21 selected * For Outgoing Calls 1/21 selected *	Set Local Area Code Set your local area code to allow seven-digit (XXX-XXXX) outbound dialing. Area Code: Bingback Tone Ringback Tone United States Ringback Tone (on Transfer) United States Operator Extension When a caller dials "0", transfer them to: Automatic Provisioning If you are using the Phone Server on a network with other PBX devices, you may wish to provisioned to http://10.10.123.131/provision/. See your quick-start guide or manuary of the Provisioning Off • Automatic Provisioning Off • For incoming Calls 1/21 selected * For Outgoing Calls 1/21 selected *

Figure-4

Set Local Area Code: Enter in the Area Code the DID's will use.

Automatic Provisioning: Set this drop-down box to Automatic Provisioning Off.

Codecs: For Incoming Calls, and For Outgoing Calls unselect ALL codecs except the G.711u and G.711a option.

All other options and fields should be left blank or left to the default configuration. On this page there is no Apply Settings options, this is done automatically as soon as the options are set.

Click the **Extensions** link on the left panel. The below page will appear.





Configuration	Valid Extension Blocks	
Activation Network Security		Berrove Berrove
System Phones	1000 · 1009 Valid extension selected Add New Extension Block	
Directory Services/LDAP Backup	Add an extension block	
Event Log Updates		
Extensions		



Click the Add an Extension Block link.

A new field will prompt for an extension range, in the first box enter the starting extension such as 2000, enter 2999 in the second box.

Click the blue **Add New Extension Block** button to complete this change.

On this page there is no Apply Settings button, settings are saved by default.

On the upper panel, click the **Providers** link. Click the **New SIP Account** link. The below window will appear.

For Registration Mode: Follow the below settings.

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Provider/Type	Optimum
Name	ToEdgeMarc
Host	10.10.123.1
Port	5050
Realm	10.10.123.1
Username	6314488968
Auth. Username	6314488968
Password	
Registration	Requires Registration

Figure-6

Provider/Type: Select the Optimum option from the drop-down list.

Name: Set a Name. ToEdgeMarc was selected for this example

Host: The LAN IP address of the Optimum Business SIP Trunk Adaptor.

Port: Cannot change this setting, defaults to 5060.

Realm: Set this to the Optimum Business SIP Trunk Adaptor's LAN IP address, should match the Host field.

Username: Pilot DID or assigned username for registration to the Optimum Sip Trunk Adaptor. This must match what was entered in the Optimum Business SIP Trunk Adaptor in Step 3 of the SIP Trunk Set-up Guide.

Auth. Username: Same as Username field. This must match what was entered in the Optimum Business SIP Trunk Adaptor in Step 3 of the SIP Trunk Set-up Guide.

Password: Registration password. This must match what was entered in the Optimum Business SIP Trunk Adaptor in Step 3 of the SIP Trunk Set-up Guide.

Registration: Check the **Require Registration** box and set the second interval field to the recommended value of 120.

Services: Check both Inbound Calls and Outbound Calls.





Leave all other settings as default. Scroll to bottom and click the **Create Gateway** button.

For Non-Registration Mode: Follow the below settings.

146	UDATEL Sature Suntaines Documen	na in internet
d a **	New SIP Provider Account	
Nort-sus for	Provider/Type	Optimum 💽
Calibrative	Name	ToEdgeMarc
	Host	10.10.123.1
	Port	5060
	Realm	10.10.123.1
	Usernane	
	Auth. Username	
	Password	
	Registration	Requires Registration
	Services	III Inbound Calls III Outbound Calls

Figure-7

Provider/Type: Select the Optimum option from the drop-down menu.

Name: Set a Name. ToEdgeMarc was selected for this example.

Host: The LAN IP address assigned to the Optimum Business SIP Trunk Adaptor.

Port: Cannot change this setting, defaults to 5060.

Realm: Set this to the Optimum Business SIP Trunk Adaptor's LAN IP address, should match the Host field.

Username: Leave blank.

Auth. Username: Leave blank.





Password: Leave blank.

Registration: Uncheck and Leave blank.

Services: Check both Inbound Calls and Outbound Calls.

Leave all other settings as default. Scroll to bottom and click the **Create Gateway** button.

A new Gateway will appear in the Provider's page. Click on the new Gateway. The below page will appear.

Provider/Type	Optimum
Host	10.10.123.1
Port	5060
Realm	10.10.123.1
Username	6314488968
Auth. Username	6314488968
Password	
Registration	Requires Registration 60 second interval Refresh Registration Registered
Services	 ✓ Inbound Calls ✓ Outbound Calls ✓ Faxes

Figure-8





Scroll to the bottom to continue the configuration.

Caller ID Number	6314488958 Never use a custom Caller ID number Use a custom Caller ID number unless overridden Always use a custom Caller ID number
Outgoing Music on Hold	default 💌
Restrict Codecs To	2/21 selected +
Inbound Registration	E Allow Inbound Registration
	Apply Gateway Settings
External Numbers	(631) 448-8968 (631) 448-8969 (631) 448-8970 (631) 448-8971 Add Enternal Numbers
Outbound Routing	10 Digit Dialing
	7-digit Dialing (Area Code 631)
	International Dialing (011)
	Emergency (USA)
	Marann Routes

Figure-9

Caller ID Number: In this field enter the Pilot DID.

Select the **Use a Custom Caller ID Number Unless Overridden** radio button.This setting will use the Pilot DID by default unless an extension specifically overrides it. To force the extensions Caller ID, select the **Never Use a Custom Caller ID Number**. To always force the Pilot DID on outbound calls select the Always **Use a Custom Caller ID Number**.

Outgoing Music on Hold: There are two options, default and silence. Set this option to determine what the remote user hears when put on hold.

Restrict Codecs To: Select G.711 ulaw and G.711 alaw.

Inbound Registration: Uncheck for security reasons.

External Numbers: Click the **Add External Numbers** link. Starting with the Pilot DID, add each DID the PBX will use.

Outbound Routing: Leave to default, this will auto-populate.

Click the Apply Gateway Settings button.





In the upper panel, click the **Extensions** link. In the new left panel, click the **People** link then click the **Add New Person** link. The below window will appear.

\∡G	WATE	E .H.(10 C		
Au 11	Add New P	erson			
Chan	First Name:	2005	Assign this person to an extension:	Select by choosing a phone	
internet Carl	Last Name:	2005	Dat.		Hame
California II	PIN (4+ digits)	1357	To unaccigned phones are available for use	To use an assigned phone, remove it fo	on the sea
Publicitier Co	Group:	(No Group Membership)			
Chinada a se	Estemion:	(Hext free extension)			
line igned to		Add Cancel			
ACES					

Figure-10

Fill in the **First Name, Last Name**, and a **4 digit PIN** for the new extension. Leave all other options to default. Click the blue **Add** button.

After the creation of the user, click on the new user listed under the **People** link and the below page will appear.

(BACK x2000 - FN2000 LN2000 Secana Cheva Adventer Deleter	Insecuents
Constact Information Las Texture (U1) 44-310 your P Show this person is Contact Detectory searches Apply	Groups The thir is empty- Adm. a licesus
Voice Aail Dtaile Voice Aai Couge PMPRevoul Couge PMPRevoul Exter a see PMPRevoul Exter a see PMPRevoul Exter a see PMPRevoul Exter a see PMPRevoul Externation Do not aerd voice-mate (internation Voice-mate Forest) Courses are used Voice-mate (internation Courses are used Voice-mate Courses are used Voice-mate (internation	Call Recording Policy Record calls and save for days. Serve to Guard Address Operator Extension When the user date "0", transfer to: Language Set the language used ary value promptly (Face wall instructions, etc.) Engine, UR
Add a Phene PAQ0001 Polycae	

Figure-11

Contact Information (Optional): Click the **Edit** link to modify the user's contact details.





Phones: Click the Add a Phone button, the below will appear.

Generic SIP D	evice
Manually Ente	r a MAC address:
Select a Phon	ie
Search: P	
Ext,	Name
and the second se	The second s

Figure-12

Click the **Select a Phone** button, click an unassigned phone listed. Click the blue **Add Phone** button to complete. The User page should reappear with the phone listed.

Phones	~	
Add a Phone	FN2000's Polycom IP335	
Secondary Nu	mbers and Extension	s
Secondary number	s and extensions will be for	warded to the main extension (x2000).
Add Number		Secondary Extensions and Numbers
Valid extension ran	ges: 2000-2999	×



Under the **Add Number** section select **External Number** in the dropdown box. A second drop-down box will appear, select a DID to assign to this extension. Click the **Add Extension** button.

Click the **Apply Setting** button.





Click the **Call Parking Extensions** on the left panel. Then click the **Add New Parking Extension** link.

Extensions	Parking Add New Parking Extension
People	Ext. Name
Groups	No parking extensions found
Inbound Call Queues	
Call Parking Extensions	
Multi-User Conferences	
Automated Attendants	
Unassigned Phones	
AL Extensions	
-	

Figure-14

The below window will appear.









Parking Lot Name: Name the Parking lot.

Extension Block: Enter 700 in the first field, 705 in the second field.

Music on Hold: Set to **Default** or **Silence**. The party transferred into the Parking Lot will hear what is defined.

Maximum Hold Time: The duration the party is allowed to be parked.

After Hold Time, Transfer To...: Where to send the caller after the hold time is up. 2999 in this example is the Automated Attendant (recommended).

Click the blue **Add** button to apply changes.

In the left panel click the **Automated Attendants** then click the **Add New Automated Attendant** link.

Extensions	Auto Attendants Add New Call Router Add New Automated Attendar
People	Ext. Name
Groups	No automated attendants found
Inbound Call Queues	
Call Parking Extensions	
Multi-User Conferences	
Automated Attendants	
Unassigned Phones	
All Extensions	

Figure-16

The below window will appear.

Automated Attendant	Name: AA			
Extension:	Valid on	Come	ion ranges:	2009-2009. 991-999
	Single	Exte	nsion 🔄	2999
			cremsion ser	erted
C		-		
Greeting Sound	ivr-anonymous_call	•	ivr-anonym	ious_caller.wav 💌 🛽
Greeting Sound Short Greeting Sound	ivr-anonymous_call (None)	•	Ivr-anonym (None)	nous_caller.wav 💌 🛿
Greeting Sound Short Greeting Sound Invalid Sound	ivr-anonymous_call (None) (None)	•	(None) 💌	nous_caller.wav 💌 1]]

Figure-17





Automated Attendant Name: Give the AA a name.

Extension: In the drop-down box select **Single Extension**, enter in the Automated Attendants extension in the field.

Leave other options as default.

Click the blue **Add** button to save the changes.

Click on the new **Automated Attendant** that now appears under the Automated Attendant page. Configure to each option as desired.

Important: The DTMF tone duration generated by the phones needs to be increased from the default value of 180ms-200ms to 600ms. The PBX does not have the capability to change the DTMF settings, the change must be done on the phones. The Optimum Business Sip Trunk Adaptor needs to be configured to Convert Inband DTMF. This is Step 3 in the Optimum Business Sip Trunk Set-up Guide.